

TALKSWITCH USER GUIDE

TALKSWITCH VS MODELS

RELEASE 4.10

CT.TS005.003901

For use in
North America

talkswitch®



ANSWERS WITH INTELLIGENCE®

Copyright Information

TalkSwitch Copyright 2007 — All Rights Reserved.

TalkSwitch is a division of Centrepoint Technologies Inc.
TalkSwitch and Appello® are registered trademarks of Centrepoint Technologies Inc.

Reproduction, adaptation or translation without prior written permission is prohibited, except as allowed under the copyright laws.

Information in this user guide is subject to change without notice and does not represent any commitment on the part of TalkSwitch. No part of this user guide may be reproduced or transmitted in any form or by any means, electronic or mechanical, including photocopying, recording, or information storage and retrieval systems, or translated to another language, for any purpose other than the licensee's personal use and, as specifically allowed in the licensing agreement, without the express written permission of TalkSwitch.

First Edition, March 2007
CT.TS005.003901

TABLE OF CONTENTS

PREFACE

What's in this guide?	VII
Single unit installation	VIII
Networked units installation.....	VIII
VoIP installation.....	VIII
Connecting devices	VIII
Finding the information you need	IX
Using the table of contents and the index	IX
Navigating with cross-references	IX
Where to go for further information	IX
Guide conventions	X

CHAPTER 1: TALKSWITCH INSTALLATION

1.1 TalkSwitch package contents	1
1.2 Configuration software system requirements.....	1
1.3 Unit front panel	2
1.4 Unit back panel	3
1.5 Installing the configuration software	5
1.5.1 Installing the software for the first time.....	5
1.5.2 Upgrading the TalkSwitch software and firmware.....	5
1.6 Initial configuration	8
1.7 Select Location for Installation	9
1.8 Connect TalkSwitch to a network or a PC	10
1.8.1 Ethernet connection	10
1.8.2 USB Connection.....	11
1.8.3 Internet connection	12
1.8.4 File connection.....	14
1.9 Connecting devices	15

1.9.1	Connecting incoming telephone lines	15
1.9.2	Connecting local extension telephones and other devices	16
1.9.2.1	Connecting a regular single-line telephone	16
1.9.2.2	Connecting a regular dual-line telephone	17
1.9.2.3	Connecting an IP phone	18
1.9.2.4	Attach an internal or an external modem	18
1.9.2.5	Connecting fax machines	19
1.9.3	Connecting devices to the music jack	21
1.9.4	Connecting to the PA (public address) jack	21
1.10	Networking TalkSwitch units on a LAN	21
1.10.1	Connecting TalkSwitch units to a LAN	21
1.10.1.1	Ethernet switch	21
1.10.2	Power up all the TalkSwitch units	22
1.10.3	Setting or changing the unit ID	22
1.10.4	How unit IDs affect system extension numbers	23
1.10.5	Keep track of the lines and extensions	24
1.10.6	Optimizing the system for networked use	25
1.11	Increasing Memory Capacity	26
1.11.1	TalkSwitch memory cards	26
1.11.2	Adding internal memory	26
1.11.3	Verifying additional memory	27
1.12	Upgrading TalkSwitch Units	27
1.13	Enabling Licensed Add-Ons	27
1.14	What the flashing lights mean	28
1.15	Verifying the connections	29

CHAPTER 2: TALKSWITCH CONFIGURATION

2.1	System Configuration	31
2.1.1	The configuration screen	31
2.1.1.1	File Menu	32
2.1.1.2	View Menu	32
2.1.1.3	Tools Menu	33
2.1.1.4	Help Menu	33
2.1.1.5	Configuration navigation	34
2.1.2	Special considerations when connecting multiple units to a LAN	35
2.2	System Information	37
2.2.1	Administration	37
2.2.2	IP Configuration	39
2.2.3	Analog incoming telephone lines	42
2.2.4	Line hunt groups	44
2.2.5	Automatic Route Selection and Toll Restriction	45
2.2.5.1	Activating ARS and Toll Restriction	48
2.2.6	Fax information	51
2.2.6.1	Dedicated fax line	52
2.2.6.2	Distinctive ring fax detection	54

2.2.6.3	Automatic fax detection	55
2.2.7	Local extensions	56
2.2.8	Remote extensions	61
2.2.9	Extension ring groups	64
2.2.10	On-Hold/Ringback	66
2.2.10.1	External audio source	67
2.2.10.2	Internal audio file	68
2.3	Voicemail	70
2.3.1	Mailbox options	70
2.3.2	Voicemail notification	72
2.3.2.1	Dialed notification	72
2.3.2.2	Message waiting light	74
2.3.2.3	E-mail notification	75
2.3.3	Global Settings	77
2.3.3.1	Voicemail manager	79
2.4	Call Handling	81
2.4.1	Modes	81
2.4.2	Auto attendant	83
2.4.2.1	Adding/configuring an auto attendant	84
2.4.2.2	Record, play or erase auto attendant messages	85
2.4.2.3	Select routing options for each auto attendant	87
2.4.2.4	Automatic fax detection	89
2.4.2.5	No selection was made at the auto attendant	89
2.4.2.6	Additional features at the auto attendant	89
2.4.3	Incoming telephone Lines	90
2.4.3.1	Incoming telephone line call handling options	91
2.4.4	VoIP Numbers	92
2.4.5	Call Cascade	93
2.4.5.1	Local extensions	93
2.4.5.2	Remote extensions	96
2.4.5.3	Extension ring groups call cascade	98
2.4.5.4	Call cascade examples	100
2.5	Call Back/Call Bridge	102
2.5.1	Auto call back	103
2.5.2	Prompted call back	105
2.5.3	Call bridge	107
2.6	Options	108
2.6.1	Permissions	108
2.6.2	Audio Controls	111
2.6.3	Transfer Options	112
2.6.4	Miscellaneous	114
2.6.5	Troubleshooting	116
2.6.5.1	Troubleshooting — Advanced	118
2.7	Licensed Add-ons	120
2.7.1	Enabling Add-on demos	122

2.7.2	Enabling Add-ons using an Activation Key	122
2.7.2.1	Step 1: Obtaining a License Code	122
2.7.2.2	Step 2: Obtaining the TalkSwitch System ID	122
2.7.2.3	Step 3: Activating the Licensed Add-on	123

CHAPTER 3: USING TALKSWITCH

3.1	In the Office — Receiving calls with or without the auto attendant	125
3.1.1	Receiving calls using the auto attendant	125
3.1.2	Receiving calls without the auto attendant	126
3.2	In the Office — Making and receiving calls using an analog phone	126
3.2.1	Making calls from a local extension without direct line access.	126
3.2.2	Receiving calls at a local extension	127
3.2.3	Placing calls on hold at a local extension	128
3.2.4	Transferring calls from an extension	128
3.2.4.1	Unscreened transfer	128
3.2.4.2	Screened transfer	129
3.2.5	Transferring calls from a local extension to an external phone.	129
3.2.6	Parking and retrieving calls at a local extension	130
3.2.6.1	Parking a call	130
3.2.6.2	Retrieving a parked call at another local extension	130
3.2.6.3	Using call park with the paging option	130
3.2.7	Queuing and retrieving callers	131
3.2.7.1	Queuing calls to a single extension	131
3.2.7.2	Queuing callers to an extension ring group	132
3.2.8	Using call waiting	132
3.2.9	Conference calling	133
3.2.9.1	Two local TalkSwitch extensions and one outside caller. .	133
3.2.9.2	Two outside callers and one local extension	133
3.3	In the Office — Making and receiving calls using an IP phone	134
3.3.1	Making calls from a local IP extension (IP phone)	134
3.3.2	Receiving calls at a local IP extension	135
3.3.3	Hold and transfer	136
3.3.3.1	Hold	136
3.3.3.2	Unscreened transfer	136
3.3.3.3	Screened transfer	136
3.3.3.4	Transfer from a IP extension to any outside number	137
3.3.4	Call park — Parking and retrieving callers	137
3.3.4.1	Parking a caller	137
3.3.4.2	Retrieving a parked call	138
3.3.4.3	Using call park with the paging option	138
3.3.5	Queuing and retrieving callers	138
3.3.6	Using the TalkSwitch call waiting feature	139
3.3.7	Conference calling with TalkSwitch	139
3.3.7.1	Two TalkSwitch local extensions and one outside caller. .	139
3.3.7.2	Two outside callers and one local extension	139

3.4	Making and receiving calls using VoIP	140
3.5	Using phones connected in parallel to TalkSwitch	141
3.6	Modems and telephone line access	142
3.7	Out of the Office — Receiving calls with call forwarding (analog and IP) ...	143
3.7.1	Three ways to forward calls	143
3.7.2	Transferring calls from a remote extension	143
3.7.3	Screening options for forwarded calls	144
3.7.4	Calls over VoIP with IP phones and Gateways	145
3.8	Using the voicemail system	146
3.8.1	Activating voice mailboxes	146
3.8.2	Retrieving messages and accessing a voice mailbox	147
3.8.3	Recording an announcement on a regular or an IP phone	149
3.8.4	Pager and cell phone notification	149
3.9	Music on hold	150
3.10	Mode switching options	151
3.11	Out of the Office — Making calls with call Back and call bridge	151
3.11.1	Using call bridge	152
3.11.2	Using call back	152

CHAPTER 4: CALL DETAIL RECORD (CDR) LOGGING

4.1	Enabling call detail record (CDR) logging	155
4.2	Retrieving data	156
4.2.1	Web interface — Store to File on TalkSwitch	156
4.3	Analyzing the data	158

CHAPTER 5: VOIP INFORMATION

5.1	Introduction to VoIP	161
5.2	Optimizing Your IP Network for VoIP	162
5.2.1	The broadband connection	162
5.2.2	The router/NAT/firewall	162
5.2.3	Connecting to a LAN and IP network	163
5.2.3.1	Confirm sufficient network capacity for VoIP	163
5.2.3.2	Confirm router/firewall path for voice data	164
5.3	Setting up a VoIP network	165
5.3.1	Connect TalkSwitch or SIP-compatible gateways	165
5.3.2	Select which TalkSwitch to use as the SIP network server	166
5.3.2.1	Which location and unit should be the SIP server?	166
5.3.3	Assign phone numbers to each VoIP location	167
5.4	VoIP Configuration	168
5.4.1	Configure TalkSwitch IP addresses	168
5.4.1.1	Set the TalkSwitch local IP address	168
5.4.1.2	Set the TalkSwitch public IP address	169
5.4.2	Configure TalkSwitch Profile	170

5.4.2.1	TalkSwitch Profile	170
5.4.2.2	Service Provider profile	175
5.4.3	VoIP Numbers	176
5.4.3.1	Assign VoIP phone numbers	177
5.4.3.2	Configure call handling for VoIP numbers	178
5.4.3.3	Assign service provider phone numbers	179
5.5	FAQ	180
5.6	VoIP network administration form	186

CHAPTER 6: TROUBLESHOOTING AND SUPPORT

6.1	Troubleshooting	187
6.1.1	Problems that may occur during configuration	187
6.1.2	Problems that may occur while using the TalkSwitch features	188
6.1.2.1	The auto attendant	188
6.1.2.2	Music on hold	189
6.1.2.3	Call routing with local extensions and home phones	190
6.1.2.4	Answering and fax machines	190
6.1.2.5	Other possible local extension problems	191
6.1.2.6	Multiple TalkSwitch units connected to the same LAN	192
6.2	Support	194

APPENDICES

Appendix A:	Functions and Commands	195
Appendix B:	TalkSwitch and Telephone Company Calling Services	201
Appendix C:	TalkSwitch and Power Interruptions	205
Appendix D:	Safety and Regulatory Information	207
Appendix E:	TalkSwitch One-Year Warranty and Return Policy	211
Appendix F:	Specifications	215
Appendix G:	Copyright and Licensing Notices	217
Appendix H:	Home/Office Wiring Guide	221

GLOSSARY	229
----------------	-----

INDEX	237
-------------	-----

PREFACE

Thank you for purchasing the TalkSwitch phone system. Please read this section before you start installing the product.

WHAT'S IN THIS GUIDE?

The *TalkSwitch User Guide* contains all the information you need, whether you are installing a single TalkSwitch unit or multiple VoIP enabled units. It is intended to be a complete reference accompanying the *TalkSwitch Start Guide* that ships with every TalkSwitch unit.

This preface contains important information to help you maximize your installation effort and to get the most out of the features and the flexibility of your TalkSwitch system. We strive to make your experience with installation and configuration the most rewarding possible.

We would ask you to read important information concerning power interruptions and safety precautions contained in the appendices in order to ensure your TalkSwitch equipment is set up in the safest way possible while avoiding equipment damage.

What you should know

While TalkSwitch is customer installable, certain skills are required if you need to route cables or to configure a network. The following points will help you determine the required skills:

- Configuring the TalkSwitch system using the configuration software can be performed by anyone with basic computer skills once the system is physically installed with proper networking equipment configurations (if two or more units are networked on a LAN).
- While most buildings are wired to accommodate TalkSwitch system setup, the need to route telephone and/or network cabling can occur occasionally. If your organization does not have someone with this skill set, we recommend the use of an outside telephony system technician.

- Connecting TalkSwitch to network equipment such as routers, switches or hubs with a connection to the Internet, as well as configuring firewalls, computers and TalkSwitch for networked use internally and with Internet. If your organization does not have someone with this IT skill set, we recommend the use of an outside IT technician.

If you are installing a single, non-VoIP, TalkSwitch unit, anyone with basic computer skills can use the *TalkSwitch Start Guide* and this user guide to perform a full installation and configuration.

TalkSwitch system installations can be categorized into some general configurations, such as single unit installation, networked units installation, VoIP or non-VoIP installation, etc. This section helps you determine your best possible plan of action using this guide, according to your installation type. While not every possible installation scenario is detailed below, determine which one better fits your situation and use its guidelines as a starting point. *Troubleshooting and Support* on page 187 provides solutions for common problems and the index at the back is a means to quickly navigate to specific areas.

Single unit installation

When installing a single, non-VoIP TalkSwitch unit, you can skip the sections marked with the Networked and VoIP icons.



Networked units installation

When installing networked TalkSwitch units, you require the sections marked with the Networked icon.



VoIP installation

When installing a VoIP enabled unit, you require the sections marked with the VoIP icon. *Chapter 5: starting on page 161* provides an additional resource.

Connecting devices

When connecting phones and other devices to your TalkSwitch unit(s), *1.9 Connecting devices* on page 15 provides the information you need.

FINDING THE INFORMATION YOU NEED

The following user guide functionality helps you find the information you need quickly and enables you to skip the information you don't need. This will make your experience with this user guide and your TalkSwitch system installation and configuration much more efficient.

Using the table of contents and the index

The table of contents at the front of this user guide contains all of the section headings and page numbers throughout the manual. The index at the back of the user guide contains a keyword reference with page numbers. If you are using this guide online in PDF format, the headings and keywords will contain hyperlinks, allowing you to quickly navigate to the sections you need.

Navigating with cross-references

Cross-references are contained throughout the user guide in order to help you access related information and illustrations quickly. As with the headings and keywords located in the table of contents and index, these cross-references contain hyperlinks, allowing you to quickly navigate to the sections you need.

Where to go for further information

The guides listed below and other guides can be found on the TalkSwitch software CD, in the TalkSwitch folder in the Windows Start menu once the software has been installed, and in the support section of our website at **<http://www.talkswitch.com/support>**.

- For basic installation and configuration instructions, refer to the *TalkSwitch Start Guide* that ships with every TalkSwitch unit.
- For information on setting up a TalkSwitch system to use VoIP, refer to the *TalkSwitch VoIP Network Configuration Guide*.

GUIDE CONVENTIONS

The *TalkSwitch User Guide* uses the following text elements and icons as visual aids, making the manual more accessible.

Text elements

<i>Italic</i>	Italicized text highlights configuration software fields located on the various software windows, as well as references to other sections of this guide or to other TalkSwitch documents.
Bold	Bolded text highlights configuration software menu selections located on the software top menu drop-down lists and on the left navigation section. Non-numbered paragraph headings are also bolded.
<i>“Italic”</i>	Italicized text in double quotes highlights TalkSwitch system voice prompts you hear from a telephone.
<Italic>	Italicized text in brackets highlights text you are asked to type.

Icons

	The Networked icon is used to mark sections with information pertaining to a setup where two or more TalkSwitch units are networked together on a LAN.
	The VoIP icon is used to mark sections with information pertaining to a setup where VoIP is being used.
	The IP extension icon is used to mark sections with information pertaining to the set-up and use of IP phones as local extensions.
	The Alert icon is used to mark sections that are important. They may contain warnings or cautions alerting the user to pay special attention.

CHAPTER 1

TALKSWITCH INSTALLATION

1.1 TALKSWITCH PACKAGE CONTENTS

TalkSwitch units ship with everything you need to get started. A power adapter, cables, a software CD and documentation comes with every unit. Refer to the *Start Guide* included with the unit to verify you have everything you need. For users with VoIP enabled units, the package also contains a *VoIP Network Configuration Guide*.

If any of these items are missing, please contact your TalkSwitch dealer.



Warning: If TalkSwitch has been exposed to low temperatures prior to installation, wait until the system has reached room temperature before connecting the power cord to avoid condensation.

1.2 CONFIGURATION SOFTWARE SYSTEM REQUIREMENTS

- PC running Windows XP/2000 or Vista (32 bit)
- 160 MB free hard disk space
- 256 MB RAM
- USB or Ethernet port
- Minimum 800 x 600 minimum video resolution

1.3 UNIT FRONT PANEL

The front panel consists of a power button and five lights indicating the usage of the incoming telephone lines with different states of illumination.

Figure 1: Front panel lights



Light	State	Description
Line 1/Line 2	on	Line 1/Line 2 is currently in use.
	flickering	Line 1/Line 2 is ringing.
	pulsing slowly	Line 1/Line 2 caller is on hold.
	quick pulse	Line 1/Line 2 is engaged by a device that is sharing the line with TalkSwitch.
Power/Data	on	TalkSwitch is powered on.
	flickering	The PC connected (USB) to TalkSwitch is either sending or retrieving information from TalkSwitch.
	pulsing slowly	Global Message Waiting Indicator (configurable).
Line 3/Line 4	on	Line 3/Line 4 is currently in use.
	pulsing slowly	Line 3/Line 4 caller is on hold.
	flickering	Line 3/Line 4 is ringing.
	quick pulse	Line 3/Line 4 is engaged by a device that is sharing the line with TalkSwitch.

Line 3 and Line 4 lights apply to TalkSwitch models with four incoming telephone lines only. For more details on line light error codes, see 1.14 *What the flashing lights mean* on page 28.

1.4 UNIT BACK PANEL

Figure 2: TalkSwitch 240vs and 244vs



Figure 3: TalkSwitch 280vs and 284vs



Figure 4: TalkSwitch 480vs and 484vs



The PF box between E4 and L1/L2 represents power failure support. In the event of a power failure or loss of power to TalkSwitch, extension port E4 is able to receive and make calls on Line 1.

Ports	Purpose
MUSIC	The music connector is a 1/8" (3.5mm) phono jack used as an audio input for the music on hold feature. Mono cables are recommended. For more information, see 1.9.3 Connecting devices to the music jack on page 21.
PA	The PA (public address) connector is a 1/8" (3.5mm) phono jack used as an audio output for the external paging feature. Mono cables are recommended. For more information, see 1.9.4 Connecting to the PA (public address) jack on page 21.

Ports	Purpose
LAN	The LAN connector is an RJ-45 port used for connection to a PC, connection of IP phones, connection of VoIP lines, and unit-unit networking. For more information, see <i>1.8.1 Ethernet connection</i> on page 10, <i>1.9.2.3 Connecting an IP phone</i> on page 18, <i>1.10 Networking TalkSwitch units on a LAN</i> on page 21 and <i>VoIP Information</i> on page 161. There are 3 green LEDs on the right of the LAN port. The top LED indicates network synchronization (connection), the middle LED indicates data Rx (receiving data) and the bottom LED indicates data Tx (transmitting data).
E1-E4 E5-E8	The extension connectors are RJ-11 ports used for connecting phones and other analog devices. For more information, see <i>1.9.2 Connecting local extension telephones and other devices</i> on page 16.
L1-L2 L3-L4	The line connectors are RJ-11 ports used for connecting incoming telephone lines. For more information, see <i>1.9.1 Connecting incoming telephone lines</i> on page 15. Plug in your RJ-11 telephone lines, beginning with L1. If you have two lines out of one phone jack, you will require a dual-to-two-single-line adapter or replace the two-line cord with two single-line cords. Note that L3 and L4 do not exist on units with only two incoming telephone line connectors.
USB	The USB connector is a standard USB port used for connecting to a PC for unit configuration. For more information, see <i>1.8.2 USB Connection</i> on page 11.
MEMORY SLOT	The memory card slot is a socket used with TalkSwitch memory cards. For more information, see <i>Installing a memory card</i> on page 26. (Not available in some markets)
POWER	The power connector is a receptacle rated at 16 VAC/1.5A for connecting the supplied AC adapter input line.

1.5 INSTALLING THE CONFIGURATION SOFTWARE

1.5.1 Installing the software for the first time

If your TalkSwitch is already connected with a USB cable, disconnect the unit before you install/upgrade the software. Reconnect the USB cable to TalkSwitch after the configuration software is installed.

1. Turn on your computer and insert the TalkSwitch CD into your CD drive. The install program starts automatically.
2. Click **NEXT** and follow the instructions.

If you prefer to use Windows Explorer or if the install process does not launch automatically:

3. Double-click the **My Computer** icon.
4. Double-click the **CD-ROM/DVD** drive.
5. Double-click the **Startscreen.exe** icon and follow the instructions. After you click the **Finish** button, you will see the TalkSwitch icon on your desktop. This means that you have successfully installed the configuration software.

The latest version of the TalkSwitch software can also be downloaded from the TalkSwitch website www.talkswitch.com/support.

1.5.2 Upgrading the TalkSwitch software and firmware

We are continually looking for ways to enhance your communications capabilities. When new features are added, we provide our users with immediate access to an update directly from Window's **Start** menu.

1. **Start -> Programs -> TalkSwitch 4.10 -> TalkSwitch Auto Update.**
2. If your configuration software is due for an update, a dialog box informs you of the status of your software version. See *Figure 3*. If you need an update, you have the option to download the update. The update file is stored in C:\Program Files\TalkSwitch\TalkSwitch Configuration 4.10.

3. After the update is downloaded, click **Yes** in the dialog box to install the update. See *Figure 3*. You can install the update later from the folder C:\Program Files\TalkSwitch\TalkSwitch Configuration 4.10.

The TalkSwitch configuration software must be closed in order to install the update.

Figure 5: Check for updates

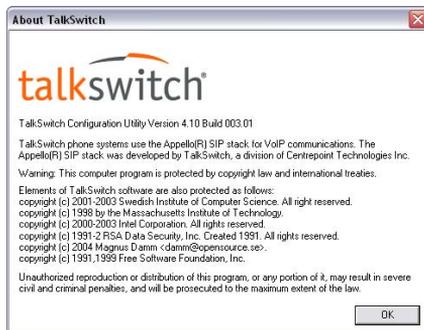


Alternatively, you can check the versions you are running once the system is connected by following the three steps below.

Step 1 — Check current version

To identify the appropriate update file, you must determine your TalkSwitch software and firmware versions. To find your TalkSwitch software version number, click **Help -> About TalkSwitch Configuration Utility...**

Figure 6: About TalkSwitch software version



Write down the software number. Go to the TalkSwitch website at **www.talkswitch.com/support**. The instructions help you select and download the appropriate upgrade.

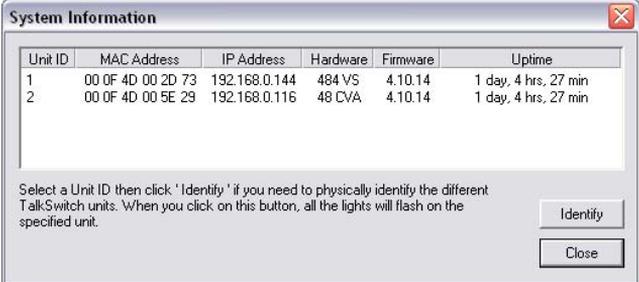
The instructions on our website take precedence over any instructions in this user guide.

The TalkSwitch firmware version number of each unit is listed in the *System Information* window.

1. Open the TalkSwitch configuration software if not already open. Click **About TalkSwitch**.
2. Click *View System Information*.

Networked units have to run the same firmware version to operate properly on the same LAN. Use this window to confirm that all the units have matching firmware versions. See *Figure 7*.

Figure 7: TalkSwitch firmware version



The screenshot shows a window titled "System Information" with a table containing the following data:

Unit ID	MAC Address	IP Address	Hardware	Firmware	Uptime
1	00 DF 4D 00 2D 73	192.168.0.144	484 VS	4.10.14	1 day, 4 hrs, 27 min
2	00 DF 4D 00 5E 29	192.168.0.116	48 CVA	4.10.14	1 day, 4 hrs, 27 min

Below the table, there is a text box with instructions: "Select a Unit ID then click 'Identify' if you need to physically identify the different TalkSwitch units. When you click on this button, all the lights will flash on the specified unit." There are two buttons: "Identify" and "Close".

Step 2 — Download new software and firmware

When you double-click on the appropriate version for downloading, a dialog box displays, asks you to enter a file name and select a folder for the download. We suggest leaving the file name as it is (e.g. TSC410006.exe) and saving it to your PC in C:\Program Files\TalkSwitch\TalkSwitch Configuration 4.10. Once the download is complete, begin your upgrade by double-clicking on the install file or using the *Run* option on your Windows *Start* menu. A series of windows will guide you through the installation process.

This process updates your configuration software to a newer version. You still need to update the firmware to take advantage of new features. The new firmware file is placed in the TalkSwitch directory after running the install program.

Step 3 — Updating the firmware

The last step is to update the firmware. Make sure your TalkSwitch is connected to your PC.

1. Open the configuration software.
2. **File -> Update Firmware.**

The update time varies depending on the connection type, how many files are needed and how many units are updated. On average the process takes between 1 and 5 minutes. The LED lights on the TalkSwitch front panel show diagnostic indicators for the update. See *1.14 What the flashing lights mean* on page 28. When the update is completed, you are prompted to reboot TalkSwitch. In the dialog box, click the **Proceed** button.

Should the firmware update fail, please repeat the update process above. If the problem persists, contact your reseller or TalkSwitch.

Most firmware updates do not alter your system configuration. In certain circumstances, during a major system update, the configuration file may get reset to default values.

When you configure a unit for the first time and every time you modify the configuration, we recommend that you save your configuration file to your PC and preferably to removable media (3.5" Floppy, CD-ROM disc or Flash stick) as a back-up. This avoids having to reconfigure the entire system in case of a hardware failure.

The *LastSavedConfig4.10* in C:\Program Files\TalkSwitch\TalkSwitch Configuration\ is accessed when you save and retrieve the configuration software. For more information, see *2.1.1.1 File Menu* on page 32.

If you have to replace your computer, please download the latest install file from our website and follow steps 2 and 3 above. Run the install program. In case of a major system update, copy the back-up of your configuration file to C:\Program Files\TalkSwitch\TalkSwitch Configuration.

1.6 INITIAL CONFIGURATION

Once your software is installed and your TalkSwitch is connected directly to a PC or via a switch/hub to the LAN, you can customize the TalkSwitch system with the TalkSwitch configuration software.

If you connect TalkSwitch to your phone system, it will do the following:

- An incoming call on any line will ring all analog local extensions.

- A receptionist can manually transfer calls to local extensions or voicemail.
- Every local extension will have a voice mailbox.

Recommendation: To minimize the disruption time of your incoming telephone lines during the system setup, we recommend that you configure the unit before you connect all your phones and incoming telephone lines to TalkSwitch. For more information, see *Chapter 2*: starting on page 31.

1.7 SELECT LOCATION FOR INSTALLATION

Select a suitable location for installation of TalkSwitch.

TalkSwitch can be installed on a desktop or mounted on a wall. It can be installed in a office or communications or utility room.

Choose a location that meets the following requirements:

- Ambient temperature between 0°C-35°C
- Away from heat sources, and direct sun light
- Avoid rooms with excessive humidity or dust
- Within 1.5m of a power socket.
- Close to telephone company demarcation point for telephone lines, or sockets, and any in-building telephone wiring connection point.
- Away from any electronic devices that emit radio signals.
- Away from hazardous or explosive materials.

When installing multiple TalkSwitch units, do not stack units on top of each other.

If TalkSwitch has been exposed to low temperatures prior to installation, wait until the system has reached room temperature before connecting the power cord to avoid damage due to condensation.

Recommendation: Use only the power adaptor and power cord supplied with your TalkSwitch. Check the supply voltage matches the rating on the power adaptor label. If your supply voltage does not match the power adaptor rating, please consult your reseller.
For safe operation, the TalkSwitch power adaptor must be connected to a power supply socket with a third pin (ground /earth).

Recommendation: Surge protection devices are recommended to protect TalkSwitch in areas subject to lightning or power surges. Please consult your reseller for recommended surge protection devices.

Wiring

Make the most of your phone system with optimal home/office wiring. If you need help wiring your house or small office and are not sure where all the jacks are or what colors of wires to use, refer to *Appendix H: Home/Office Wiring Guide* on page 221.

1.8 CONNECT TALKSWITCH TO A NETWORK OR A PC

Connect the AC adapter provided with TalkSwitch from an electrical outlet to the TalkSwitch unit. Never use a power adapter other than the one that came in the TalkSwitch package.

To configure the TalkSwitch settings, connect the unit to the PC with the configuration software installed.

There are four connection methods.

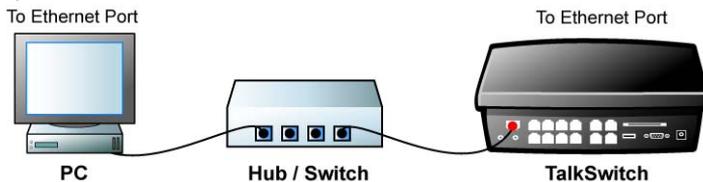
1.8.1 Ethernet connection



An Ethernet cable is supplied with specific unit models. This connection requires a LAN and a PC on the same switch. Any PC on the LAN can configure the unit.

Sharing computer data and TalkSwitch voice data, on the same hub can cause voice degradation during high traffic periods. For best results, place TalkSwitch units on a switch, or on their own hub.

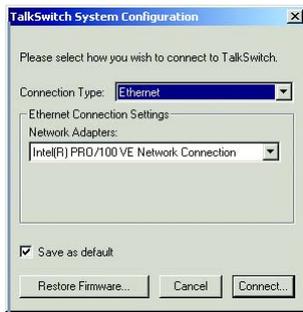
Figure 8: LAN connection



1. Connect TalkSwitch to your hub or switch using the supplied Ethernet cable and connect your PC to the hub or switch using another standard Ethernet (CAT 5) cable with RJ45 connectors.

2. If not already turned on, press the **Power** button on the front of your TalkSwitch and turn on your computer.
3. Double-click the TalkSwitch icon on your desktop to open the configuration software. If the software was unable to detect your TalkSwitch automatically, a dialog box opens requesting you to select the connection type.
4. Select *Ethernet* and the adapter you wish to use from the drop-down list and click **Connect...**

Figure 9: Ethernet connection



5. Once connection has been established, the *TalkSwitch System Configuration* window opens. See *Figure 29* on page 31.

A PC can connect directly to the TalkSwitch LAN port using a 'cross-over' cable equipped with RJ-45 connectors.

If you encounter difficulties opening the TalkSwitch configuration software, check that all your wires and plugs are securely connected.

1.8.2 USB Connection

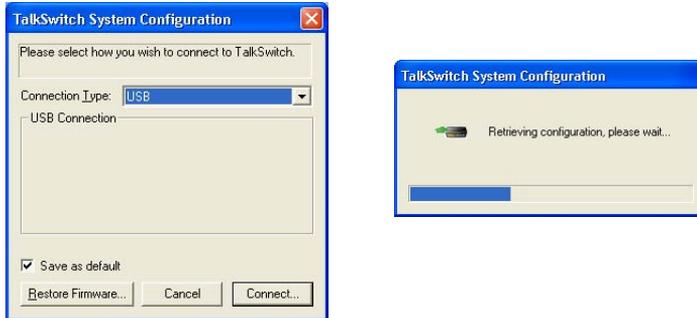


A USB cable is supplied with specific models. The PC and TalkSwitch unit must be in close proximity.

1. Connect the cable to a USB port on your PC and to the USB port on the back of TalkSwitch.
2. If not already turned on, press the **Power** button on the front of your TalkSwitch and turn on your computer.

3. Double-click the TalkSwitch icon on your desktop to open the configuration software. The dialog box in *Figure 8* is displayed.
4. Select *USB* from the drop-down list and click **Connect...** A progress bar indicates that the configuration is being retrieved. When this process is completed, the *TalkSwitch System Configuration* window opens. See *Figure 29* on page 31

Figure 10: USB connection

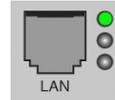


The computer connection is only needed to change the configuration settings. It can remain disconnected at other times.

1.8.3 Internet connection

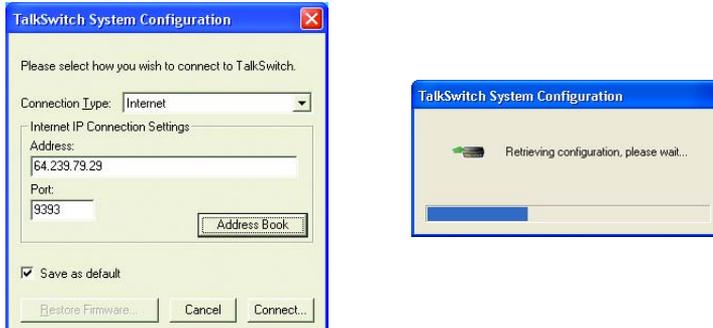
Select *Internet* from the drop-down list and click **Connect...** A progress bar indicates that the configuration is being retrieved. When this process is completed, the *TalkSwitch System Configuration* window opens.

See *Figure 29* on page 31. Ensure your unit is connected to a LAN with the supplied Ethernet (CAT 5) cable. The top LED on the the back panel beside the LAN port indicates that a connection has been established with a switch or hub. To support remote configuration over IP, map the TCP 9393 port from your firewall to TalkSwitch. Please refer to the manual for your router/ firewall to activate port forwarding.



1. Double-click the TalkSwitch icon on your desktop to open the configuration software. The dialog box below is displayed.
2. From the drop-down list in the dialog box, select *Internet* as the connection type.

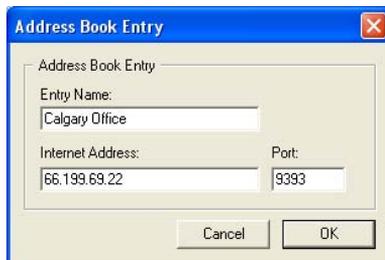
Figure 11: Internet connection



3. In the *Address* field of the *Internet IP Connection Settings* section, enter the public IP address or the FQDN of the TalkSwitch location and click **Connect....** A progress bar indicates that the configuration is being retrieved. When this process is completed, the TalkSwitch system configuration window opens. See *Figure 29* on page 31.

You have the option to store several locations in the address book.

Figure 12: Address book



1. Click the **Address Book** button.
 - a) To add a new address, click **New**.
 - b) Type the name of the location into the *Entry Name* field.
 - c) Type the Internet address of the location into the *Internet Address* field and click **OK**.
 - d) To edit or delete an existing entry, highlight the item you wish to edit or delete and click **Edit** or **Delete**.

1.8.4 File connection

You can load the latest configuration file saved on a PC to make edits and save it again as a file or save it directly to TalkSwitch.

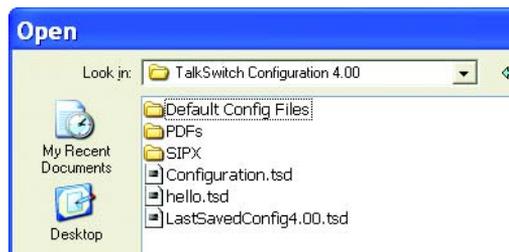
1. Double-click the TalkSwitch icon on your desktop to open the configuration software. The dialog box below displays.
2. From the drop-down list next to *Connection Type*, select *File*.

Figure 13: File connection



3. Click the **Browse** button and select the configuration file you wish to open and click **Open**.

Figure 14: Open TalkSwitch configuration file



4. Click **Connect...**
5. The TalkSwitch System Configuration window opens. See *Figure 29* on page 31.

When a unit is being configured, it is locked to prevent other computers or persons using a phone from configuring changes at the same time. If you leave the software open for longer than one hour, the unit unlocks itself to allow configuration changes.

Restore Firmware (button)

This button is available as an alternate method to update firmware if an update attempt has failed. Click the Restore Firmware button, then select which firmware file to load onto the unit. The firmware update will then begin. Follow the instructions presented at each step. Progress will be displayed as the update proceeds.

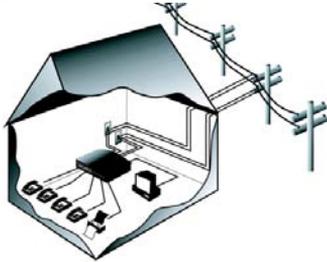
If you are downgrading to an earlier firmware version, and have more than one networked TalkSwitch unit, you must perform the Restore Firmware one unit at a time, with the unit separated from the network during the restore.

For more information, see *1.5.2 Upgrading the TalkSwitch software and firmware* on page 5.

1.9 CONNECTING DEVICES

Move the unit(s) to a location where it can be attached to incoming telephone lines and your telephone devices.

Figure 15: An example of residential wiring



1.9.1 Connecting incoming telephone lines

Advisory: In order to minimize disruption to your business, you may want to configure TalkSwitch before connecting it to your incoming telephone lines and extension phones.

1. Choose a convenient wall jack to attach to your TalkSwitch and connect your incoming telephone line(s).

2. Connect the TalkSwitch unit to the telephone company's line sockets using the dual-wire telephone cables provided. Take note of which incoming telephone line is connected to each line port. This information will be used in the configuration section.

- Connect the first incoming telephone line to L1.
- Connect the second incoming telephone line to L2.

If you have a unit with more than two incoming telephone lines, connect your RJ-11 telephone lines, beginning with L1. If you have two incoming telephone lines out of one phone jack, you will require a dual-to-two-single-line adapter or replace the two-line cord with two single-line cords. Note that L3 and L4 do not exist on units with only two incoming telephone line connectors.

All incoming calls will be routed to the extension phones by TalkSwitch. Call handling includes call forwarding, transferring, voicemail, etc. Incoming telephone lines are not directly connected to any of the local extensions. The software determines which incoming telephone lines are routed to which local extensions.

1.9.2 Connecting local extension telephones and other devices

Connect your single-line analog device to one of the unit extension jacks the same way you would plug it into a standard telephone wall jack.

You can connect any regular analog extension telephone set, cordless telephone or fax machine to the extension ports, E1-E4 (all unit models) and E5-E8 (for unit models with more than four extension jacks).

1. Plug each internal telephone into extensions beginning with E1.

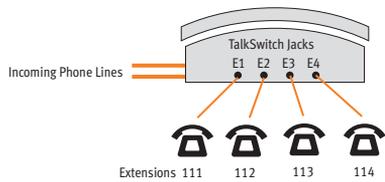
Connect a phone to E4 to ensure communication during a power failure. See *1.4 Unit back panel* on page 3 for more details.

2. If required, plug the fax line into E3 or E8.

1.9.2.1 Connecting a regular single-line telephone

This is the basic, simple set-up. All functionality including transferring calls, forwarding calls and voicemail, is handled by the unit. Even the most basic of phones, once connected, will inherit these functions.

Figure 16: Single-line telephone



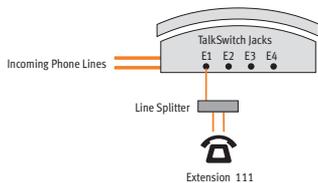
Each phone is connected to a single extension. A call transferred to extension 111 will ring the phone connected on jack E1. A call transferred to extension 112 will ring the phone on jack E2 and so on.

1.9.2.2 Connecting a regular dual-line telephone

Most dual-line phones have separate plug-ins for two incoming telephone lines (Line 1 and Line 2). If your dual-line telephone does not have separate plug-ins, you can use a Line 1/Line 2 splitter to separate the two lines as shown in *Figure 19* and *Figure 20* on page 18.

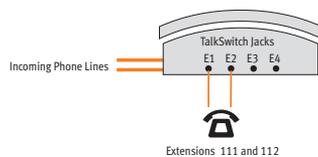
You can plug the dual-line phone into the TalkSwitch unit using one or two extension jacks. *Figure 17* below shows a dual-line phone using one jack and *Figure 18* shows a dual-line telephone plugged into the unit using two jacks.

Figure 17: Dual-line telephone on one jack



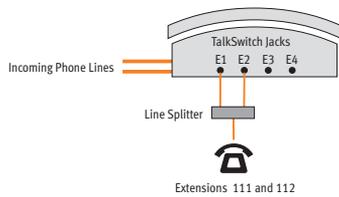
TalkSwitch extension 111 is connected to a dual-line phone using one extension jack. Calls on extension 111 show Line 1 as active.

Figure 18: Dual-line telephone on two jacks



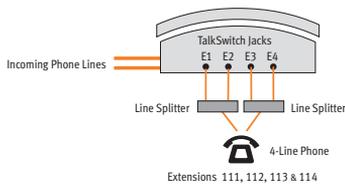
TalkSwitch extensions 111 and 112 are connected to a dual-line phone using two extension jacks. Calls on extension 111 show Line 1 as active. Calls on extension 112 show Line 2 as active.

Figure 19: Dual-line telephone on splitter



TalkSwitch extensions 111 and 112 are connected to a dual-line phone using a line splitter. Calls on extension 111 show Line 1 as active. Calls on extension 112 show Line 2 as active.

Figure 20: Four-line telephone on two splitters



Jacks 111, 112, 113 and 114 are all wired to the four-line phone. When a call is transferred to extension 111, Line 1 on the phone shows as active. Calls on extension 112 show Line 2 as active and so on. You may configure each incoming line to ring any given number of extensions. Configure incoming Line 1 to ring extension 111, Line 2 to ring extension 112, etc.

1.9.2.3 Connecting an IP phone

Plug the IP phone into your switch/hub using the supplied RJ-45 cable. Refer to the *IP Phone Configuration Guide* by clicking the link **Start -> TalkSwitch 4.10 -> Documentation -> IP Phone Guides -> Adding IP Phones to TalkSwitch**. If you are using a third-party IP phone, refer to the user guide of the third-party IP phone for configuration instructions.

1.9.2.4 Attach an internal or an external modem

Plug the modem's telephone cable into a unit extension jack. Your modem is now an extension that can access all lines and take advantage of call routing features.

Modems connected to a unit operate at a maximum of 28 kbps.

If you don't want to change your dial-up settings for the modem, enable **Direct Line Access** for the extension associated to the modem. For more information, see *2.2.7 Local extensions* on page 56.

1.9.2.5 Connecting fax machines

If you are connecting a fax machine to a unit with two or four incoming telephone lines, plug it into local extension E3 or E8.

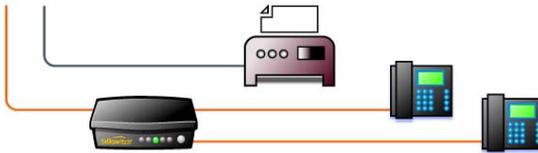
TalkSwitch provides you with three options to configure your fax machine. Choose the best option for your office.

Option 1 — Dedicated Fax Line

You may already have an incoming dedicated fax line. You can leave this line directly connected to your fax machine. Connect the remaining incoming telephone lines to your unit. Incoming telephone lines will be handled by the unit and your fax will work the way it has always worked.

Figure 21: Dedicated fax line

After installing TalkSwitch, the fax machine remains on its own dedicated line. All incoming telephone lines are shared by the remaining office phones.



Advantages: It's easy to set up and you don't need to change the way your fax handles calls.

Disadvantages: The dedicated fax line cannot be shared with the other phone devices (i.e. you cannot make an outbound voice call on your fax line). The dedicated fax line costs money, and may not be used as frequently as it could be.

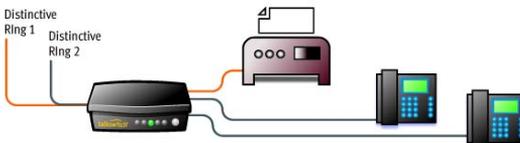
Option 2 — Distinctive Ring

Distinctive ring is a service provided by your phone company. A second phone number is added to a phone line. A call to either phone number will ring the same line with different ring patterns. You can use one ring pattern to indicate a fax.

Here's how the TalkSwitch handles incoming calls:

Figure 22: Distinctive ring

After installing TalkSwitch, it automatically detects the distinctive ring pattern. Faxes are automatically routed to the fax machine and calls to the phones.



Advantages: This option does not require a second physical line for faxes. More incoming telephone lines can be added as you grow. You can keep the same fax arrangement.

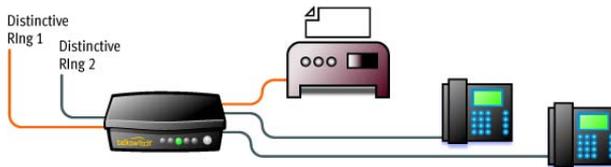
Disadvantages: Your phone company may charge a fee for the distinctive ring. If you choose this option, you will require the distinctive ring service from your telephone company.

Option 3 — Automatic fax detection via the auto attendant

The auto attendant can detect incoming fax calls and route them accordingly. To enable this, an auto attendant must answer all incoming calls with your pre-recorded message. It will listen for a fax tone to determine if the call is an incoming fax. If so, the call is routed to the fax machine, otherwise the call follows your **Call Handling** settings. You must create an auto attendant to use this method.

Figure 23: Automatic fax detection via auto attendant

All incoming calls are answered by the Auto Attendant.
The Auto Attendant then directs the call to the fax or local phone extensions.



Advantages: Calls are automatically handled for you. No distinctive ring service and no listening to ring patterns is required. Incoming telephone lines are shared between all phones and faxes, making better use of your resources.

Disadvantages: Some older fax machines do not emit a fax (CNG) tone. Therefore, the auto attendant cannot detect incoming faxes from those devices.

If your auto attendant volume is set too high or if your incoming telephone lines are noisy, the auto attendant may have difficulty detecting the incoming fax tone and therefore not route the call to the fax machine. You can solve this problem by adjusting the volume. For more information, see *2.6.2 Audio Controls* on page 111.

Options 2 and 3 require settings to be programmed into the unit using the configuration software. For more information, see *2.2.6 Fax information* on page 51 and *2.4.2.4 Automatic fax detection* on page 89.

1.9.3 Connecting devices to the music jack

The music jack is designed to support audio sources like a CD player, audio, tape player or other sound devices for playing music or messages to callers while they are on hold. Connect the audio source via its headphone output to the music jack.

The music jack requires a 1/8" (3.5 mm) mono phono connector. If you have more than one unit connected to a LAN, load a .wav file onto each unit to eliminate the need for a separate music source for each unit.

1.9.4 Connecting to the PA (public address) jack

The PA jack can be connected to a PA system for external paging or to an amplification system to screen voicemail or to use as a line simulator. The PA jack requires a 1/8" (3.5 mm) mono phono connector. If you have more than one unit connected to a LAN, you need to provide a connection onto the system to the PA system.



1.10 NETWORKING TALKSWITCH UNITS ON A LAN

Up to four units can be networked together over a LAN to increase the number of lines and extensions in your phone system.

1.10.1 Connecting TalkSwitch units to a LAN

You can plug your unit(s) into your existing LAN or operate it on its own LAN with an Ethernet switch. Ensure that the firmware on all units is updated before adding them to the LAN.

Check <http://www.talkswitch.com/support/> for the latest updates.

TalkSwitch unit enclosures are not designed for stacking. We recommend wall-mounting units to maximize airflow and keep the units from overheating.

1.10.1.1 Ethernet switch

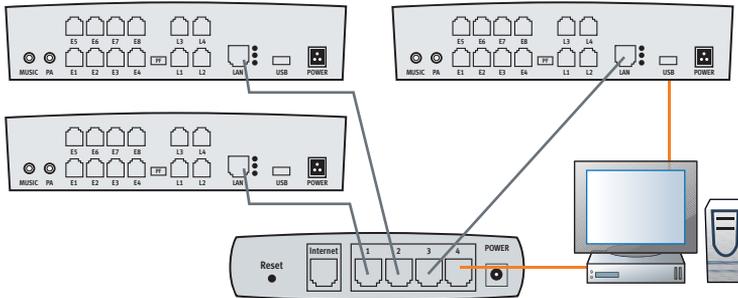
A switch provides direct communication between units, thus keeping the voice-over-LAN data isolated from other data on the network.

1. Connect up to four TalkSwitch units to the LAN switch.
2. Connect a computer to the phone system using the LAN connection or directly to the unit through the USB or LAN port.

3. Power up and configure your units before adding any IP phones that will be part of the system. It is important that you place the IP phones on the same subnet as TalkSwitch.

Example: If your TalkSwitch unit has the IP address 192.168.1.200, your IP phone should use IP addresses in the range 192.168.1.xxx that do not conflict with other IP addresses.

Figure 24: Three units connected via a switch



1.10.2 Power up all the TalkSwitch units

Connect the provided AC adapters to each unit and plug the adapters into available power outlets. Turn the units on by pressing the **Power** button on the front of each unit. The lights on the front panels of the units will flash. This means that they are powered up and connected to the LAN, but their ID numbers are not set.

1.10.3 Setting or changing the unit ID

Units are shipped from the manufacturer with unit ID 1. If two or more units are placed on the same LAN with the same unit ID number, it causes a conflict. All the line lights on the front panel of the units are flashing. You have to assign a different unit ID to each of the units.

To change the unit ID, the configuration software must be closed!

1. Pick up a telephone handset connected to one of the extension jacks on the unit and press **#** to enter command mode.
2. Enter a password if prompted.

3. Now, you have the following options:

Keys	Result
00 #	Responds with the unit ID number of that particular unit.
01 #	Sets the unit ID to 1.
02 #	Sets the unit ID to 2.
03 #	Sets the unit ID to 3.
04 #	Sets the unit ID to 4.

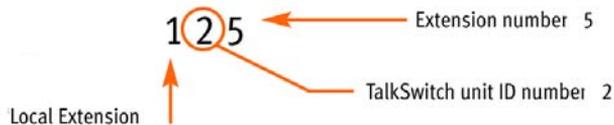
The system indicates that the update was successful and the front panel lights stop flashing after a few seconds. When the front panel *Data* light is on solid, the units are ready for network use.

No IP information is required to configure TalkSwitch since it does not use the TCP/IP protocol for voice traffic over the LAN. IP configuration is required for remote management, voicemail to e-mail notification or for VoIP.

1.10.4 How unit IDs affect system extension numbers

After successfully changing each unit ID to a unique number, it can be identified by the network. More importantly, each local extension (including additional IP extensions), remote extension, and voice mailbox has a unique numbering system. The second digit in the three digit identifier is the unit ID number, up to unit 4. See the following diagram and table.

Figure 25: Examples of three-digit identifiers



	Unit ID 1	Unit ID 2	Unit ID 3	Unit ID 4
Local analog or IP extensions	111–118	121–128	131–138	141–148
Additional IP extensions	151-158	161-168	171-178	181-188

	Unit ID 1	Unit ID 2	Unit ID 3	Unit ID 4
Remote extensions	210-219	220-229	230-239	240-249
Local mailboxes	111-118	121-128	131-138	141-148
Remote mailboxes	210-219	220-229	230-239	240-249
General mailboxes	410-419	420-429	430-439	440-449

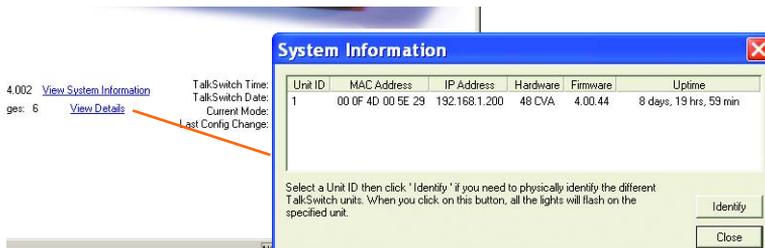
1.10.5 Keep track of the lines and extensions

To keep track of the extensions and incoming telephone lines connected to each unit, label each unit with its unit ID. It simplifies matters when you need to add or remove extensions and lines.

If the units are located in a room away from the extensions and you need to identify the units, the *View System Information* utility in the configuration software allows you to identify each unit as follows:

1. Open the TalkSwitch configuration software.
2. Click the *View System Information* link below the TalkSwitch image. See *Figure 26*.
3. Select the *Unit ID* to verify.
4. Click the **Identify** button and observe the flashing LEDs. The lights stop flashing when you click the **Close** button or when 5 minutes have elapsed.

Figure 26: System Information window



1.10.6 Optimizing the system for networked use

TalkSwitch units have been designed to operate optimally when in a networked state. Below are a few items that have been designed for better network use.

Configuration settings

All units are cloned with identical settings. If a unit has disappeared off the network (adapter unplugged, LAN disconnected, LAN failure, etc.), the system can still handle the calls since it retains the configuration settings of the missing unit. If an extension or a voice mailbox cannot be reached, the caller hears a system prompt: *"The extension you are trying to reach is currently unavailable, please try again later."*

Outgoing line hunt groups

When two or more units are on a LAN, the system tries placing calls out on the same unit the call originates from. This avoids using a line across the LAN on another unit, which helps to minimize the LAN traffic and optimizes the opportunity for all inbound calls to connect across the LAN if required.

Auto attendants

There are a total of nine auto attendants that are shared by all units on the LAN. When an auto attendant is recorded on any unit, it is automatically copied to all other units on the LAN. This design minimizes LAN traffic and provides functionality back-up in case a unit or units have disappeared off the network (adapter unplugged, LAN disconnected, LAN failure, etc.). In this event, all units can still answer inbound calls with the same auto attendant messages.

Voicemail

All local extension and remote extension voicemail data is stored on the unit where the extensions normally reside. For example, all greetings, directory names and voicemail messages for extensions 121-128, 220-229 and mailboxes 420-429 are stored on the unit with unit ID 2. If a unit is completely filled with voicemail messages, new messages will not be stored on other units. The system cannot accept anymore messages for mailboxes belonging to that unit. If you need more memory, TalkSwitch memory cards can be purchased from your reseller. To view the memory usage of each TalkSwitch unit and each mailbox, click **Tools -> Memory Usage -> Voicemail** or click **Tools -> Voicemail Manager**.

1.11 INCREASING MEMORY CAPACITY

There are two ways to increase memory capacity on a TalkSwitch unit. 1 and 2 hour memory expansion can be enabled by purchasing License Codes. 4.5 and 9 hour expansion is enabled using TalkSwitch Memory cards. Contact your reseller to purchase memory upgrades.

1.11.1 TalkSwitch memory cards

TalkSwitch memory cards are specifically formatted for use with TalkSwitch units. They allow you to add additional voicemail message capacity to your unit. Voicemail memory is used to store messages left in voice mailboxes. It is also used to store auto attendant greetings and announcements. TalkSwitch comes with 30 minutes built-in memory.

Installing a memory card

Figure 27: Memory card



Memory can be increased in increments of 1 hour, 2 hours, 4.5 hours and 9 hours.

1. Insert the memory card into the memory slot with the gold contact face down and the *This side up* tab showing. You don't have to turn the unit off or disconnect any of the telephones.

Figure 28: Memory card slot



1.11.2 Adding internal memory

It is possible to add additional internal memory capacity in 1 hour increments to a maximum of 2.5 hours using TalkSwitch optional features. See 2.7 *Licensed Add-ons* on page 120 for more details.

1.11.3 Verifying additional memory

To verify the addition of memory, use the *Summaries* section of the **Mailbox Data** window.

Summaries	
Unit ID	Total Unit Time
1	30 min. 2 sec.
2	30 min. 2 sec.
<hr/>	
Total	1 hr. 4 sec.

Summaries	
Unit ID	Total Unit Time
1	2 hr. 30 min. 13 sec.
2	30 min. 2 sec.
<hr/>	
Total	3 hr. 15 sec.

Unit 1 before
adding memory

Unit 1 after adding
2 hours of memory

1.12 UPGRADING TALKSWITCH UNITS

As your business grows, your TalkSwitch unit(s) may need upgrading. You have the option of upgrading your units to VoIP, as well as adding incoming telephone lines and extension ports.

You can purchase VoIP modules and upgrade boards at your reseller. Visit http://www.talkswitch.com/products/upgrades_accessories.asp on the TalkSwitch web site for more details.

1.13 ENABLING LICENSED ADD-ONS

It is possible to enable the following features on your TalkSwitch:

- additional IP extensions
- additional voicemail capacity
- Attendant Console software

For instructions on enabling TalkSwitch licensed add-ons, see *2.7 Licensed Add-ons* on page 120.

1.14 WHAT THE FLASHING LIGHTS MEAN

Diagnostics for the lights flashing on the unit front panel

All line lights flashing simultaneously:

The unit ID of this unit is in conflict with another unit on the same LAN. Make sure you assign a different unit ID to each unit. For more information, see *1.10.3 Setting or changing the unit ID* on page 22.

Line 2 and 3 lights flashing simultaneously:

This indicates a file system error. Terminate calls in progress. Reboot the unit. **Tools -> Reboot TalkSwitch** or, on the front panel, turn the **power** button off and back on. It will do a file system check and fix any problems.

Line 1 and 4 lights flashing simultaneously:

This indicates a voice mailbox error. Terminate calls in progress. Reboot TalkSwitch. **Tools -> Reboot TalkSwitch** or, on the front panel, turn the **power** button off and back on. It will do a file system check and fix any problems.

Line 1, 2, 3 and 4 lights flashing simultaneously:

The firmware needs to be updated or reloaded. Ensure that you update with the last version, do not use the version on the installation CD. There might be a newer version. Open the TalkSwitch configuration software.

File -> Update Firmware...

Line 1 and 4, then Line 2 and 3, lights flashing alternatively:

The system prompts are not loaded or corrupted. Reload the firmware. Open the TalkSwitch configuration software.

File -> Update Firmware...

Line 1 and 3 lights, then Line 2 and 4 lights flashing repeatedly:

This is usually the case after a firmware update. Reboot TalkSwitch.

Tools -> Reboot TalkSwitch or, on the front panel, turn the **power** button off and back on.

1.15 VERIFYING THE CONNECTIONS

If you are having problems, they could be due to loose cables. Check the following:

1. Are your telephone cables connected to a wall jack at one end and the unit at the other? Ensure the devices you are using as extensions (phones, faxes, etc.) are plugged into the extension jacks located on the back panel of the unit. Do not plug your extensions directly into a wall jack unless that jack is wired to a unit extension jack.
2. Check the USB or LAN connection between TalkSwitch and your PC.
3. Ensure that the unit AC adapter is plugged into a working power outlet and that the unit is turned on. Make sure that you are using the AC adapter supplied with your unit. This adapter cannot be substituted with any other adapter. If your adapter appears to be defective, please contact your TalkSwitch dealer to order the proper replacement.



If networked, ensure that you have the latest firmware loaded on all units. To check the firmware version, open the configuration software, and click the link *View System Information*. All units listed in the *System Information* dialog box should have the same firmware version.

CHAPTER 2

TALKSWITCH CONFIGURATION

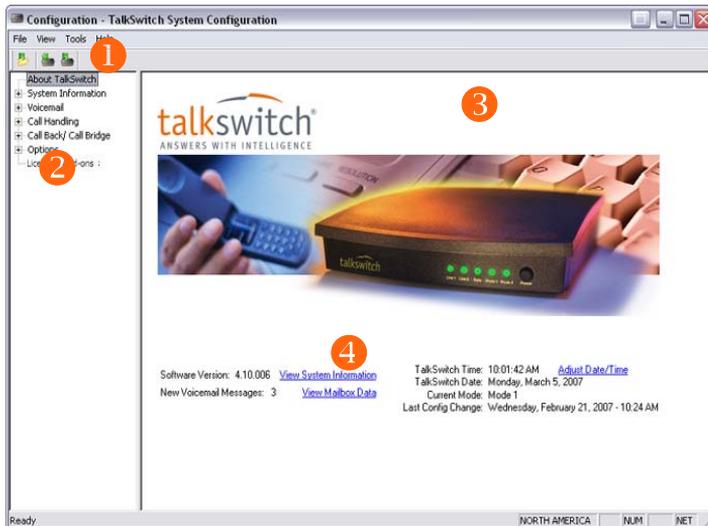
2.1 SYSTEM CONFIGURATION

This chapter contains detailed information about all the features in the TalkSwitch configuration software with step-by-step instructions on how to customize these features to best suit your needs.

2.1.1 The configuration screen

The configuration screen consists of the following parts:

Figure 29: Initial configuration screen



The numbers in *Figure 29* are explained below.

Context-sensitive help information is available for each option in the configuration software. Rest your mouse cursor over any of the controls and the information related to that control displays in a pop-up.

-
- 1 Menu items.

 - 2 Configuration navigation controlling the display in the configuration window.

 - 3 Configuration window displaying configuration information.

 - 4 *View System Information* is a link to the MAC address, IP address, hardware and firmware version of the unit.
-

2.1.1.1 File Menu

Open...	Opens the existing configuration files from the PC. TalkSwitch backs up the configuration to a file called <i>Configuration</i> every time you save the settings.
Save to TalkSwitch	Saves the current configuration to the TalkSwitch unit.
Save to File...	Saves the current configuration to a file on your PC.
Exit	Closes the configuration software.
Retrieve Settings	Retrieves the settings from the unit connected to the PC.
Update Firmware...	Updates the firmware. For more information, see <i>1.5.2 Upgrading the TalkSwitch software and firmware</i> on page 5.

2.1.1.2 View Menu

Toolbar	Toggles the toolbar on/off
Status Bar	Toggles the status bar on/off

2.1.1.3 Tools Menu

Memory Usage	Voicemail: Displays a dialog box showing internal memory usage for voicemail messages. Auto Attendant: Displays a dialog box showing internal memory usage for auto attendant messages.
Voicemail Manager	Displays a dialog box with options to check the voice mailbox data, delete a mailbox password and reset mailboxes to factory default.
Terminal Window	Displays a command line interface. This option is useful for troubleshooting in conjunction with technical support.
Call Logging Output	Store to file on TalkSwitch: TalkSwitch can store call detail records to a file stored on TalkSwitch. This file can be retrieved/viewed from a web browser.
Defaults...	Resets the current page or the entire configuration to defaults.
Reboot TalkSwitch	Prompts you to save the configuration and provides options when to reboot.
View Keys Installed	Lists your registration key(s) with the expiry date(s).
Check for updates	Checks the web site for configuration software updates.

2.1.1.4 Help Menu

About TalkSwitch Configuration Utility...	Displays the TalkSwitch software version number, firmware version and copyright information.
User Guides...	Opens the folder containing the TalkSwitch documentation.
Support on the Web...	Launches your browser and links you to our support site.

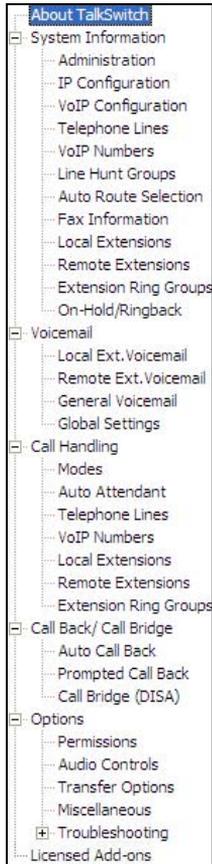
2.1.1.5 Configuration navigation

The configuration navigation organizes all the configuration topics within folders. The following is a brief description of each folder seen in the navigation section to the left of the window. See *Figure 2.1.2* on page 35.

1. Open the TalkSwitch configuration software.
2. Click the  sign beside each folder to expand it and display configuration option sub-categories. When you configure TalkSwitch or when you wish to modify the setup, it is helpful to be familiar with this structure.

About TalkSwitch	This link displays version information about the TalkSwitch software. It also displays new voicemail messages, the time and date and the current mode of operation.
System Information	Configure the system setup. From here you can activate incoming telephone lines and extensions.
Voicemail	Configure voice mailboxes and global settings for the voicemail system.
Call Handling	Configure how to handle incoming calls. It includes auto attendant and individual call handling setups for all phone numbers, extensions and modes of operation.
Call Back/ Call Bridge	Configure the TalkSwitch call back/call bridge features in this folder. Both of these features reduce your company's long-distance expenses.
Options	Configure audio controls, troubleshooting and various operation settings in this folder.
Licensed Add-ons	Permits users to activate additional IP extensions, activate additional memory, or enable Attendant Console.

Figure 30: Configuration navigation



The graphic to the left shows all the configuration screen options available in the TalkSwitch configuration software.

Note that **VoIP Configuration** and **VoIP Numbers** (in **System Configuration** and **Call Handling** categories) screens are not present for a unit or units without VoIP ports.



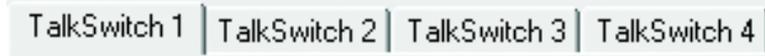
2.1.2 Special considerations when connecting multiple units to a LAN

Multiple units on a LAN will function as a single phone system. Some features are expanded as you add units (e.g. number of voice mailboxes), others are global in nature and are not expanded when you add more units (e.g. park orbits).

To determine the number of local extensions and mailboxes your TalkSwitch system supports, add the extension jacks (E1–E8) on all the units. In addition, each unit supports ten remote extensions and mailboxes.

Each unit is identified with a tab at the top of the configuration window, **System Information -> Telephone Lines** to allow you to select the unit in the group.

Figure 31: Unit tabs



The features described below are not affected by the number of units on the LAN.

Feature	Description
Auto Attendant	There are 9 auto attendants shared by all units on the LAN, so all inbound calls can be answered by the same auto attendant on an incoming telephone line. When a new auto attendant is recorded on a unit, it is automatically copied onto all the other units. (See 2.4.2 <i>Auto attendant</i> on page 83)
Call Back	There are 4 auto call back accounts per unit. To use an announced message for a particular auto call back account, you need to record the announced message on the same unit as the account. This means calling in on a line that is physically connected to the unit with the account, or using an extension connected to the unit. (See 2.5 <i>Call Back/Call Bridge</i> on page 102)
Ring Groups	There are 10 global ring groups (300-309) regardless of the number of units on a LAN. Each group is capable of ringing any extensions on the network. A combination of 32 analog and IP extensions can be part of a ring group. (See 2.2.9 <i>Extension ring groups</i> on page 64)
Park Orbits	There are 10 global park orbits (500-509). Any person can put a caller on hold in a temporary park orbit so that the call can be retrieved from any extension in the office.
Music on Hold	There are two ways to add music on hold to your system: <ul style="list-style-type: none"> • through an external audio source • by loading an internal music file (See 2.2.10 <i>On-Hold/Ringback</i> on page 66)
Public Address (PA)	Each unit has a PA output for connection to a public address system. The PA output for a TalkSwitch unit is only accessible from extensions connected to that unit. To connect all outputs to a single amplifier, you will require an adapter.

For more details on the Music on Hold and Public Address ports see [1.4 Unit back panel](#) on page 3.

2.2 SYSTEM INFORMATION

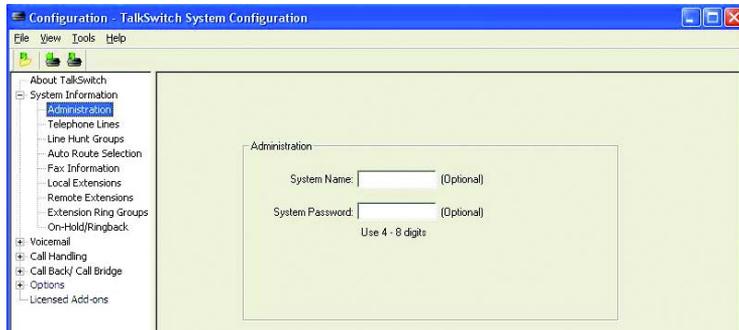
2.2.1 Administration

For security reasons, we recommend that you use password protection at the administrator level. The password is entered at configuration start-up and when you access the configuration through an extension phone.

To activate the use of a system password:

1. Click the **+** next to **System Information -> Administration**.

Figure 32: Username and password

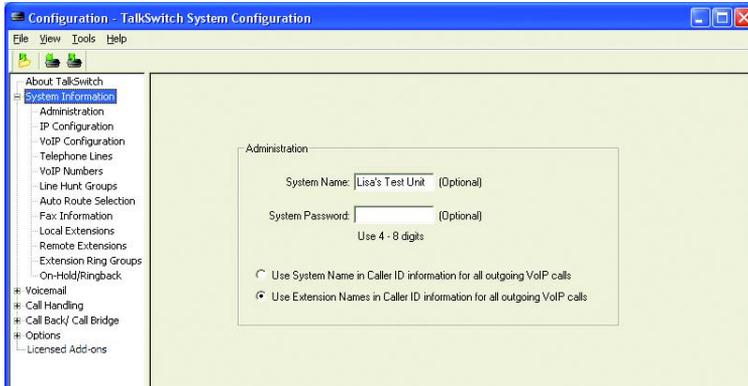


2. If you have units in different locations, we recommend that you assign a system name to each unit for tracking purposes. Enter a name in the *System Name* field. The system name will also be required when you are logging in through the web interface of the unit for status or Call Detail Record (CDR) information. For more information, see [Call Detail Record \(CDR\) Logging](#) on page 155.
3. Enter a system password in the *System Password* field. It has to be a 4- to 8-character numeric password, so that you can also enter it on a touchtone phone.

PBX Fraud and Misuse

Change the password frequently to prevent unauthorized users from making calls through your PBX.

Figure 33: Username and password for a VoIP enabled unit



Caution: If you leave the password field blank and you are connected to the Internet using TCP port 9393 mapped to the unit, the system will be vulnerable to configuration changes by anyone on the Internet!

To enter a new password:

1. Click in the *System Password* field.
2. Type a password and click **OK**.

To change the password:

1. Click in the *System Password* field.
2. Type a new password and click **OK**.

To delete a password:

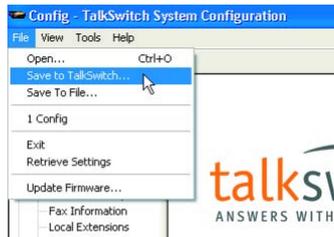
1. Click in the *System Password* field.
2. Do not enter anything and click **OK**.

Figure 34: Password dialog box



- After entering, changing or deleting a password, click **File -> Save settings to TalkSwitch** or click the middle icon in the tool bar to save the settings.

Figure 35: Save settings



New settings must be saved to the TalkSwitch phone system to take effect. A progress bar will show the data exchange to the TalkSwitch unit. It takes only a few seconds to complete.



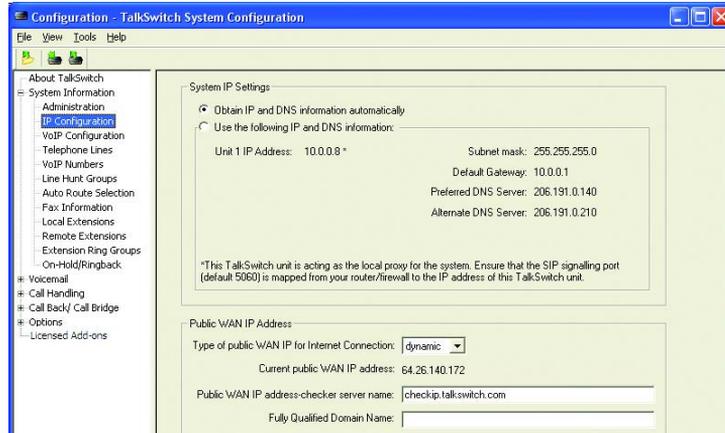
2.2.2 IP Configuration

System IP Settings section

The *Obtain IP and DNS information automatically* radio option is selected by default. This selection refers to DHCP (Dynamic Host Configuration Protocol). If you are on a DHCP server, the IP and DNS information will be filled in automatically.

- Click the **+** next to **System Information -> IP Configuration**.

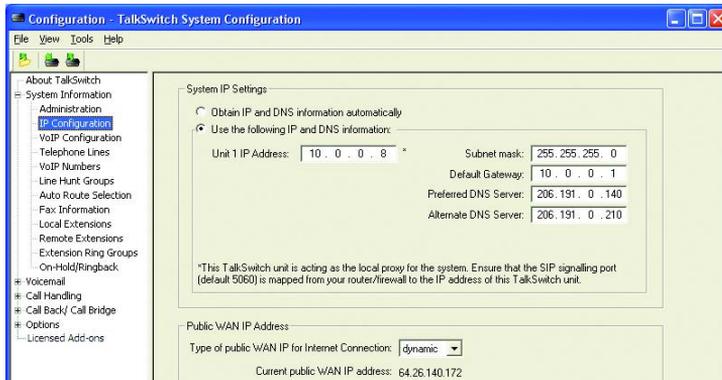
Figure 36: Automatic IP configuration



Additional VoIP configuration information is contained throughout this guide. For more information, see *5.4 VoIP Configuration* on page 168.

2. To configure the IP information manually, select *Use the following IP and DNS information*.

Figure 37: Manual IP configuration



If you do not have a DHCP server, the following fields need to be completed if you plan on using the VoIP capabilities.

Unit IP Address:

Assign an available static IP address to each unit (1-4). If you use a DHCP server, you have to reserve an IP address for each unit. To reserve an IP address, you need the MAC address of each unit. To view the unit MAC address, click the *View System Information* link on the **About TalkSwitch** configuration page. For more information, see *2.1.1 The configuration screen* on page 31.

Subnet Mask:

Enter the subnet mask for the LAN. This address determines what subnet the specific addresses belong to.

Default Gateway:

Enter the IP address of the gateway on your network. A gateway is a hardware device (i. e. router/NAT) that connects the office network to the Internet. The gateway allows you to share a DSL, cable modem or other Internet connections with all of the computers and IP devices in your office network.

Preferred DNS Server:

Enter the IP address of the preferred/primary DNS server. DNS is a service that is used to resolve a domain name to an IP address.

Example: A DNS server is used by your web browser to resolve www.talkswitch.com to an IP address such as 64.26.166.52. For more information, see *VoIP Information* on page 161.

Alternate DNS Server:

Enter the alternate/secondary DNS servers IP address if applicable.



Public WAN IP Address

Type of public WAN IP address for Internet Connection:

This field is set to *dynamic* by default. The unit checks the public IP address every 5 minutes. If it changes, it automatically updates to the correct information in order to manage VoIP calls properly.

The *dynamic* setting will not work in closed network environments. In a closed network environment, change the field to *static* and enter the appropriate IP address. If the unit is not behind a router, enter the IP address of the unit acting as the local proxy.

Current public WAN IP address:

For the *dynamic* setting, this entry shows the *current public IP address* and is not editable. For the *static* setting, enter the IP address or the FQDN of your location.

Public WAN IP address-checker server name:

For the public IP address type *dynamic*, this entry shows the current server name for the IP checker utility. If the public IP address type is *static*, no information is required.

The default location is checkip.talkswitch.com. An IP checker application is running on the server that responds to requests from units for the public IP address at its location.

Fully Qualified Domain Name:

A Fully Qualified Domain Name (FQDN) is required if this location acts as the SIP server and does not have a static public IP address from your service provider. You can obtain FQDNs for free at www.dyndns.org. If your router supports DDNS, program the required information to update the DNS servers.

If not, download one of the applications specified on the site. It needs to run on a PC connected to the same LAN to update the DNS servers. If the unit is enabled as a SIP server and no FQDN was entered, the prompt below displays.

Figure 38: Fully Qualified Domain Name (FQDN)

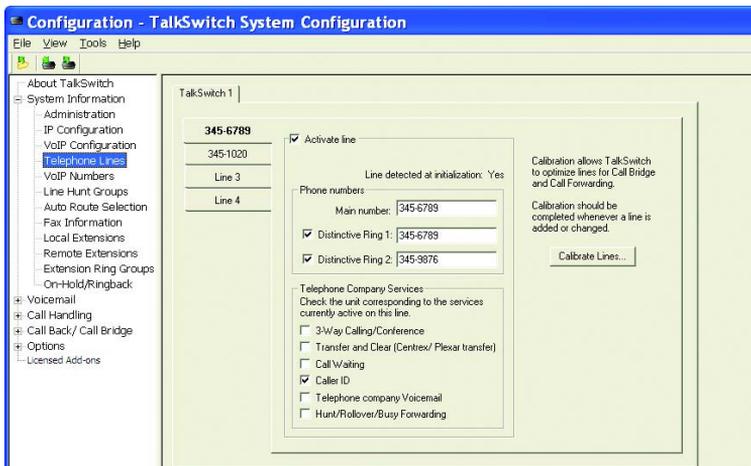


2.2.3 Analog incoming telephone lines

To configure incoming telephone lines go to the **System Information->Telephone Lines** folder. Select the tab for the telephone line to be configured. *Line Detected* shows if the incoming telephone line was detected when the system configuration was last retrieved. To update the detected status select File and Retrieve Settings. from the menu. Save any changed settings before retrieving the systems current settings.

1. Open the configuration software, if not already open.
2. Click **System Information -> Telephone Lines**.

Figure 39: Telephone lines



In the *Phone numbers* section

Enter the main telephone number for the incoming telephone line connected to TalkSwitch. Enable the *Distinctive Ring 1* and/or *Distinctive Ring 2* if you have this service on the current incoming telephone line and wish to have incoming calls handled differently on the basis of the distinctive ring. If you want distinctive ring numbers handled the same way as your *Main* number, do not place a checkmark in the box.

If you checked the box, enter the distinctive ring telephone number for each of the distinctive ring numbers associated with this incoming telephone line.

- *Distinctive Ring 1* corresponds to any double ring pattern.
 - *Distinctive Ring 2* corresponds to any triple ring pattern.
1. Click the *Activate line* checkbox to enable the incoming telephone line port.
 2. Enter the normal telephone number for the incoming telephone line as the *Main number*.

In the *Telephone Company Services* section

Check the boxes that correspond to telephone company services that are on this incoming telephone line. Descriptions of these services and how your unit deals with them are provided in *Appendix B: TalkSwitch and Telephone Company Calling Services* on page 201.

Calibrate Lines (button):

Click this button whenever a line is added or changed. Calibrate Lines allows TalkSwitch to optimize lines for call bridge and call forwarding. Only incoming telephone lines that are detected get calibrated.

Click this button whenever a line is connected to equipment other than a telephone company line, or if poor audio performance is encountered on the line. Calibrate Lines allows TalkSwitch to optimize transmission performance for the type of line or line equipment. The system's defaults are set for local telephone company lines, where the model of TalkSwitch is intended to be used.

For information about assigning and configuring the VoIP numbers in a TalkSwitch or a service provider setting, see 5.4.3 VoIP Numbers on page 168 and its subsections.

Reversal Signaling:

TalkSwitch supports Line Reversal options. Countries or regions utilizing this supervisory signal will have the reversal signaling field on this screen and have the ability to turn reversal detection on or off, as well as selecting one of the four following options:

- *Ignore*
- Reversals occur on *Seizure and Release*
- Reversals occur on *Seizure and Answer*
- Reversals occur on *Answer and Release*

Note that the default options for the specific country or region will be pre-selected in the configuration software.



For information about assigning and configuring the VoIP numbers in a TalkSwitch or a servicer provider setting, see 5.4.3 *VoIP Numbers* on page 176 and its subsections.

2.2.4 Line hunt groups

There are a total of 9 line hunt groups that control outbound call line selections. These line hunt groups are used by local and remote extensions, and by the call bridge feature. Each hunt group can support several line choices and will attempt to use an available line in the order that they are listed for that hunt group. Hunt group settings have no effect on incoming calls.

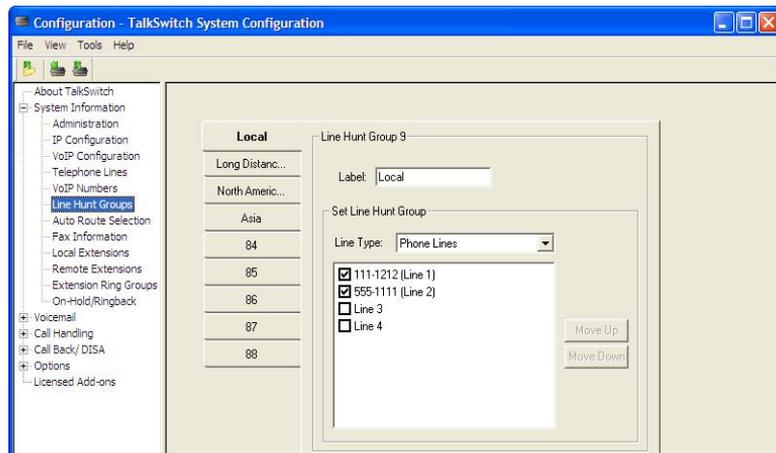
Default setup:

Hunt Group 9:	selects any available telephone line.
Hunt Group 81:	selects telephone Line 1.
Hunt Group 82:	selects telephone Line 2.
Hunt Group 83:	selects telephone Line 3 (four-line models only).
Hunt Group 84:	selects telephone Line 4 (four-line models only).
Hunt Group 85 - 87:	no lines assigned (by default)
Hunt Group 88:	selects any available VoIP number (only applies when a VoIP unit is present on the LAN).

1. Select **System Information** -> **Line Hunt Groups**.
2. Select a *Line Hunt Group* tab and enter a unique name in the *Label* field to identify this group elsewhere in the configuration software.
3. Under *Set Line Hunt Group*, set the *Line Type*. If you wish to use this hunt group to access telephone lines, select *Phone Lines*. To use this hunt group to access VoIP numbers, select *TalkSwitch VoIP Numbers*, or *Service Provider VoIP Numbers*, optionally.

4. If the hunt group uses phone lines:
 - a) Click the checkbox(es) next to the line(s) you wish to add to the hunt group.
 - b) The order of Line use can be adjusted to suit your needs. By default, TalkSwitch will first search higher line numbers (e.g. Line 4) for outgoing calls, since lower Line numbers may be more heavily used for incoming calls. The sequence can be adjusted by selecting each Line in the *Set Line Hunt Group* list, and clicking the *Move Up* or *Move Down* buttons on the right side of the window.

Figure 40: Hunt group configuration



2.2.5 Automatic Route Selection and Toll Restriction

What is Automatic Route Selection (ARS)?

ARS directs selected outgoing calls to specific telephone lines, alternative carriers, or VoIP trunks, for least cost routing.

What is Toll Restriction?

Toll Restriction blocks selected outgoing calls.

The ARS and Toll Restriction features offer extensive flexibility to control the routing and blocking of calls.



If multiple units are connected to the LAN, a checkbox will appear for the *For all Hunt Groups, hunt lines on same unit first* feature. This feature ensures that the TalkSwitch system will first use available lines on the unit where the extension is connected, to minimize network traffic between units. This feature is set automatically, but can be disabled if necessary.

Example: ARS could be used to route international calls to a VoIP service provider; national fixed line calls can be routed to an alternative carrier offering a discounted calling plan, using a carrier selection code; and calls to cellular phones could be routed to telephone lines attached to fixed wireless terminals for cellular-to-cellular calling. Toll Restriction could be used to block calls to premium rate services.

ARS and Toll Restriction uses the first dialed digits (referred to as 'leading digits') of the dialed telephone number to determine how the call will be handled. Toll Restriction can block calls based on the leading digits dialed. ARS can redirect calls to different Line Hunt Groups, based on the leading digits dialed, and has the option of adding carrier selection prefix codes to outgoing calls. ARS routes calls to Line Hunt Groups, which can be configured to direct calls to separate groups of telephone lines or VoIP trunks.

Planning Automatic Route Selection and Toll Restriction

To use ARS and Toll Restriction:

- Identify the leading digits of the dialed numbers for the calls to be redirected by ARS or blocked by Toll Restriction.
- Identify which telephone lines and VoIP trunks are connected to different service providers.
- Assign telephone lines and VoIP trunks to Line Hunt Groups, to provide separate access to each different service provider (see section 2.2.4).
- Identify any carrier selection codes and time of day when they will be used (if required).
- Identify which extension's calls will be controlled by ARS or blocked by Toll Restriction.

Ensure that the line hunt groups are configured before using ARS. For more information, see *2.2.4 Line hunt groups* on page 44.

The actual leading digits used in each country are subject to the national numbering plan. Your telephone directory will provide information on the numbers used.

Entering leading digits and assigning actions

What are leading digits?

Leading digits are the first digit or digits dialed in a call. They are the digits used to decide if a call is to be redirected or blocked. The leading digits could be international access code, country code, national code, STD code, area code, leading digits of premium rate service codes, or phone number.

- Leading digits can be 1–11 characters in length.
- Leading digits can include numeric characters 0–9 and * (wildcard).
- The leading digits exclude the Line Hunt Group access code.

To redirect or block calls based on the leading digits:

1. Open the TalkSwitch Configuration program.
2. Select **System Information -> Auto Route Selection**.
3. Enter the leading digits in the *Leading Digits* field.
4. Select the desired *Action* (route calls to a specified Line Hunt Group or block calls) from the drop-down list.

Figure 41: ARS — leading digit table

Activate Automatic Route Selection

TalkSwitch can route or block outgoing calls based on the digits dialed.
Enter up to eleven digits per field and select the action required (i.e. route to a specified Line Hunt Group, or block the call).

ARS Hunt Groups...

Leading Digits	Action	Carrier Code 1	All Day	Start Time 1	Carrier Code 2	Start Time 2
			<input type="checkbox"/>	9:00 AM		5:00 PM
1	long-distance ...		<input type="checkbox"/>	9:00 AM		5:00 PM
01161	Australia (82)		<input type="checkbox"/>	9:00 AM		5:00 PM
01	Block Calls		<input type="checkbox"/>	9:00 AM		5:00 PM
			<input type="checkbox"/>	9:00 AM		5:00 PM
			<input type="checkbox"/>	9:00 AM		5:00 PM
			<input type="checkbox"/>	9:00 AM		5:00 PM

Calls with leading digits that do not match the entries in the ARS table will be routed to the Line Hunt Group originally dialed by the caller.

Entering Carrier codes and start times.

What is a carrier code?

A carrier code is a carrier selection prefix that precedes the normally dialed digits to instruct the telephone line service provider that the call is to be routed to an alternative carrier.

To use a carrier selection code:

1. Enter the carrier code in *Carrier Code 1* corresponding to the leading digit entry.
2. If a second alternative carrier code is to be used at different times of the day:
 - a) uncheck *All-Day*,
 - b) enter the start time of the first carrier code in *Start Time 1*,
 - c) enter the second carrier's code in *Carrier Code 2* and
 - d) enter the second carrier's start time in *Start Time 2*.

If only one carrier is used for limited hours of the day:

- a) uncheck *All-Day*,
- b) enter the start time in *Start Time 1*, and
- c) enter the finishing time in *Start Time 2*, leaving the second carrier code field blank.

A carrier code can be up to 24 digits long, including "*" , comma "," (for 2-second pause); and "w" (for wait-for-dial-tone). A 2-second pause is automatically added after dialling a carrier code.

Use of Carrier Selection Prefix and Carrier Codes may require subscription to a carrier's discount calling plan. This service may not be available in some countries and on some telephone companies' telephone lines. Carrier Selection Prefix may not be required if a "Pre-Selection" service is provided by the telephone line service provider.

2.2.5.1 Activating ARS and Toll Restriction

To activate ARS and Toll Restriction:

1. Open the TalkSwitch Configuration program.
2. Select **System Information -> Auto Route Selection**.

3. Check *Activate ARS*.

By default ARS and Toll Restriction is disabled; and when enabled, applies to all Line Hunt Groups.

For selective activation of ARS and Toll Restriction on specific Line Hunt Groups:

1. Open the TalkSwitch Configuration program.
2. Select **System Information -> Auto Route Selection**.
3. Click on the **ARS Hunt Groups** button.
4. Check or uncheck the *Line Hunt Groups* to which ARS and Toll Restrictions are to be applied.

To complete ARS and Toll Restriction activation after entering all the required ARS and Toll Restriction data, save the setting to TalkSwitch using the **File > Save to TalkSwitch** from the Configuration program menu.

Extension Access to Line Hunt Groups for ARS & Toll Restriction

By default, all extensions have access to all Line Hunt Groups. As part of an ARS and Toll Restriction scheme it may be desirable to control extensions' access to Line Hunt Groups:

1. Open the TalkSwitch Configuration program.
2. Select **System Information -> Local Extensions**.
3. Click the **Hunt Group Access** button.
4. Check or uncheck each *Line Hunt Group* access to enable or disable extension access as required.

Repeat these steps for all extensions requiring restricted access to Line Hunt Groups.

Restricting extension access to one Line Hunt Group, allows the administrator to use ARS to control the appropriate selection Line Hunt Groups for each call.

Allowing Select Extensions to Bypass ARS and Toll Restriction

To allow select extensions to by-pass ARS and Toll Restriction requires dedicated Line Hunt Groups. These Line Hunt Groups must be configured for access only by the selected extensions, and excluded from ARS and Toll Restriction. The selected extensions with access to these ARS and Toll Restriction by-pass Line Hunt Groups, just dial the Line Hunt Group access

codes for access to the outgoing lines or VoIP trunks by-passing ARS or Toll Restriction.

To configure ARS and Toll Restriction bypass:

1. Include the required lines or VoIP trunks in the dedicated by-pass Line Hunt Groups (see section 2.2.4).
2. Open the TalkSwitch Configuration program.
3. Select **System Information -> Auto Route Selection**.
4. Click on the **ARS Hunt Groups** button.
5. Remove the checkmark next to each *Line Hunt Group* used to by-pass ARS and Toll Restriction.
6. For each extension select **System Information -> Local Extension** and click on the **Hunt Group Access** button, and check or uncheck access to *Line Hunt Groups* as required.

Direct Line Access

When Direct Line Access is enabled on an extension, calls placed from that extension are routed to a designated Line Hunt Group. If ARS is enabled on the designated Line Hunt Group, the calls are subject to ARS and Toll Restriction.

Telephone Line 3-Way Calling Service

The use of the telephone company's three-way calling and conference call services are not recommended on telephone lines when ARS or Toll Restriction are used. The system's ARS and Toll Restriction features do not have any control over the routing of calls through these telephone company services.

Emergency Service Numbers

The emergency access numbers, shown in the Emergency Service Section of the ARS configuration window, will not be blocked by Toll Restriction nor redirected by ARS. Calls to these emergency service numbers will be routed to Line Hunt Group 9.

It is recommended that any other local emergency service numbers be entered in to the ARS table to ensure that calls are routed to a Line Hunt Group with telephone lines providing access to emergency services.

Emergency Service Access

Ensure that Line Hunt Group is set-up to allow access to telephone lines with access to emergency services.

ARS Inter Digit Time-Out

ARS and Toll Restriction are implemented on each call when the longest matching entry is found in the ARS table or after an inter-digit time-out period. The default inter-digit time-out is set to 3-seconds. If it is necessary to change the inter-digit time-out:

1. Open the TalkSwitch Configuration program.
2. Select **Options -> Troubleshooting**.
3. Select the required time-out period from the pull-down menu of options on the ARS Interdigit time-out field.

Password Protection

It is recommend that the administrator password protection is enabled when ARS or Toll Restriction are used. Enable the password protection in the TalkSwitch configuration software under System Information -> Administration and enter a 4- to 8-digit numeric password.

Change Password Frequently

Change the administrator password frequently to prevent unauthorized changes to ARS or Toll Restriction,

Regulatory Advisory Notice

Call Redirection & Service Provider Billing Advisory

Use of the Automatic Route Selection, Toll Restriction and Call Detail Recording features does not imply any guarantee whatsoever by regulatory authorities, your telephone service provider(s), TalkSwitch or its distributors and resellers, with regard to the accuracy of these features and that the use of such a features may not be considered by a telephone company in any disputes which may arise regarding the accuracy of any subscriber's telephone account.

2.2.6 Fax information

TalkSwitch provides you with different options to configure your office for fax calls. The advantages and disadvantages of each option are described below, so that you can choose the best solution for your office.

Dedicated fax line

You may already have an incoming dedicated fax line. For more information, see *2.2.6.1 Dedicated fax line* on page 52.

Advantages: It is easy to set up and you don't need to change the way your fax handles calls.

Disadvantages: The dedicated fax line cannot be shared with the other phone devices (i.e. You cannot make an outbound voice call on your fax line). The dedicated fax line costs money and may not often be used.

Distinctive Ring fax detection

Distinctive ring is a service provided by your phone company. A second phone number is added to the same physical incoming telephone line. For more information, see *2.2.6.2 Distinctive ring fax detection* on page 54.

Advantages: It does not require a second physical incoming telephone line for faxes. More telephone lines can be added as you grow and you can keep the same fax arrangement.

Disadvantages: You only have one incoming telephone line and can only handle one call at a time. The phone company may charge a fee for a distinctive ring.

Automatic fax detection with the auto attendant

An auto attendant can detect incoming fax calls and route them accordingly. For more information, see *2.2.6.3 Automatic fax detection* on page 55.

Advantages: Calls are automatically handled for you, no distinctive ring service and no listening to ring patterns are required. Incoming telephone lines are shared between all phones and faxes, making better use of your resources.

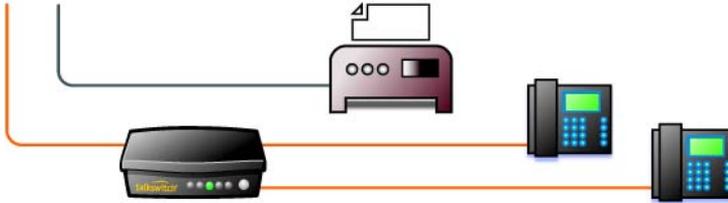
Disadvantages: Older fax machines do not emit a CNG tone and the auto attendant cannot detect them.

2.2.6.1 Dedicated fax line

If you have a dedicated fax line or number, your fax machine should be connected to an incoming phone line. Connect the remaining incoming telephone lines to your unit. Incoming telephone lines will be handled by the unit and your fax will work the way it has always worked. If however, you have a dedicated incoming telephone line for your fax machine that will be controlled by TalkSwitch, perform the steps below.

Figure 42: Dedicated fax line

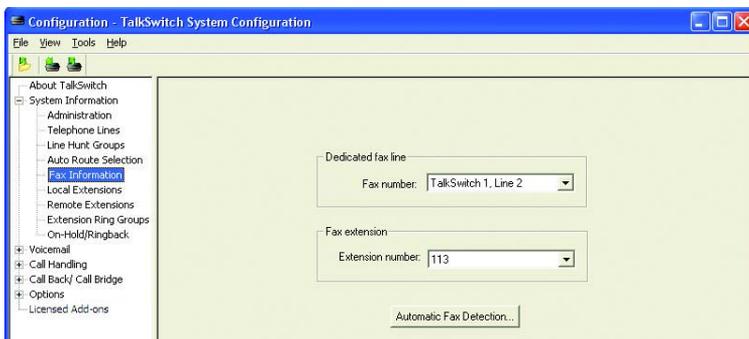
After installing TalkSwitch, the fax machine remains on its own dedicated line. All incoming telephone lines are shared by the remaining office phones.



1. Open the TalkSwitch configuration software.
2. Select **System Information -> Fax Information**.
3. In the *Dedicated fax line* section, select the fax number from the drop-down list next to *Fax number*.
4. In the *Fax extension* section, select the extension number where your fax is connected.
5. Save settings to TalkSwitch.

Example: If your fax machine is connected to E8 on the rear of unit 1, select extension number 118 for units with 8 extension jacks or 113 for units with 4 extension jacks. If your fax machine is connected to E8 on a unit with Unit ID 2, the fax extension number is 128.

Figure 43: Fax information



If you do not see your fax number in the list, make sure you have activated the line in the TalkSwitch configuration software under **System Information -> Telephone Lines**.

By default, when a dedicated line and a fax extension have been selected for the first time, the software automatically configures the incoming call handling properties for that line to ring only the fax extension that you have specified.

2.2.6.2 Distinctive ring fax detection

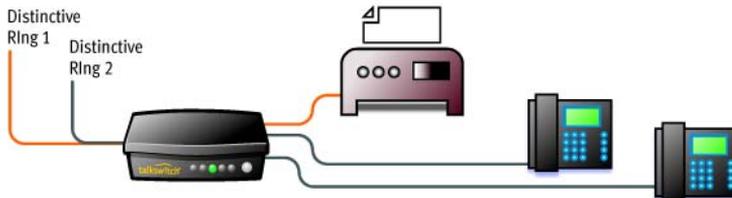
A single incoming telephone line can have two assigned numbers with a different ring pattern for each number. Based on the ring pattern, you pick up the phone or let the fax answer the call.

Figure 44: Distinctive ring without TalkSwitch



Figure 45: Distinctive ring with TalkSwitch

After installing TalkSwitch, it automatically detects the distinctive ring pattern. Faxes are automatically routed to the fax machine and calls to the phones.



Here is how TalkSwitch handles incoming calls:

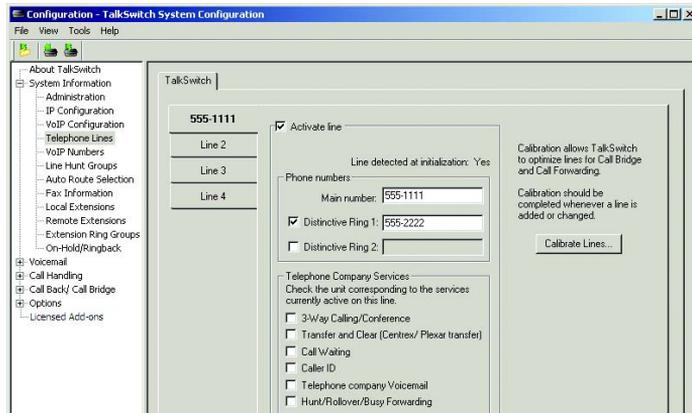
The unit automatically routes faxes to the fax machine and calls to the telephone.

One of the incoming telephone lines will have a distinctive ring number assigned to it. The unit recognizes this number as an incoming fax line. Follow the steps below:

1. Open the TalkSwitch configuration software.

2. Select **System Information -> Telephone Lines**.
3. Choose the line number for the incoming fax and click in the box next to *Activate line*.
4. In the *Phone numbers* section beside *Main number*, enter the telephone number that people dial to call your office.

Figure 46: Telephone lines



5. Click the checkbox beside *Distinctive Ring 1*:
6. Enter the distinctive ring number in the text field. This is the number people will call to get your fax machine.
7. Click **Fax Information**.
8. In the *Dedicated fax line* section, select the fax number from the drop-down list next to *Fax number*:
9. In the *Fax extension* section, select the extension number where your fax is connected.
10. Save settings to the unit.

2.2.6.3 Automatic fax detection

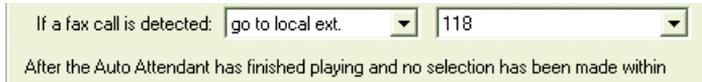
Provided all calls are answered by the auto attendant, it can detect incoming fax calls and will route calls to their destination.

If an auto attendant is already configured, do the following:

1. Open the TalkSwitch configuration software if not already open.

2. Select **Call Handling** -> **Auto Attendant**.
3. Select *Auto Attendant 1*. You can configure additional auto attendants later.
4. In the drop-down menu next to *If a fax call is detected*:, select *go to local extension* and select *118* (*113* for units with 4 extension jacks).

Figure 47: Fax routing



5. Save settings to the unit.

Set up at least one auto attendant to answer all incoming calls, play a recorded message, allow callers to dial their party's extension or dial 0 for a receptionist. For more information, see [2.4.2.1 Adding/configuring an auto attendant](#) on page 84.

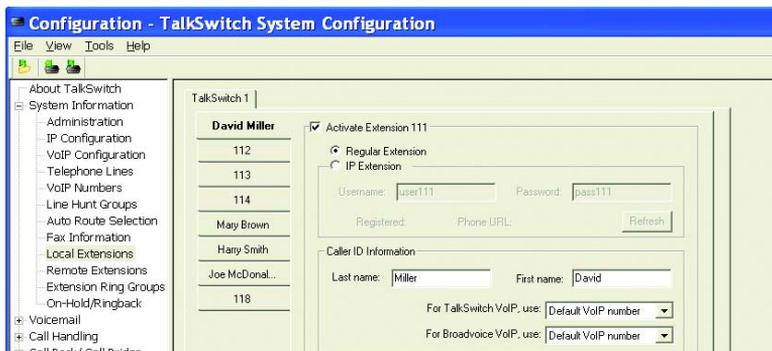
2.2.7 Local extensions

Local extensions can be analog phones connected to any of the extension jacks at the back of TalkSwitch or IP phones on the same subnet registered with your TalkSwitch system. For more details on connectivity, see [1.9 Connecting devices](#) on page 15. Local extensions are activated with the associated voice mailbox by default. Voice mailboxes for each local extension can be de-activated in the **Voicemail** folder.

Activating a local extension for use with an analog (regular) phone

By default, all analog extension ports are active. Plug phones into the extension ports and you are ready to make and receive calls.

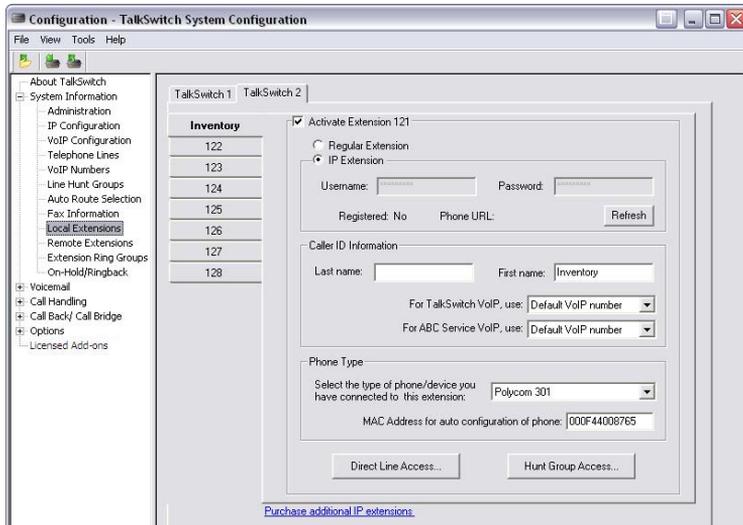
Figure 48: Analog extension



Configuring a local extension for use with a IP phone

1. Open the configuration software.
2. Select **System Information -> Local Extensions**.
3. Change the radio button from *Regular Extension* to *IP Extension*.
4. The default *Username* and *Password* are *<user ext#>* and *<pass ext#>*. These are used to manage calls between TalkSwitch and the IP phone and must match the username and password on the IP phone. Change only if required.

Figure 49: IP extension



Refer to the *IP Phone Configuration Guide* by clicking the link **Start -> TalkSwitch 4.10 -> Documentation -> IP Phone Guides -> Adding IP Phones to TalkSwitch**. If you are using a third-party IP phone, refer to the user guide for that phone. You can also click the *Purchase additional IP extensions* link at the bottom of the window (Figure 49) for information and for entering an activation key.

Once the phone has been programmed with the extension number, username, password, registrar and proxy information you can confirm registration as follows:

1. Connect the IP phone to the LAN with the Ethernet cable.
2. Connect the power to the IP phone.
3. Click the **Refresh** button when the phone indicates that it is registered.

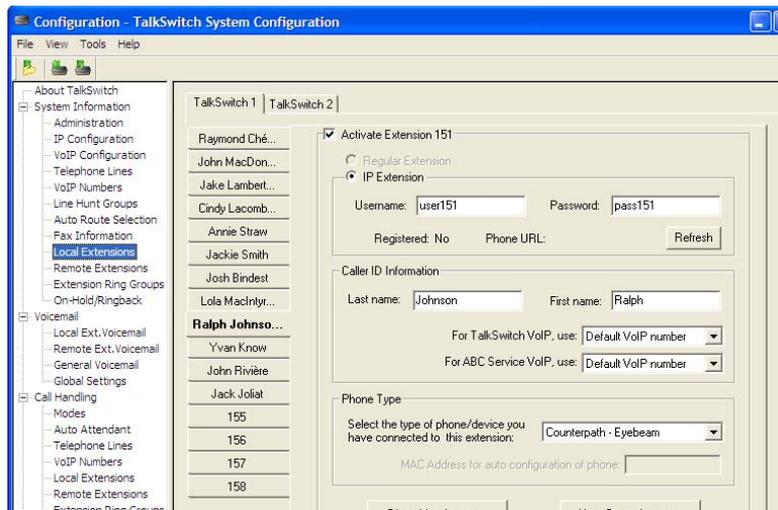
Configuring additional IP extensions

If you have added IP extensions as a licensed add-on (see 2.7 *Licensed Add-ons* on page 120 for more details), tabs appear for the configuration of these extensions (151-158 for unit 1, 161-168 for unit 2, etc.). The *Regular Extension* selection is unavailable as this series of tabs can only be configured as IP extensions.

1. Open the configuration software.
2. Select **System Information -> Local Extensions**.
3. The default *Username* and *Password* are *<user ext#>* and *<pass ext#>*. These are used to manage calls between TalkSwitch and the IP phone and must match the username and password on the IP phone. Change only if required.

When Polycom or Grandstream is selected, the username and password are fixed and not editable.

Figure 50: Additional IP extensions



Additional fields

Caller ID Information section

Last name/First name:

Enter the last and first name of the person associated with the extension. If you want to include the person in the dial-by-name directory used by the auto attendant, you must enter the last name. The name entered displays on

the Caller ID of other local extensions. The person using that extension is required to record her/his name for the directory.

If a name was not recorded before you set up the associated voice mailbox, pick up the handset, press *** * *** + <mailbox number> and press 4.



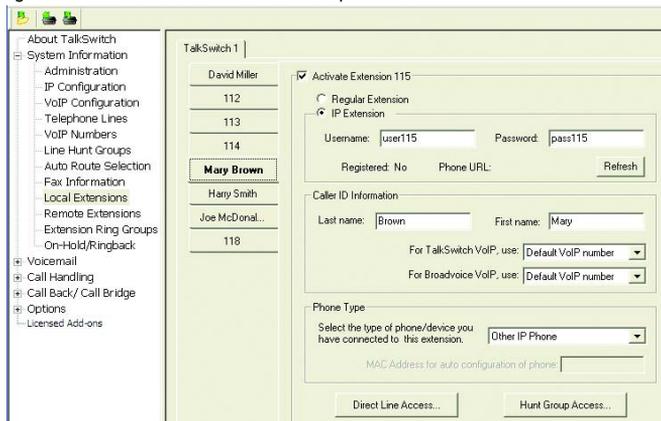
For TalkSwitch VoIP calls, use:

Select the number you want to display at the remote location when VoIP calls are placed from this extension. If *Default VoIP number* is selected, the phone number associated with the outgoing VoIP port is used. This selection makes the most sense if your VoIP numbers have been configured to handle all inbound calls the same way.

For Service Provider VoIP calls, use:

Select the number you want to display at the remote location when VoIP calls are placed from this extension through a service provider.

Figure 51: IP extension username and password



Phone Type

If you have a TalkSwitch phone connected to this extension, select the model of phone from the pull-down menu. TalkSwitch uses this information to configure the phone to support the features that match the capabilities of the phone. Features include intercom paging, group paging over the speaker, and message waiting counter on the TS-80, TS-200, TS-400 and TS-600 phones.

The IP phones officially supported by TalkSwitch are:

- Polycom Series — 301, 430, 501, 601 (minor support differences between models)
- Grandstream — GXP-2000 (excellent low budget phone)
- Counterpath — Eyebeam soft client.

When Polycom or Grandstream is selected, the username and password are fixed and not editable.

Other phones that support the G.711 codec may work with the system, but not all the features may be supported. **We strongly recommend that you only use the IP phones supported by TalkSwitch.** If you connect an unsupported IP phone, select *Other IP Phone* from the pulldown menu. As TalkSwitch cannot enable special features, or customization of these phones, further configuration will be limited to the programmable options on the phone itself.

Figure 52: Phone types



Direct Line Access... (button)

This feature allows users to access an outbound line without dialing extra digits. When the extension goes off-hook or a number is dialed on a IP phone without first dialing a hunt group, **Direct Line Access** allows you to configure the extension to access a specific hunt group automatically for outbound calls.

Example: You may have a fax machine connected to this extension and don't want to reprogram the speed dial numbers. Click **Direct Line Access** and choose the hunt group. As soon as the fax goes off-hook, the unit finds an available line.

Warning: When using **Direct Line Access** with an analog phone, you hear the telephone company dial tone when you pick up the handset. The following features are only available from the internal dial tone:

- Intercom calling
- Call pick-up
- Hold and retrieve
- P.A. access
- Line hunt groups
- Parked call retrieval
- Voicemail retrieval/access
- Do Not Disturb functions
- Entering command mode

To obtain an internal dial tone at an extension which has **Direct Line Access** enabled, dial **Flash** after going off-hook on an analog set. On a IP phone, dial *** * ***, then press **Dial, Send** or **#**.

Hunt Group Access... (button)

You can permit or restrict specific hunt group access to the extension. Click the checkbox next to the Line hunt groups that you want to allow for this extension.

2.2.8 Remote extensions

A remote extension allows you to connect an external telephone number. This could be a cell phone or a number to your office. You can have it operate similar to a local extension in your office. Any external telephone number can be configured as a remote extension (i.e. cell phone, PDA, home telephone or branch office).

With remote extensions, you can have calls forwarded directly from your office to any phone number you choose. You can choose to screen callers. If the remote extension is busy or unanswered, calls can be transferred back to your office voicemail, another extension or forwarded to another out-of-office telephone number (e.g. to a business partner's cell phone).

To set up remote extensions, see *Figure 53* and follow the steps below.

1. Open the TalkSwitch configuration software.
2. Select **System Information** -> **Remote Extensions**.
3. Click on a remote extension number, starting with 210. Add more numbers as needed.

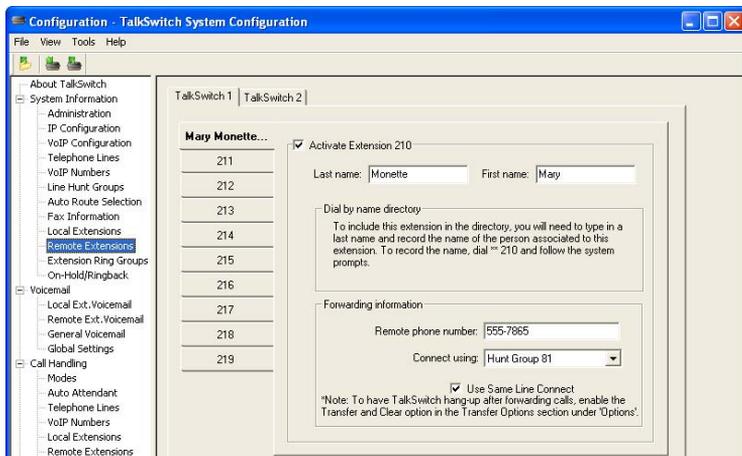
If a TalkSwitch has Unit ID 2, the remote extension numbers are 220–229. Unit ID 3 has remote extensions 230-239 and Unit ID 4 240-249.

4. Click in the checkbox next to *Activate Extension <ext>*.
5. Enter a last and a first name into the *Last name:* and *First name:* fields to identify the extension.
6. Enter the external phone number.

Enter the number the same way you are dialing it from a regular phone. You may use digits 0–9, space, comma, dash, **#** and *****. A comma creates a two-second pause in the dialing string.

7. In the drop-down list next to *Connect using:*, we recommend that you leave it at *Hunt Group 9* unless you have pre-configured another group.
8. If you have a *3-Way Calling* or *Transfer and Clear* service from the phone company, click the checkbox next to *Use Same Line Connect*.
9. Save settings to the unit.

Figure 53: Remote extensions



Once your remote extensions are configured, calls can be forwarded by the auto attendant or transferred by someone in your office. There are 10 remote extensions available per TalkSwitch unit.

Use Same Line Connect

Check this box to allow incoming calls to use the same line when forwarding it to the remote extension. This feature allows only one line to be used when forwarding incoming calls to a remote extension. It can only be used when the line of the originating incoming call has *3-Way Calling/Conference* or *Transfer* service (Centrex/Plexar). If the line does not have any of these services, the call will be forwarded using the hunt group chosen in the *Connect using*: drop-down list.

If you have a *Transfer and Clear* service on your line(s) and want to have TalkSwitch clear lines after forwarding calls to remote extensions, you need to enable *Clear telephone line after call transferred* (**Options -> Transfer Options**). If your service requires the remote extension to answer the call before the transfer can be made, use one of the screening options in **Call Handling -> Remote Extensions** in the drop-down list under the *Answered* tab.

Remote extensions do not have the ability to transfer calls manually when using *Same Line Connect*.

When using *Same Line Connect*, callers hear silence while they are on hold. The callers are on hold at the telephone company and not at TalkSwitch. If you want callers to hear music while they are on hold for calls forwarded to remote extensions, don't use *Same Line Connect*. It is best to experiment with different settings in order to decide what works best for your configuration.

Forwarding calls to a remote extension

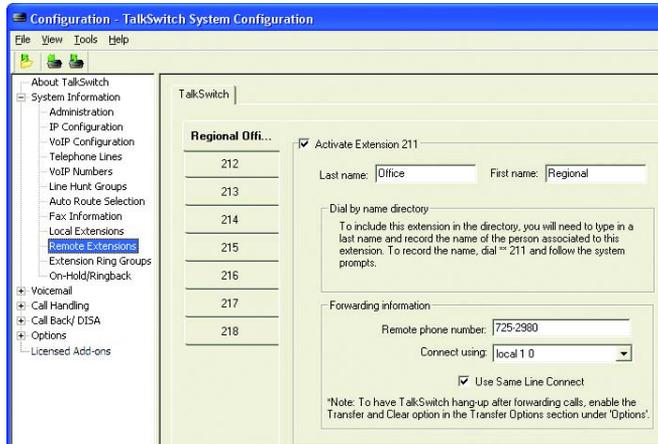
- Manually from a local extension:
Analog phone: Press **Flash** and the remote extension number.
IP phone: Press **Transfer** and the remote extension number followed by **Dial, Send** or **#**. See *Appendix A: Functions and Commands* on page 195 for common analog and IP phone commands.
- From an auto attendant:
The auto attendant can be configured to route the calls to your remote extension. The auto attendant might say: *"To reach Mr. A's cell phone, press 1, to reach his office, press 2 or press 0 to reach a receptionist."* You can also instruct callers to dial the extension directly at the auto attendant. You could use this auto attendant as a company greeting or within your call cascade sequence at your local extension. For details, refer to *2.4.2 Auto attendant* on page 83.
- From a call cascade:
A call cascade sequence can route a call to a remote extension.
If there is no answer at your desk, the call is transferred to your cell

phone. If you don't answer your cell phone, the call is forwarded to your office voicemail. The caller hears music on hold. For details, refer to *2.4.5.1 Local extensions* on page 93.

Transferring calls from a remote extension:

1. Press *** * *** and dial any local extension, remote extension, or ring group, or press *** *** again and a mailbox number to transfer the call to voicemail.

Figure 54: Remote extensions



2.2.9 Extension ring groups

An extension ring group is a group of local extensions that ring in unison. TalkSwitch supports 10 ring groups with numbers 300 to 309. They are accessible from local extensions and through the auto attendant. By default, extension 300 is configured to ring all extensions.

A phone system with 4 units can support a combination of 32 analog/IP extensions plus 32 optional IP extensions, for a maximum of 64 extensions.

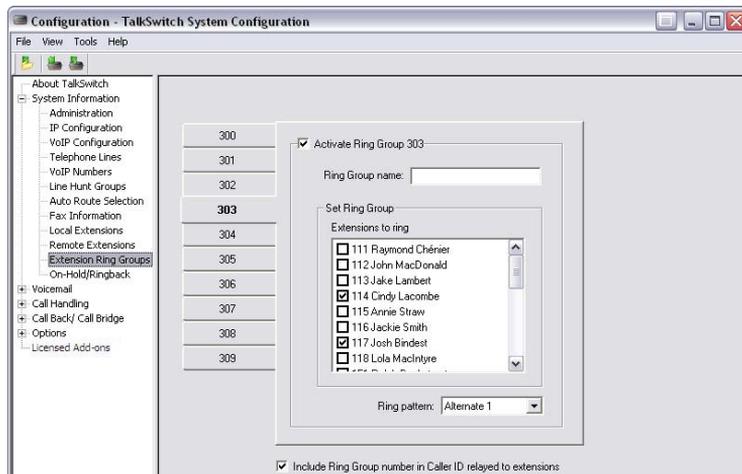
Ring groups features

- independent call cascade
- distinctive ring cadence
- call screening capabilities

Setting up extension ring groups

1. Open the TalkSwitch configuration software.
2. Select **System Information -> Extension Ring Groups**.
3. Choose an *Extension Ring Group* number, starting with 300.
4. Click in the checkbox next to *Activate Ring Group 300*.
5. Enter a name in the field next to *Ring Group Name*.
6. In the *Set Ring Group* window, choose the local extensions you want to ring in unison.
7. Select a *Ring pattern*: from the drop-down list to identify the incoming call before you pick up the phone.
8. Save settings to TalkSwitch.

Figure 55: Ring groups



Different uses for extension ring groups

You can send calls to specific groups of employees.

Example: If you have a company with several departments, you can configure extension ring group 300 for the sales department to ring extensions 111, 112, 113, and 114. Ring group 301 might be for technical support, ringing extensions 115 and 116.

Identify types of calls by different ring patterns

Example: You are the president of a company and you don't want to handle calls from the general public, yet you want to be alerted when important colleagues try to contact you. Your regular local extension may be 111 for most people, but you can configure an extension ring group 305 to ring your extension with a different ring pattern. You tell your colleagues to choose extension 305 at the auto attendant or ask to be transferred to extension 305 when speaking to someone in the office. You can tell by the ring pattern if the call is to 305 or 111.

Forwarding calls to a ring group

- Manually from a local extension:
Analog phone: Press **Flash** and the ring group number
IP phone: Press **Xfer** or **Transfer** and enter the ring group number, then press **Dial** or **#**.
- From an auto attendant:
Auto attendants can be configured to send calls to ring groups just as they do to extensions. The auto attendant might say: *"To reach the sales department, press 1, to reach Mrs. A's office, press 2, or press 0 to reach a receptionist."* You could use this auto attendant as a company greeting or within a call cascade sequence at a ring group.
- From a call cascade:
The call cascade sequence can route the call to ring group 300.
Example: If there is no answer at your desk, the call is transferred to the sales department. If no one answers the phone in sales, the call is forwarded to your voicemail. If music on hold is enabled, the caller hears music throughout the entire process.

2.2.10 On-Hold/Ringback

The *Music on Hold* feature lets you provide music or a recorded message for callers while they are waiting to be routed or transferred.

Callers tend to react more positively to transfers or waits if they hear music. Companies can also play recorded advertisements or provide information about their business.

There are two ways to add music on hold to your system:

- an external audio source
- an internal file

2.2.10.1 External audio source

An external audio source could be a CD player, tape player or a PC media player that you connect to TalkSwitch. See 1.9.3 *Connecting devices to the music jack* on page 21 for additional details.

Advantages: The variety and length of music are virtually unlimited.

Disadvantages: Requires extra, dedicated equipment to be connected and running constantly with separate sound sources for each unit

1. Connect your external audio source to the unit using a 1/8" mono cable.

Figure 56: Music port

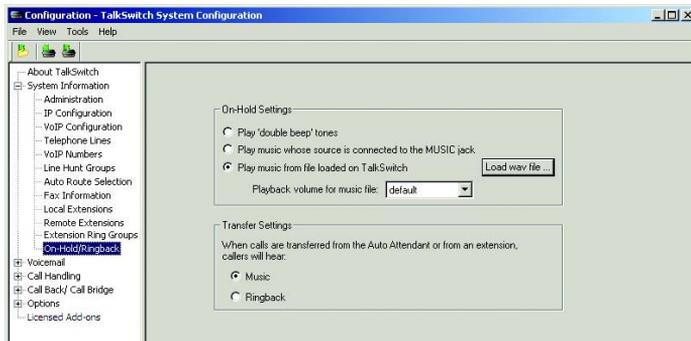


If you have more than one unit connected to the LAN and you are using an external music source, you need to connect the source to all units. Use a 1/8" audio splitter, available at most electronics stores. If you are using a .wav file loaded on the unit, the music file is copied to all units on the LAN.

You can use separate audio sources for each unit.

2. Once the audio source(s) is (are) connected to TalkSwitch, turn it (them) on.
3. Connect the unit to your computer and open your configuration software.
4. Select **System Information -> On-Hold/Ringback**.
5. In the *On-Hold Settings*, select *Play music whose source is connected to the MUSIC jack*. See Figure 57.
6. Save settings to the unit.

Figure 57: Music on hold



On-Hold Settings

There are three options when callers are placed on hold:

1. *Play 'double beep' tones* to callers when they are on hold. This is the default setting. No additional configuration is required.
2. *Play music whose source is connected to the MUSIC jack*, if you have a music source connected to the music jack.
3. *Play music from file loaded on TalkSwitch*, if you have a sound file loaded. You can convert MP3 and other file formats using a utility like Audio Converter or Ripper. Download a converter at: www.audoi.com. Check the volume of music on hold with a test call. Adjust the levels as required. This setting is not real-time. Save the configuration to the unit and make another test call.

2.2.10.2 Internal audio file

You can load a digital audio file into memory.

Advantages: No additional equipment is required. You can use the same file for all units in the system. This provides seamless music for calls transferred across the network.

Disadvantage: The length of the recording is limited by your memory capacity.

See *1.11 Increasing Memory Capacity* on page 26 for more details.

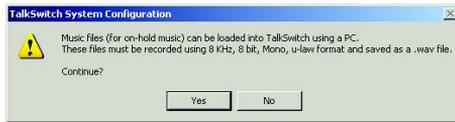
To load a digital audio file into memory:

1. Connect the unit to your computer and open your configuration software.

2. Select **System Information -> On-Hold/Ringback**.
3. In the *On-Hold Settings* section, select *Play music from file loaded on TalkSwitch* See *Figure 68*.
4. Click the **Load wav file...** button. An information box displays the required format for the digital audio file. This file must be 8 KHz, 8-bit, mono, μ -law and saved as .wav format.

If your desired digital audio file conforms to the required format, click **Yes**, or else click **No** to exit.

Figure 58: Music file format

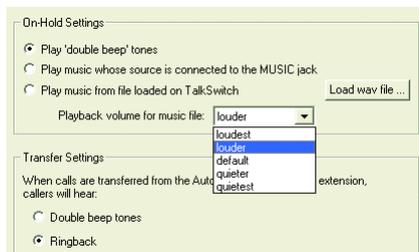


5. If you clicked **Yes**, the *Load Audio File* window displays. **Browse** to the location of the desired digital audio file, select the file and click on **Open**.
6. In *Transfer Settings*, seen in *Figure 59* on page 69, select *Music* if you want the caller to hear music instead of a ringback tone.
7. Save settings to the unit.

Set the volume for Music on Hold

TalkSwitch offers the ability to adjust the volume of music on hold playback. In *On-Hold Settings*, to increase or decrease the sound level, use the drop-down list for *Playback volume for music file*:

Figure 59: Volume adjustment



Transfer Settings

Decide if you would like your callers to hear ringing, music or double beep tones while they are being transferred.

If you connect a music source to the music jack but do not enable *Music on Hold*, the caller hears both music and beep sounds. When the callers make a selection at the auto attendant to go to extensions with call screening enabled, they will hear ringing and music.



2.3 VOICEMAIL

This section deals with the internal voicemail system included in TalkSwitch. TalkSwitch provides three types of voicemail:

- **Local**
The mailbox is associated to a local extension (analog or IP). By default, each local extension and additional IP extension has its own voice mailbox. For information on enabling additional IP extensions using TalkSwitch licensed add-ons, see *2.7 Licensed Add-ons* on page 120.
- **Remote**
The mailbox is associated with a remote extension, such as a mobile phone. These mailboxes must be assigned by the user at the extension.
- **General**
The general system mailboxes are for general use or group voicemail. They are not tied to any specific extension.

2.3.1 Mailbox options

Local, remote, and general voice mailboxes can be configured the same way.

The following procedure describes the setup of local extension voicemail. You can use the same steps for setting up remote and general voicemail.

1. Open your TalkSwitch configuration software.
2. Select **Voicemail -> Local Ext. Voicemail**.
3. Click *Activate as mailbox <ext>*.

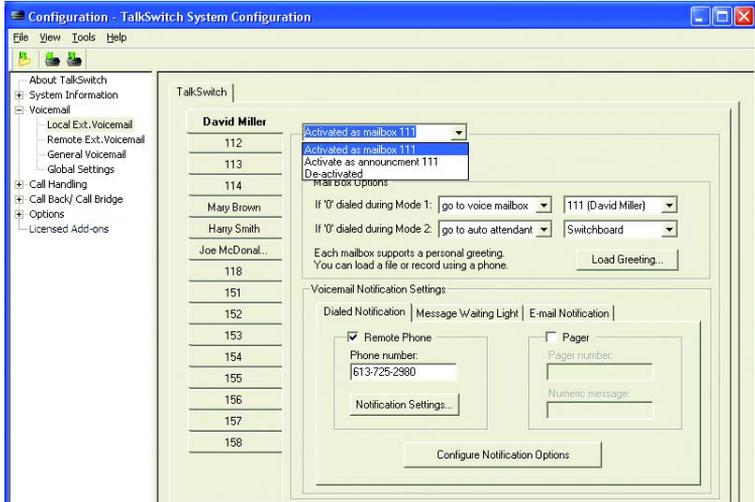
Under remote extensions, make sure that the remote extension has been enabled, otherwise you cannot activate its mailbox.

4. You can use this option to relay information. If you want to prevent callers from leaving messages, select *Activate as announcement <ext>* from

the drop-down list. Callers hear an announcement. During the announcement, callers can dial **0** or ***x** to get back to the auto attendant before getting disconnected. (note that **0** is a configurable option and by default does not route to the auto attendant).

5. In the *Mail Box Options* section, select one of the actions to perform when a caller presses 0 during mode 1 or during mode 2. The action options are the same for the different modes.

Figure 60: Mailbox and mode configuration



If you require additional memory for voicemail and auto attendants, memory cards can be purchased from www.talkswitch.com or from your local reseller. Additional memory capacity can also be added using TalkSwitch licensed add-ons (see 2.7 *Licensed Add-ons* on page 120). For details on how to use voicemail, see 3.8 *Using the voicemail system* on page 146.

Load Greeting (button)

You can load a pre-recorded voicemail greeting from your PC to TalkSwitch. The file must be an 8KHz, 8-bit, mono, μ -law .wav file.

2.3.2 Voicemail notification

2.3.2.1 Dialed notification

Voicemail Notification Settings section

Under the *Dialed Notification* tab, choose the process by which you want to be notified — *Remote Phone* or *Pager*, or both.

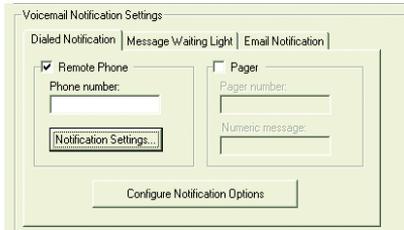
Remote Phone

If you want to be notified at another location when new messages arrive, use remote phone notification.

1. Click the checkbox next to *Remote Phone*.
2. Enter the remote phone number as it would normally be dialed from where the unit is located.

Example: If your remote phone number is 905-555-1111 and you dial 905-555-1111 to call your phone locally, you would enter 905-555-1111 as the remote phone number.

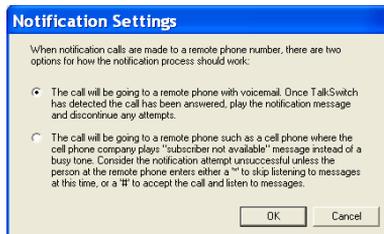
Figure 61: Dialed Notification



Notification Settings (button)

Click on the **Notification Settings** button and choose one of the two options to handle notification calls.

Figure 62: Notification settings



1. Perform the notification process once:
When the unit detects the call was answered, it will play the notification message once only, then discontinue any attempts to re-connect.
2. The unit continues to re-connect until the recipient presses ***#** or **#** and considers the notification attempt successful if the person at the remote phone presses ***#** to skip listening to messages or **#** to accept the call and listen to messages.

This option is useful for cell phones where the cell phone company plays a “*subscriber not available*” message instead of a busy tone. Once the cell phone is answered, TalkSwitch prompts you to respond to the incoming voicemail with ***#** or **#**.

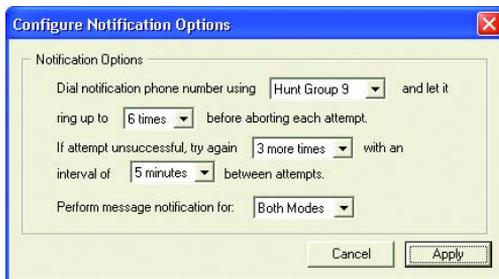
1. Click the checkbox next to *Pager* if you want to be notified by pager when new messages arrive.
2. Enter the pager number as you would normally dial it from your TalkSwitch location.
3. Enter the numeric message to be displayed on the pager.

Configure Notification Options (button)

Define how TalkSwitch dials out of your office when notifying you of a voicemail. If the first connection is not successful, determine the number of times TalkSwitch continues to attempt the connection.

When you choose to be notified by pager and remote phone, TalkSwitch ensures that both locations have been contacted before finishing the notification process.

Figure 63: Configure notification options



Dial notification phone number using:

Select the hunt group you want to use for the remote phone and pager notification calls. You can also adjust the number of rings before it aborts the attempt and follows the parameters listed in the next section.

If attempt unsuccessful, try again:

These parameters control how many times TalkSwitch tries to notify the remote phone or pager before it aborts the notification attempt. You can specify the time interval between attempts.

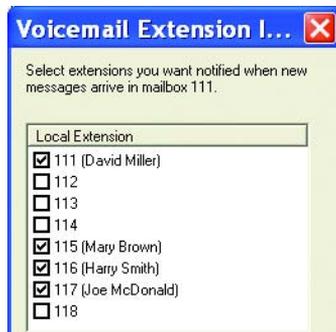
2.3.2.2 Message waiting light

You can set a message waiting light to indicate when a message has been received by an extension. Select the local extensions in your system where you want the light to flash when a message arrives.

You can configure this feature for each mailbox.

1. Click on the *Message Waiting Light* tab.
2. Click **Edit** to select the extensions to notify when new messages arrive in a mailbox. The *Voicemail Extension Indicators* window opens. By default, the message waiting light turns on at the extension associated with a local mailbox when a message is received.
3. Click the checkbox next to the extensions to notify when new messages arrive in a mailbox.
4. Click on **OK** when done.

Figure 64: Extension indicators



2.3.2.3 E-mail notification

Voicemail notification and messages can also be forwarded to any e-mail account with or without an audio file of the message. If you configure the unit to send the voicemail message, you do not need to call to pick up voicemail. Notifications include caller ID data.

TalkSwitch has a limit of 250 e-mail addresses. You can assign up to four e-mail addresses for notification to each mailbox.

User must configure the e-mail server settings in the **Voicemail -> Global Settings** section for this feature to work.

Example 1: 1 unit: 8 local extensions (can be a combination of analog and IP), 10 remote extensions and 10 general mailboxes with 4 e-mail notification addresses each (28 x 4 e-mails = 112).

Example 2: 4 units on a LAN: 32 local extensions, 40 remote extensions and 10 general mailboxes (250/82 mailboxes = 3 e-mails/mailbox).

Certain e-mail addresses may apply to more than one mailbox (e.g. sales@abccompany.com).

To configure e-mail notification:

1. Click on the *E-mail Notification* tab.

Figure 65: E-mail notification tab



2. Click the checkbox next to *Enable e-mail notification*.
3. Click **Edit** to add or modify or remove e-mail addresses.

Figure 66: Add, remove or modify e-mail addresses



To add an e-mail address:

1. Click **Add**.
2. Enter the name of the recipient of the e-mail notification.
3. Enter the e-mail address of the recipient of the e-mail notification.
4. Click one of the radio buttons next to *Attachment*.
 - a) **Yes** — The entire voicemail message gets attached to the e-mail as a sound file in .wav format.
 - b) **No** — The e-mail includes notification information, but there is no file attached.
5. Click **OK**.

Figure 67: Enter name and e-mail address



Example of an e-mail notification:

New message from Joe Smith (555-1234)

The body of the e-mail reads:

Voice Mailbox 111

New message from Joe Smith (555-1234)

Received: Monday, April 5 at 5:45pm
Length: 23 seconds

Total un-reviewed messages: 13
Total saved messages: 2

To modify an e-mail address:

1. Select the address you wish to modify.
2. Click **Modify**. Modify the entries as needed following steps 2–5 above.

To remove an e-mail address:

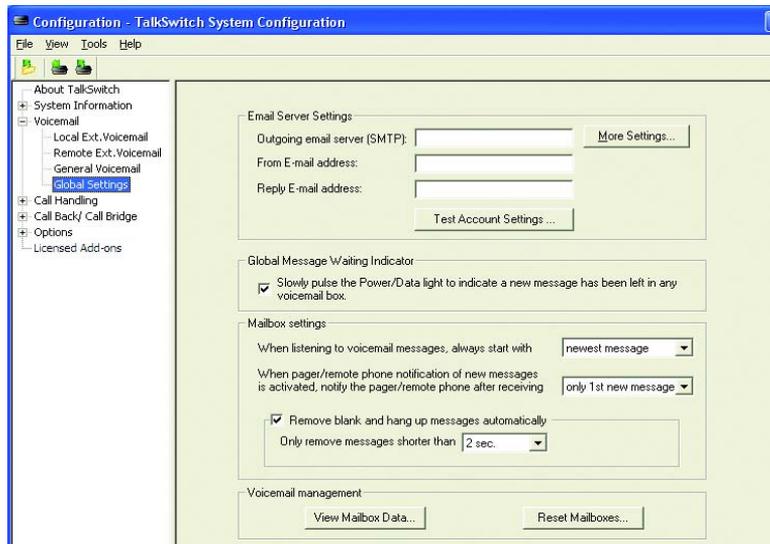
1. Select the address you wish to remove and click **Remove**.

Make sure you configure the *E-mail Server Settings* under **Voicemail -> Global Settings**. For more information, see 2.3.3 *Global Settings*.

2.3.3 Global Settings

These settings allow you to configure your e-mail server parameters and fine-tune your voicemail system. Any changes made to **Global Settings** affect all system voice mailboxes.

Figure 68: Global Settings



1. Select **Voicemail -> Global Settings**.

Figure 69: E-mail server settings



In order to send e-mail notifications, the system must be programmed with e-mail account information from your Internet Service Provider (ISP) or administrator. If using MS Exchange for e-mail, there are cases when notification may not work properly if certain security features are enabled.

2. Enter the name of your *Outgoing e-mail server (SMTP)*.
Example: *mail.mymailserver.com*
3. Enter an e-mail address in the *From E-mail address* field. When e-mail notifications arrive from the system, this e-mail address will show up in the *From* field.
4. Enter an e-mail address in the *Reply E-mail address* field if you want responses to be handled by someone like an administrator.
Example: *administrator@companyABC.com*. If you leave the field empty, the system will use the e-mail address entered in the *From* field.
5. Click **More Settings** to enter the parameters for server authentication.

Figure 70: More e-mail settings



6. If you wish to verify your settings, click **Test Account Settings**.

Figure 71: Test account settings



Global Message Waiting Indicator

Click the checkbox next to *Global Message Waiting Indicator* if you would like the *Power/Data* light on the TalkSwitch unit to flash when you have voicemail in any TalkSwitch voice mailbox.

Mailbox Settings section

Next to *When listening to voicemail messages always start with*, select *newest message* (default) or *oldest message* to play first.

Next to *When pager/remote phone notification of new messages is activated, notify the pager/remote phone after receiving*, select *every new message* or *only 1st new message*.

For the Maximum recording length option, select any duration between 1 and 8 minutes. The default setting is *2 minutes*.

Click the checkbox next to *Remove blank and hang-up messages automatically*, if you want to delete blank messages (callers hang up when they reach your voicemail). If you have the caller ID service from your telephone company and choose to keep the messages, you will be able to check the caller's number when they leave a blank message.

2.3.3.1 Voicemail manager

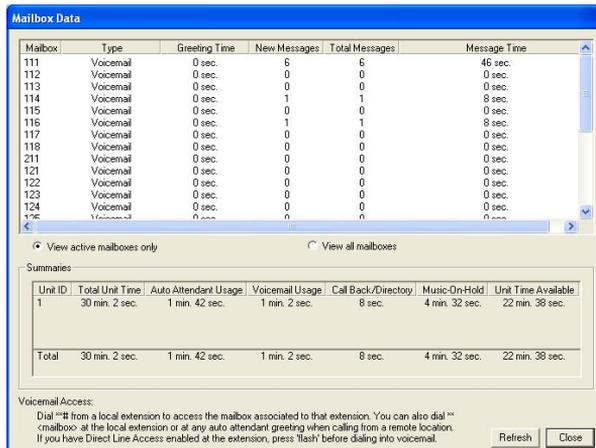
The two options, View Mailbox Data and Reset Mailboxes, in the Voicemail Management section of the Global Settings window are also available under the main menu bar: **Tools -> Voicemail Manager...**

Figure 72: Voicemail manager



View Mailbox Data (button) in the *Global Settings* window, or **View Status** (button) in the *Voicemail Manager* window:
View the details for each voice mailbox and announcement followed by a summary to manage the voicemail resources of the system.

Figure 73: Mailbox data



Reset Mailboxes (button)
All greetings, messages and settings for the selected mailbox(es) can be reset.

Figure 74: Reset Mailboxes



To delete a password for a particular voice mailbox:

1. **Tools -> Voicemail Manager** (main menu).
2. Click the **Delete Password...** button
3. Select the mailbox from the *Delete password for mailbox:* drop-down list.

Figure 75: Delete the password of a voice mailbox



4. Click **OK**.

2.4 CALL HANDLING

2.4.1 Modes

You may want the unit to handle calls differently depending on the time of the day, the day of the week or on holidays. These configurations are called *Modes*. You can configure all aspects of call handling (i.e. call cascade and auto attendants) differently for each mode. Define when you would like the unit to switch modes and name the modes to suit your environment.

1. Select **Call Handling -> Modes**.
2. Create a name for *Mode 1* and *Mode 2*. Here, we have named them *Business Hours* and *After Hours*. See *Figure 76* on page 82.

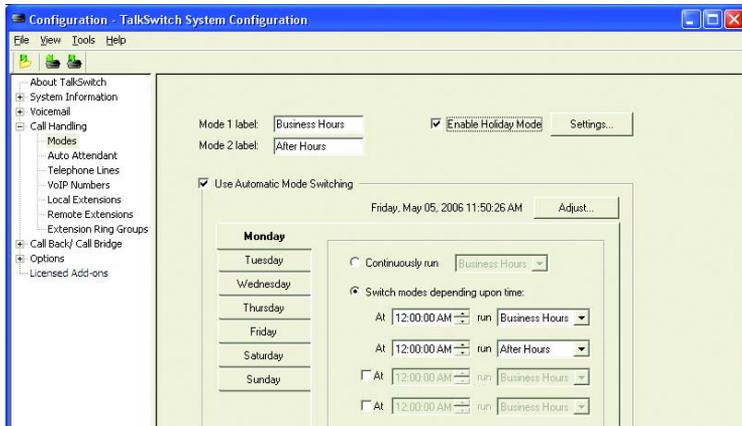
3. Click the checkbox next to *Use Automatic Mode Switching* if you want the modes to switch automatically.
4. Set the day and time when you would like the modes to change. You can click Monday through Sunday and create unique settings for each day.
5. To use manual mode switching, change the modes at any local extension.

Switch modes manually using telephone touchtone commands.

1. Dial **#** or ***55** on a IP phone to enter command mode.
Enter your password if required.
 - b) Dial any of the following commands and press **#** to confirm.
 - 30 TalkSwitch tells you what mode it is currently using.
 - 31 Change TalkSwitch to mode 1.
 - 32 Change TalkSwitch to mode 2.
 - 33 Change TalkSwitch to holiday mode.

Warning: The mode will not change if you are logged into the configuration software tool at the time of the scheduled mode switch.

Figure 76: Modes

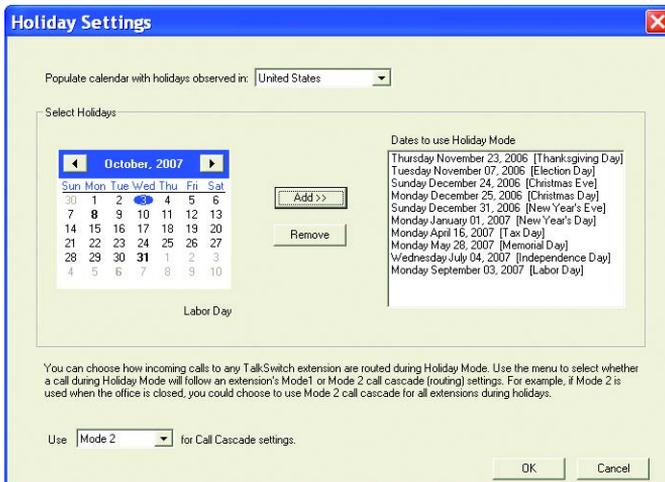


To set holiday mode:

1. Click the checkbox next to *Enable Holiday Mode*.
2. Click **Settings**.

- a) Select the country in the drop-down list next to *Populate calendar with holidays observed in*. See Figure 77 on page 83.
- b) Select any date in the calendar.
- c) Click **Add >>**.
- d) Repeat b) and c) until all required holidays are listed.
- e) Select the mode to *Use for Call Cascade settings* and click **OK**.

Figure 77: Holiday mode



You can choose how incoming calls to any TalkSwitch extension are routed during Holiday Mode. Use the menu to select whether a call on a holiday will follow an extension's Mode 1 or Mode 2 call cascade (routing) settings. For example, if Mode 2 is used when the office is closed, you could choose to use Mode 2 call cascade for all extensions during holidays.

2.4.2 Auto attendant

TalkSwitch can handle inbound calls with an auto attendant. Auto attendants can answer all incoming calls, play a recorded message and route callers to their requested extensions.

If the receptionist is away from the desk, or if you do not have a receptionist, an auto attendant responds and gives callers options. Calls cannot follow an extension's call cascade sequence until an auto attendant has answered or the caller has been manually transferred.

TalkSwitch has nine auto attendants. They can be used together to create powerful call routing options.

Callers can dial local extensions, remote extensions, extension ring groups (300-309), voicemail to leave and retrieve messages, access call back/call bridge and enter command mode to make changes to the system configuration from the handset.

400 series mailboxes can be dialed at the auto attendant without pressing ***6** first. This is useful for providing virtual extensions associated with a mailbox for telecommuters or other part-time staff.

2.4.2.1 Adding/configuring an auto attendant

Follow the steps below to set up an auto attendant to answer all incoming calls, play your recorded message and enable callers to dial their party's extension or dial 0 for a receptionist.

1. Open the TalkSwitch configuration software, if not already open.
2. Select **Call Handling** -> **Auto Attendant**.
3. Select *Auto Attendant 1* and type a name into the *Auto Attendant Name* field. You can configure and name 2-9 later.
4. Follow these settings as shown in *Figure 85*. Set up extension 114 as your reception and enter the appropriate values for your office.

Figure 78: Configuring auto attendant

Auto Attendant 1

Auto Attendant Name:

At the Auto Attendant, perform the following actions:

If the caller selects '0':	<input type="text" value="go to voice mailbox"/>	<input type="text" value="114"/>
If the caller selects '1':	<input type="text" value="go to auto attendant"/>	<input type="text" value="1 (Switchboard)"/>
If the caller selects '2':	<input type="text" value="go to remote ext."/>	<input type="text" value="210 (my remote number)"/>
If the caller selects '3':	<input type="text" value="go to ring group"/>	<input type="text" value="300 (Documentation)"/>
If the caller selects '4':	<input type="text" value="go to auto attendant"/>	<input type="text" value="1 (Switchboard)"/>
If the caller selects '5':	<input type="text" value="go to auto attendant"/>	<input type="text" value="1 (Switchboard)"/>
If a fax call is detected:	<input type="text" value="go to local ext."/>	<input type="text" value="118"/>

After the Auto Attendant has finished playing and no selection has been made within

seconds, then

Utilities for Auto Attendant

Before an Auto Attendant can be fully functional, you need to record your Auto Attendant message using a telephone or load a pre-recorded Auto Attendant.

5. Callers who don't have a touchtone phone can be routed to the receptionist at extension 114.
6. Set the time from the drop-down list *After the auto Attendant has finished playing and no selection has been made within* (0–30 sec.).
7. Select *go to local ext.* and select 114 from the drop-down lists.

You can record the other 8 auto attendants. By default, the total combined recording time is 30 minutes. The 30 minutes are shared with voicemail. You can expand the built-in memory using memory cards or internal memory add-ons. See *1.11 Increasing Memory Capacity* on page 26 for more details.



When two or more units are networked, they share the same 9 auto attendants. When an auto attendant message is recorded on any TalkSwitch unit, it is automatically copied to all the other units on the same LAN. This feature helps reduce network traffic and allows the system to continue functioning even if a unit loses power or is disconnected from the LAN.

2.4.2.2 Record, play or erase auto attendant messages

You can record an auto attendant from any local extension or remote phone using touchtone commands.

1. Pick up a local extension or dial into TalkSwitch from an external source. At the auto attendant:
2. Press **#** to enter command mode.
3. Enter the administrator password if required.
4. Begin entering the touchtone commands listed below and follow the TalkSwitch instructions.
5. Press **#** to confirm the entry before continuing with other commands or exiting command mode.
6. Press ***** or hang up to exit.

Example: From the local extension, pick up the handset and press **#** to enter command mode. If you want to record *Auto Attendant 1*, dial **4 1 #** and follow the prompt. Press **5 1 #** to listen to your recording. To record auto attendant 2, dial **4 2 #** and so on.

Record Instructions (button)

Click this button to display a dialog box showing the instructions on how to record and play an auto attendant using a telephone. The unit adjusts the recording volume for optimal recordings.

Figure 79: Recording instructions



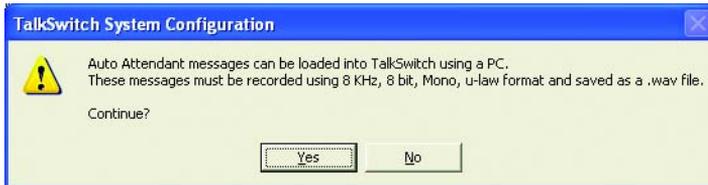
What should an auto attendant say?

Example: “Welcome to the ABC company. If you know your party’s 3-digit extension you may dial it now. To reach our receptionist, press 0 or stay on the line.”

Load Auto Attendant (button)

Click this button to display a dialog box with instructions on how to load an auto attendant using a pre-recorded .wav file. If you want to have auto attendants professionally recorded, make sure to specify that the files have to be 8KHz, 8bit, mono, and saved as a μ -law format .wav file.

Figure 80: Audio file format



Failure to record or load an auto attendant will cause your office phone to answer with no message. The caller will hear silence and might assume that the call did not get connected.

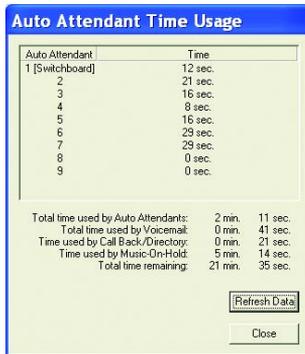
Erase Instructions (button)

Click this button to display a dialog box showing the instructions on how to erase an auto attendant using a telephone. It is a good idea to erase unused auto attendants. They share the memory with voicemail.

Figure 81: Erase auto attendant



Figure 82: View auto attendant time usage



View Auto Attendant Time Usage

To view the amount of time used by all auto attendants, go to the main menu: **Tools -> Memory Usage -> Auto Attendant.**

You may have unused auto attendant messages. If you delete them, you can free up memory.

2.4.2.3 Select routing options for each auto attendant

By default, TalkSwitch waits 1.5 seconds after your caller enters a digit before following with another digit. If no other digit is received, TalkSwitch follows the action set for the single digit. This allows caller to use 1 for the auto attendant option and 111 as an extension number.

The caller can select the single-digit options 0, 1, 2, 3, 4 or 5.

If the caller selects 0 then:

The 0 option is used to direct callers to a receptionist. You may want to configure this option to ring a local extension. For an *After Hours* auto attendant, use this option to forward calls to a remote extension for emergency purposes or direct callers to a voice mailbox.

If the caller selects 1, 2, 3, 4 or 5 then:

TalkSwitch can perform one of several actions in the table below at the selected auto attendant.

perform no action	If you don't want to use a particular option, set it to perform no action.
go to voice mailbox	You can configure any single-digit option to send a caller immediately to voicemail. Select the appropriate mailbox.
go to local extension	You can provide a single-digit option to ring a particular extension.
go to remote ext.	You can provide a single-digit option to ring a particular remote extension.
go to VoIP location	Select a VoIP location in the 250-299 range and TalkSwitch will forward the call automatically to the specified VoIP location.
go to ring group	You can provide a single-digit option to ring an extension ring group.
go to auto attendant	TalkSwitch can direct callers to multiple levels of auto attendants. For example, you might want to use your main auto attendant to provide language options like <i>"press 1 for English, 2 for Spanish."</i> In this example, you configure option 1 to go to auto attendant 2 (English version) and option 2 to go to auto attendant 3 (Spanish version).
queue at ring group	This directs calls to a group of extensions and keeps callers on hold until a member of the ring group is free to take the call.
play announcement	You can set any unused voicemail to play announcement only. Announcements are useful for providing directions, business hours and/or other information. Inform callers that they can press * to return to the auto attendant or you can program a 0 option to route to any other extension or greeting.
go to directory	If you want to enable callers to locate employee extension numbers by last name, select this option.

2.4.2.4 Automatic fax detection

If a fax call is detected then:

TalkSwitch can automatically detect incoming faxes and route them to your fax machine. To do this, the auto attendant must answer all incoming calls and listen for the CNG tone emitted by the incoming fax call. If a CNG tone is detected within nine seconds, the call is identified as a fax. If no CNG tone is detected, the auto attendant will wait for the caller to dial the appropriate digits or hang up.

Some older fax machines do not emit a CNG tone and the auto attendant cannot detect them as an incoming fax. Also, if your auto attendant volume is set too high, or if your incoming telephone lines are noisy, the auto attendant may not be able to detect the incoming CNG tone.

If you do not want to use fax detection, have the unit perform no action.

If you want automatic fax detection, configure your incoming fax line to route faxes to extension 118. The default configuration assumes that your fax machine is attached to extension 118 (113 for units with four extension jacks).

2.4.2.5 No selection was made at the auto attendant

After the auto attendant has finished playing and no selection was made within a specified number of seconds, the unit can route the call to any extension including a fax machine, play an auto attendant or announcement, go to a voice mailbox, go to the dial-by-name directory, queue to a ring group or simply hang up.

2.4.2.6 Additional features at the auto attendant

The digits 6, 7, 8 and 9 are not available as programmable options. They are used for other system features.

If 6 is dialed

If you press 6 at the auto attendant, you are able to configure and use auto call back and prompted call back features. See *2.5 Call Back/Call Bridge* on page 102 for more details about configuring call back and *3.11.2 Using call back* on page 152 for details on using call back.

If you dial 6 at the auto attendant and *Call Back* is not enabled, TalkSwitch answers with the prompt: *"I am sorry, there is no matching entry for these digits"*.

If 7 is dialed

TalkSwitch uses 7 to precede PBX extensions of another Centrex or PBX system. If TalkSwitch is not connected to another system, 7 is inactive.

If 80–88, or 9 is dialed

If you press 80–88 or 9 at the auto attendant, you can access *Call Bridge* if it has been enabled. See *2.5 Call Back/Call Bridge* on page 102 for more details about configuring call bridge and *3.11.1 Using call bridge* on page 152 for details on using call bridge.

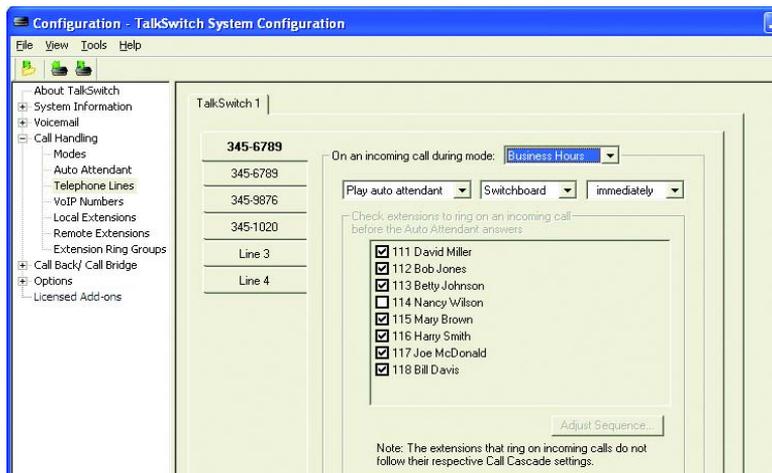
2.4.3 Incoming telephone Lines

This section describes how TalkSwitch handles incoming calls based on the incoming telephone line and distinctive ring numbers.

Configure incoming calls during mode 1, 2 and holiday mode

Each incoming telephone line has to be set to ring extensions and/or answer incoming calls using one of the auto attendants. In this example, we will use *Auto Attendant 1* and *Mode 1* to answer all incoming telephone lines. If you are not using *Mode 2* or *Holiday Mode*, you don't need to configure the settings for those modes.

Figure 83: Call handling telephone lines



1. Open the TalkSwitch configuration software.
2. Select **Call Handling -> Telephone Lines**.
3. Choose *Line 1*. Each additional incoming telephone line can be configured in succession, following steps 5 and 6.
4. Select *Business Hours*.
5. From the drop-down menus, select *Play auto attendant Switchboard and immediately*. This will pick up all incoming calls on Line 1 and play your recorded greeting.
6. Repeat the same setting for each of your incoming telephone lines. Start at step 4 and choose *Line 2* and apply steps 5 and 6.
7. Save settings to the unit.

2.4.3.1 Incoming telephone line call handling options

The other configuration options are described below.

Ring Extensions only (no auto attendant is played)

If you don't want the auto attendant to answer calls on this incoming telephone line, select this option. All the local extensions are set to ring immediately during an inbound call by default. No *call cascade* actions (i.e. voicemail) can be performed if you select this option.

Select the local extensions you would like to ring. If you require certain extensions to ring in a sequence (i.e. one extension rings first, then another one on the second ring), click the **Adjust Sequence** button to configure.

If you call on any incoming telephone line, TalkSwitch answers after 15 rings with a generic auto attendant so that you can still access the system to make configuration changes, change modes, access voicemail or dial extensions.

Go to voice mailbox

If you do not want to use an auto attendant but want the call to go to a voice mailbox, select the appropriate mailbox. The caller always hears one more ring than the number of rings you select. TalkSwitch uses the first ring to determine the ring type and to support Caller ID information.

Remember: Callers are only directed according to settings for an auto attendant. Calls do not follow the call cascade settings of any of the local extensions programmed to ring before the auto attendant answers.

Adjust Sequence... (button)

This section controls how the extensions will ring before the auto attendant answers the call.

- If you have chosen the auto attendant to answer calls immediately, this section does not require configuration and is grayed out.
- If the auto attendant is set to answer after 1 or more rings, configure when specific extensions ring before the auto attendant answers.
- If no auto attendant is selected to answer on the specified incoming telephone line, specify when particular extensions will ring up to 9 times. Remember that no calls go to voicemail unless you first have an auto attendant answer.

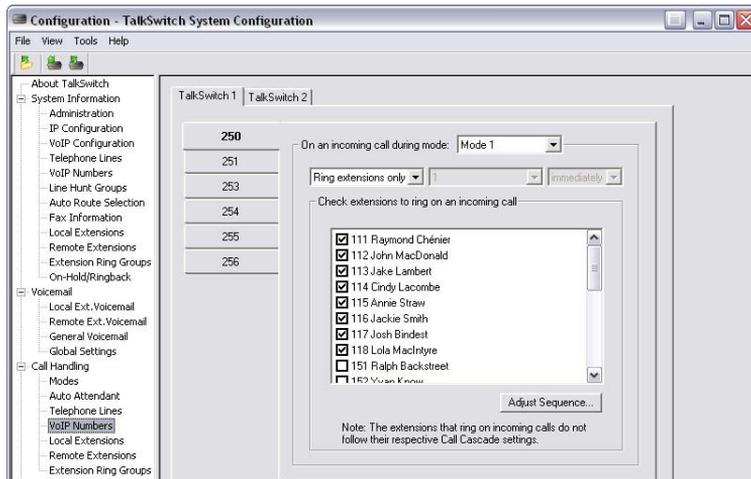


2.4.4 VoIP Numbers

This section deals with how VoIP-enabled units handle incoming VoIP calls. For information on VoIP-enabled numbers, please refer to *Chapter 5: VoIP Information* on page 161.

1. Select a VoIP number to configure.
2. Select the mode for which you want TalkSwitch to use these settings. If you entered the phone number in **Call Handling** -> **VoIP Numbers**, the number will display on the corresponding button.

Figure 84: VoIP numbers



Descriptions for the *Ring extensions only*, *Play auto attendant*, and *Go to voice mailbox* options, as well as the **Adjust Sequence** button are the same as in 2.4.3.1 *Incoming telephone line call handling options* on page 91.

2.4.5 Call Cascade

2.4.5.1 Local extensions

The call cascade feature lets you control routing options for each call to local (analog or IP), remote or ring group extensions.

This section shows how to configure the incoming call cascade options for the local extensions. The call cascade routing options are engaged if the extension is selected by the caller, or if a call is manually transferred to the extension. Once a call enters an extension's call cascade sequence, the extension directs the call. It does not follow any other extension's options, even if that extension is selected in the cascade sequence. What you see in an extension's cascade sequence is exactly what you'll get. You can set up the call cascade sequences differently for each scheduling mode.

You can customize each extension individually with a wide array of features. It is possible to configure up to three levels of call cascade for each extension.

Example: If a call reaches your desk and you are away, you can have the call sent to a partner at another local extension. If the partner's extension does not pick up, the call could be routed to your cell phone. If you do not answer, the call can be returned to your office voicemail.

Start with the steps below. You can modify the call handling options any time. The examples in 2.4.5.4 *Call cascade examples* on page 100 provide a starting point.

1. Select **Call Handling -> Local Extensions**.
2. Select the local extension for which you want to set up call cascade.
3. Select one of the two modes. Call handling can be set up differently during business hours and after hours.
4. Click on the tab to modify the call handling options for the given situation (e.g. when extension is *Busy* or *No Answer*, etc.)

Figure 85: Local extension call cascade

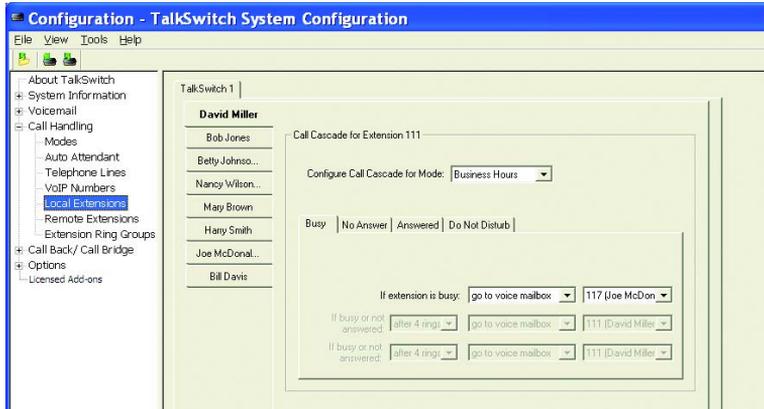


Figure 86: Local extension call cascade with IP extensions



Figure 86 above shows the **Call Handling -> Local Extensions** section with additional IP extensions enabled using TalkSwitch additional IP extensions, as described in *2.7 Licensed Add-ons* on page 120. The additional IP extensions are treated the same as a local extension.

If a call is transferred manually to a local extension that is busy or not answered, TalkSwitch follows the call cascade options for that extension.

There are four different call cascade situations for local extensions.

1. *Busy*
2. *No Answer*
3. *Answered*
4. *Do Not Disturb*

For each of these situations, three cascade destinations are available to control how you want to handle the incoming call to the extension.

1. *Busy* tab

Cascade options for forwarding calls:

- *go to voice mailbox*
- *go to local ext.*
- *go to remote ext.*
- *go to ring group*
- *play announcement*
- *invoke Call Waiting*
- *go to auto attendant*
- *queue call*
- *play busy tone*
- *hang up*
- *go to VoIP location*

For more information on these features, see *Using TalkSwitch* on page 125.

2. *No Answer* tab

Select the number of rings and select one of the actions. The action drop-down list is the same as for the *Busy* tab except that *play busy tone* is replaced by *keep ringing* and there is no *invoke Call Waiting* and *queue call*.

3. *Answered* tab

Select *play caller's name first* to prompt callers to record their name at the auto attendant. When you answer the call at the extension, the caller's name is announced. At this point, you can accept the call by pressing **#** or reroute the call to the cascade options on this tab by pressing ***** or by hanging up.

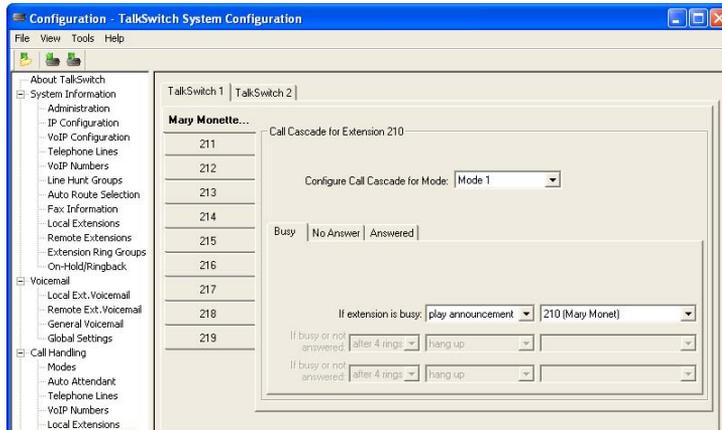
Routing/forwarding options for rejected calls:

- *go to voice mailbox*
 - *go to local ext.*
 - *go to remote ext.*
 - *go to ring group*
 - *play announcement*
 - *go to auto attendant*
 - *hang up*
 - *go to VoIP location*
4. When *Do Not Disturb* is enabled, you will hear: “*Do not disturb on*” before the dial tone as a reminder that calls will not ring at your extension.

2.4.5.2 Remote extensions

Remote extension call cascade works principally the same way as the local extension call cascade. Please refer to 2.4.5.1 *Local extensions* on page 93. In step 1, select **Call Handling -> Remote Extensions**.

Figure 87: Remote extension call cascade



For remote extensions, there are three different call cascade options. For each of these options, you can control how you want to handle the incoming call to the extension.

The remote extension call cascade does not have a *Do Not Disturb* option.

1. *Busy* tab

Cascade options for forwarding calls:

- *go to voice mailbox*
- *go to local ext.*
- *go to remote ext.*
- *go to ring group*
- *play announcement*
- *go to auto attendant*
- *play busy tone*
- *hang up*
- *go to VoIP location*

For more information on these features, see *Using TalkSwitch* on page 125.

2. *No Answer* tab — routing options if this extension is not answered after a specified number of rings

Select the number of rings and select one of the actions.

For remote extensions it is very important to allow enough time for the telephone network to connect to the phone. For example, a call to a cell phone or PDA may take 12 seconds (i.e. 2 rings) just to connect.

The action drop-down list is the same as for the *Busy* tab except that the option *play busy tone* is replaced by *keep ringing* and there is no *queue at ring group* option.

3. *Answered* tab — routing options when this extension answers or rejects a screened call

Selecting *play caller's name first* prompts callers to record their name at the auto attendant. When you answer the call at the extension, the caller's name is announced. At this point, you can accept the call by pressing **#** or reroute the call to the Cascade options on this tab by pressing *****. If you hang up or do not answer, the call will be routed to this cascade but it may take several seconds more for the system to redirect the call.

Routing/forwarding options for rejected calls:

- *go to voice mailbox*
- *go to local ext.*
- *go to remote ext.*
- *go to ring group*
- *play announcement*
- *go to auto attendant*
- *hang up*
- *go to VoIP location*

If you select *stay connected*, the call cascade options are grayed out. TalkSwitch can forward calls when prompting is enabled. If you use one of the prompted methods for this remote extension, enable *Music on Hold* and do not use 3-way calling to forward calls to this remote extension, the call cascade is seamless to the callers. They hear music throughout the entire process of trying to locate you.

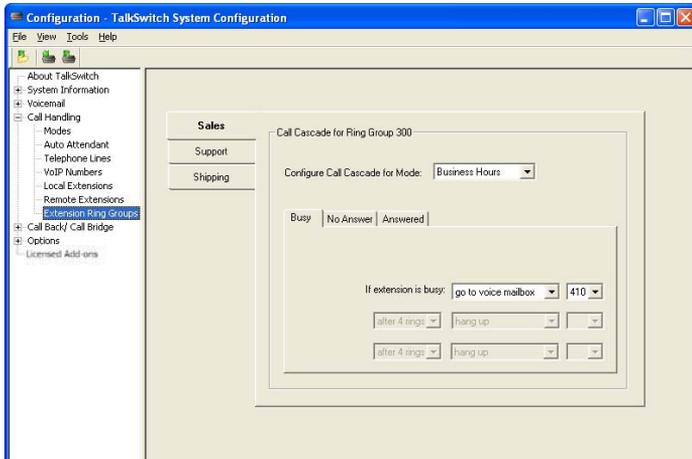
If a call is forwarded to a remote extension from a call cascade sequence and has *play caller's name first* or *play accept/reject* prompt enabled, the remote extension hears: "This is Call Cascade..." instead of "This is Call Forward...". This indicates to you that the call is rejected and sends the caller to the next call cascade option of the extension that forwarded the call. It does not follow the remote extension's answer call cascade options.

2.4.5.3 Extension ring groups call cascade

Extension ring groups — call cascade is similar to local and remote call cascade. For more information, see [2.4.5.1 Local extensions](#) on page 93 and [2.4.5.2 Remote extensions](#) on page 96.

The call cascade routing options are engaged if the ring group is selected from the auto attendant or if a call was transferred manually to the ring group. Once a call enters a ring group's call cascade sequence, the ring group directs the call. It does not follow any other extensions options, even if that extension is selected in the call cascade sequence. What you see in a ring group's call cascade sequence is exactly what you'll get.

Figure 88: Extension ring group call cascade



For ring groups, like for remote extensions, there are 3 different call cascade options. For each of these options, you can control how to handle the incoming call to the extension.

1. *Busy* tab — routing options when this extension is busy
Cascade options for forwarding calls:

- *go to voice mailbox*
- *go to local ext.*
- *go to remote ext.*
- *go to ring group*
- *play announcement*
- *go to auto attendant*
- *queue at ring group*
- *play busy tone* (example, if a fax machine is at the extension)
- *hang up*
- *go to VoIP location*

For more information on these features, see *Using TalkSwitch* on page 125.

2. *No Answer* tab — routing options if this extension is not answered after a specified number of rings

Select the number of rings and select one of the actions. The action drop down list is the same as for the Busy tab except that the option *play busy tone* is replaced by *keep ringing*.

3. *Answered* tab — routing options when this extension answers or rejects a screened call

Select *play caller's name first* to prompt callers to record their name at the auto attendant. When you answer the call at the extension, it announces the caller's name. At this point, you can accept the call by pressing **#** or reroute the call to the Cascade options on this tab by hanging up or pressing *****.

Routing/forwarding options for rejected calls:

- *go to voice mailbox*
- *go to local ext.*
- *go to remote ext.*
- *go to ring group*
- *play announcement*
- *go to auto attendant*
- *hang up*
- *go to VoIP number*

2.4.5.4 Call cascade examples

If my extension is busy

Figure 89: Example 1 — Extension busy

TalkSwitch 1

David Miller
Bob Jones
Betty Johnson...
Nancy Wilson...
Mary Brown
Harry Smith
Joe McDonal...
Bill Davis

Call Cascade for Extension 115

Configure Call Cascade for Mode: After Hours

Busy | No Answer | Answered | Do Not Disturb

If extension is busy: go to local ext. | 111 (David Miller)

If busy or not answered: after 4 rings | go to voice mailbox | 115 (Mary Brown)

If busy or not answered: after 4 rings | go to voice mailbox | 115 (Mary Brown)

If I am away from my desk

Figure 90: Example 2 — Away from the desk

TalkSwitch 1

David Miller
Bob Jones
Betty Johnson...
Nancy Wilson...
May Brown
Harry Smith
Joe McDonal...
Bill Davis

Call Cascade for Extension 115

Configure Call Cascade for Mode: After Hours

Busy | No Answer | Answered | Do Not Disturb

If this extension is not answered: after 4 rings go to local ext. 111 (David Miller)

If busy or not answered: after 4 rings go to ring group 300 (Documents)

If busy or not answered: after 4 rings go to auto attendant Switchboard

Screening calls from a remote extension

Figure 91: Example 3 — Screening calls

TalkSwitch 1

Cell phone
Donald Duck
David's office...
Ed Anderson
Allan Williams...
Doug White
Charlie Brown...
Bea Thompson...
kay itkhj

Call Cascade for Extension 210

Configure Call Cascade for Mode: Business Hours

Busy | No Answer | Answered

When a call is answered: play accept/reject pro

If a call is rejected from this ext.: go to local ext. 115 (May Brower)

If busy or not answered: after 4 rings go to voice mailbox 116 (Harry Smith)

If busy or not answered: after 4 rings hang up

If I don't want to be disturbed (only local extensions)

Figure 92: Example 4 — Do not disturb

TalkSwitch 1

David Miller
Bob Jones
Betty Johnson...
Nancy Wilson...
May Brown
Harry Smith
Joe McDonal...
Bill Davis

Call Cascade for Extension 111

Configure Call Cascade for Mode: Business Hours

Busy | No Answer | Answered | Do Not Disturb

Do not Disturb can be enabled or disabled by dialing *62 from this extension.

If Do Not Disturb is on for this ext.: go to local ext. 115 (May Brower)

If busy or not answered: after 4 rings go to voice mailbox 111 (David Miller)

If busy or not answered: after 4 rings go to voice mailbox 111 (David Miller)

2.5 CALL BACK/CALL BRIDGE

Call Back and *Call Bridge* work together as your personal long-distance operator. Whether you are across town or around the world, you can place calls from your office telephone or VoIP number(s) and access your long-distance savings plan.

You can make a direct call to TalkSwitch, activate call bridge and dial the telephone number you want. TalkSwitch dials the number from your office telephone or VoIP trunk and connects the call you made to TalkSwitch and the call TalkSwitch made from your office to the dialed party.

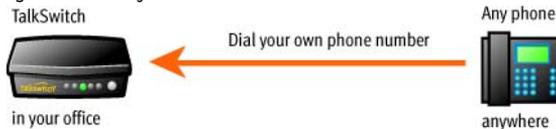
Two call back methods

- *Auto Call Back* lets you program the number where you can be reached as the call back number when you travel. Once you are at your destination, you can call your TalkSwitch location, let it ring twice and hang up. TalkSwitch will call you back.
- *Prompted Call Back* lets you change the call back number and message each time you initiate the call back. If you are traveling to more than one destination, or you don't know the telephone number, you can change the call back number once you arrive at your destination.

How it works

1. From your out-of-office location, dial one of the call back telephone numbers that you have pre-configured on your TalkSwitch unit.

Figure 93: Dial your own number



2. Let the line ring at least once, but no more than three times and hang up. Make sure there is no auto attendant configured to answer after less than 3 rings on this line.
3. Within less than 10 seconds, TalkSwitch will dial your auto call back number. When you pick up, your recorded message will play.

Figure 94: TalkSwitch calling back



4. Accept the call by pressing **#**. Enter your password. This confirms to TalkSwitch that you are an authorized user and enables call access through the TalkSwitch.
5. You will now hear a generic TalkSwitch auto attendant. You may access your phone system as if you were in the office. Dial 9 for an outside line and make a long-distance call using your office long-distance plan, listen to your voicemail or transfer to a local extension.

Figure 95: Multiple calls to anywhere



2.5.1 Auto call back

The TalkSwitch call back/call bridge feature gives you access to your office-based long-distance savings plan from anywhere around the world.

Before call back is activated, all the call back settings must be configured. To use auto call back, dial your office telephone number, let the call ring twice and hang up.

You can configure four auto call back numbers.

1. Select **Call Back/Call Bridge -> Auto Call Back**.
2. Select *Call Back 1*. You can add more as needed.
3. Click the checkbox next to *Activate Auto Call Back 1*. In the field *Number to call back:*, enter the call back number exactly as you would normally dial the number. Do not enter a line hunt group in the dialing string.
4. Click the checkbox next to *Use dialing prefix:*, if you have to dial the same prefix every time you use auto call back (i.e. long-distance calling card number). Enter the dialing prefix including any required pauses or feature keys like ***X** or **#**. Use a comma to insert a 2-second pause.

Example: If you want to disable call waiting and want to dial a long-distance prefix before dialing the long-distance number, enter the following: *70,1010555,,. This corresponds to ***X 70** for disabling the call waiting feature, a 2-second pause to wait for the new dial tone, 1010555 for the long distance prefix, and a 4-second pause to wait for the new dial tone before dialing the long-distance phone number.

Use Announced Message:

We recommend this option if the number TalkSwitch calls back is to a hotel or a location where a third party answers the call. The *Announced Message* option allows you to record a message that plays when the call is initially answered during a call back.

Example: “Please transfer this call to Bob in room 307”. The announced message can be recorded locally using an extension handset, or remotely by accessing call back from the auto attendant. The **Record Message** button gives the following instructions:

5. Pick up a telephone handset connected to the unit and use the keypad commands to record the announced message.
 - a) Press **#** on an analog phone, or *** 55** on a IP set to enter command mode. Enter the password if required.
 - b) Enter the appropriate command as listed below.

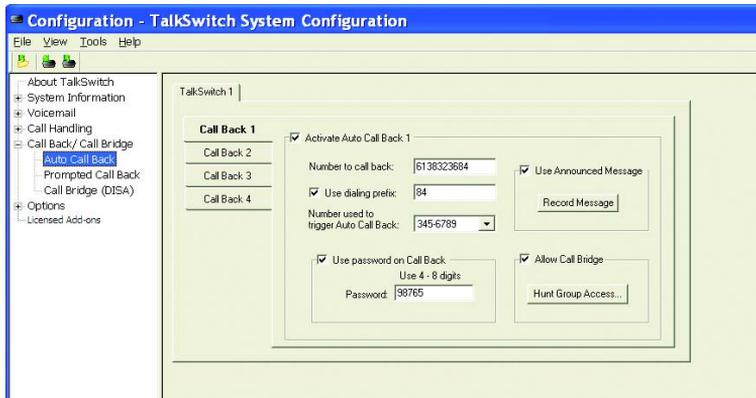
For auto call back	Record message	Play message back
1	61 #	71 #
2	62 #	72 #
3	63 #	73 #
4	64 #	74 #

6. In the field next to *Number used to trigger Auto Call Back:*, enter the home/office number that you will call to trigger the auto call back. The drop-down list allows you to select any of the incoming telephone lines connected to your unit.

There are four auto call back accounts per unit. If you need to record an announced message for a particular auto call back account, it must be on the same unit as the account.

If you need to configure and/or record an announced message from a remote location and do not have the auto attendant set up to answer this call back line, you can call your call back number and let it ring 15 times. A generic auto attendant will answer. You can then configure call back by pressing **6**.

Figure 96: Configuring auto call back



7. To prevent unauthorized use of your call back feature, click the checkbox next to *Use password on Call Back* and enter a password.
8. Click the checkbox next to *Allow Call Bridge*, to allow call bridge after a call back has been accepted. You may restrict the call bridge line access by selecting the **Hunt Group Access** button. Call bridge is enabled when call back is activated with the same password you entered at the auto attendant.

When programming the phone number remotely, for a 2-second pause, dial *** 0** and for a **#**, dial *** #**. When you have completed the entire dial string, press **#**.

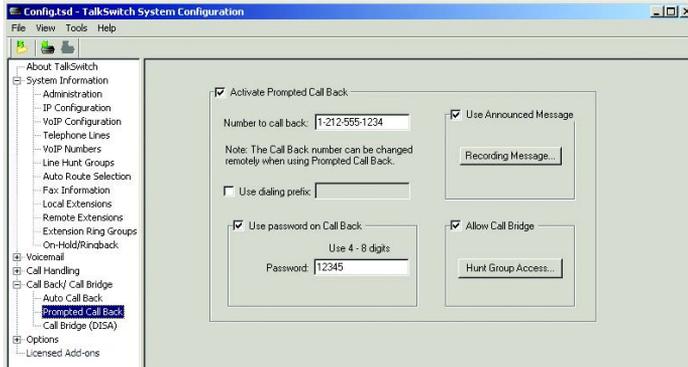
2.5.2 Prompted call back

Prompted call back works much the same way as auto call back, except that you can change the call back number remotely. The call back number is stored in TalkSwitch until you change it. This means that you can change your call back telephone number and message each time you activate call back.

If you are travelling to a location where you do not yet know the phone number, you can use prompted call back to save on your long-distance charges. For details on using call back, see *3.11.2 Using call back* on page 152.

You need to activate prompted call back before you can use this feature. The call back out of the system to the number programmed always uses the same line that was dialed.

Figure 97: Configuring prompted call back



1. Select **Call Back/Call Bridge -> Prompted Call Back**.
2. Click the checkbox next to *Activate Prompted Call Back*.
3. Enter the number TalkSwitch will call (e.g. your hotel telephone number). If you require a dialing prefix or long-distance discounts, check the *Use dialing prefix* box and enter the prefix.
4. If you expect the call back from TalkSwitch to be answered by a receptionist before it gets to you, you can have the unit play a recorded announcement when someone answers the call back. This is useful if you need the call transferred to a hotel room. The message could say: *"This is a call for Bob in room 587. Please transfer the call"*.

When you access prompted call back from out of the office, you may change this number before initiating the call back. The benefit of prompted call back is that you do not need to set the number before you leave the office.

Accessing and using prompted call back from a telephone

1. Dial your office telephone number.
2. At the auto attendant, dial **6** to access the call back settings.
3. Follow the prompts.

- a) Enter the telephone number where you would like TalkSwitch to call you.
- b) Record an announced message if required. Call back is activated right after you initiate prompted call back and hang up. A small initial long-distance charge is incurred at your current location when you activate prompted call back.

Press **#** on an analog phone or *** 55** on a IP set to enter command mode. Enter the password if required.

Enter the command as listed below.

To record a message:	65 #
To play the message:	75 #

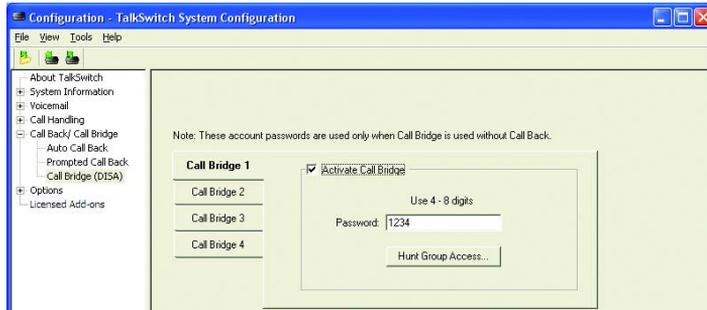
The call back number shown is always the last number the unit dials in the last prompted call back session.



2.5.3 Call bridge

Call bridge allows you to use the TalkSwitch incoming telephone lines to make outbound calls or transfers to local and remote office extensions. It gives you access to your office-based long-distance savings plan from anywhere around the world. You will first need to call into the unit from an outside phone number .

Figure 98: Activating call bridge

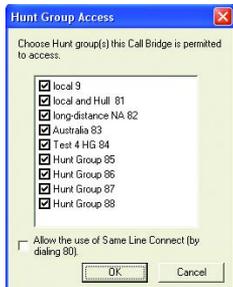


How to use call bridge:

1. Call into the system from any out-of-office telephone.
2. At the auto attendant message, dial a hunt group you are authorized to use (9 is default).

3. Enter the password and press **#** to access a TalkSwitch incoming telephone line. Each TalkSwitch unit has four password activated call bridge accounts.
4. At this point, you can dial a number the same way as from the office. Each account can be configured to allow access to certain hunt groups when using call bridge.

Figure 99: Hunt group access



To activate an account, check the *Activate Call Bridge* box and enter a 4- to 8-digit password. Click the **Hunt Group Access** button and assign the hunt groups for this account.

Line hunt group 80 is used for *Same Line Connect*. To enable line hunt group 80, click the checkbox next to *Allow the use of Same Line Connect* [by dialing 80].

2.6 OPTIONS

2.6.1 Permissions

TalkSwitch can be configured to permit or block outgoing calls made from specified extensions. On the restricted extensions users will be required to enter a password (PIN code) prior to making an outside call.

For example, outgoing calls from a lobby phone can only be made by users with assigned PIN codes.

Use the Permissions window to specify users, their PIN codes, and which extensions restrict outgoing calls.

To configure permissions:

1. Select **Options -> Permissions**.
2. Check the *Activate Outgoing Call Access* checkbox.
3. Click the *Add User* button at the bottom of the window. An **Add User** dialog window will appear. See *Figure 101* on page 109 and *Figure 102* on page 110.

Figure 100: Permissions to access outside line

Outgoing Call Access

Access Codes can be enabled for up to 150 people. When enabled on an extension, an access code is required to make outgoing calls. Emergency service numbers will not be blocked.

Access Code Assignments		Extensions Requiring Access Code
Name	Access Code	
Harry Smith	1234	<input type="checkbox"/> 111 David Miller
Joe McDonald	4321	<input type="checkbox"/> 112
		<input type="checkbox"/> 113
		<input type="checkbox"/> 114
		<input type="checkbox"/> 115 Mary Brown
		<input checked="" type="checkbox"/> 116 Harry Smith
		<input checked="" type="checkbox"/> 117 Joe McDonald
		<input type="checkbox"/> 118
		<input type="checkbox"/> 151
		<input type="checkbox"/> 152
		<input type="checkbox"/> 153
		<input type="checkbox"/> 154
		<input type="checkbox"/> 155
		<input type="checkbox"/> 156
		<input type="checkbox"/> 157
		<input type="checkbox"/> 158

Modify... Remove Add User...

Figure 101: Add user

Add User

Select an existing user on the system or enter a name.

Name: David's office

- Allan Williams
- Bea Thompson
- Betty Johnson
- Bill Davis
- Bob Jones
- Cell phone
- Charlie Brown
- Charlie Miller
- David's office
- Donald Duck
- Doug White

Access Code: 3876
(4 - 8 digits)

4. In the **Add User** window, enter the name or select one from the menu. Enter a unique 4-8 digit number in the *Access Code* field (this will be the user's PIN code). Click OK.
5. Repeat step 4 for each user that will be assigned a PIN code.
6. On the right side of the window, use the checkboxes to select the extensions that will restrict outgoing calls.
7. Save the configuration to TalkSwitch.

To modify a user's name or access code:

1. Click on the user's name in the **Access Code Assignments** window.

Figure 102: Modify user access



2. Click the *Modify* button. A **Modify** dialog window will appear (see *Figure 102*).
3. Edit the user information as required. Click OK.
4. Save the configuration to TalkSwitch.

To remove a user from the permissions list:

1. Click on the user's name in the **Access Code Assignments** window. Click the Remove button.

Permissions and Call Detail Records

With Permission enabled, CDR records will not display the password code for calls made from restricted extensions, but the name associated to the code.

Password Protection

It is recommend that the administrator password protection is enabled when Permissions are used. Enable the password protection in the TalkSwitch configuration software under System Information -> Administration and enter a 4- to 8-digit numeric password.

Change Password Frequently

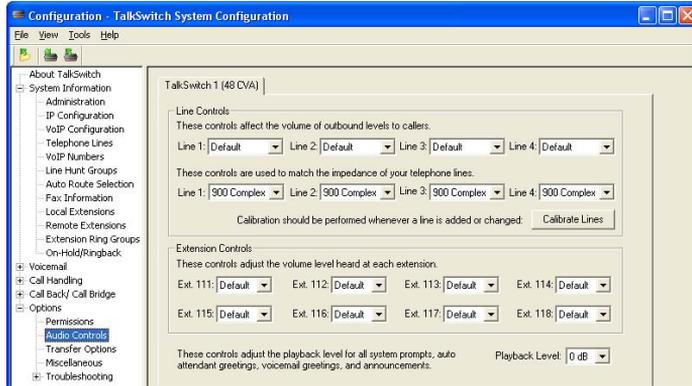
Change the administrator password frequently to prevent unauthorized changes to Permissions.



2.6.2 Audio Controls

This section deals with the adjustment of volume and system gain controls. All the controls on this page update the unit in real time. This allows you to adjust levels during a conversation without having to first save the configuration to the unit.

Figure 103: Adjusting audio controls



Line Controls

If you need to adjust the volume of conversations through the system, you can adjust each line individually. The default settings compensate for loss created by TalkSwitch and a portion of the loss created by your lines. We recommend that you adjust the level for each line separately, as the characteristics of each line may vary. These controls only affect the volume heard by the caller.

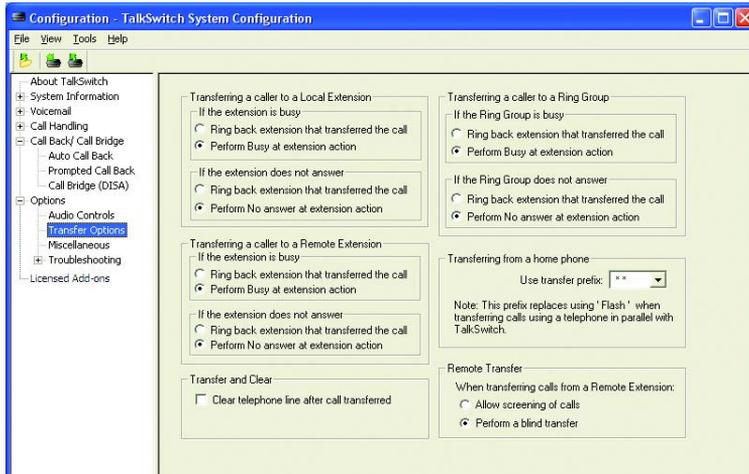
Extension Controls

These controls allow you to increase or decrease the volume level of calls and system prompts heard at each extension.

2.6.3 Transfer Options

This section describes the TalkSwitch actions when calls are transferred manually.

Figure 104: Transfer options



Transferring a caller to a local or remote extension or a ring group

You can specify how transferred calls are handled if their target is a local extension, remote extension or ring group. If the extension is busy or doesn't answer, you have the choice to have the unit ring back the extension that transferred the call or perform the *no answer* or *busy* action of the extension that the call was intended for. Any changes made in this section affect the entire system.

Transferring from a home phone

You can transfer a call from a phone connected in parallel to one of the incoming telephone lines.

Example: You can transfer a call from your home phone (for a home-based business) to a local or remote extension, ring group, an announcement or a voice mailbox.

This section allows you to choose which transfer prefix you would like to use to perform the action. The transfer prefix wakes up TalkSwitch and is considered the equivalent of the **Flash** or **Link** action at a local extension. The default prefix is *** * ***.

Example: You receive a call on a phone in your home that is not a TalkSwitch extension and you want to transfer the caller to remote extension 215. You dial *** ***, wait for the double beep, dial **215** and hang up. TalkSwitch now handles the call.

Transfer and clear

TalkSwitch has the ability to clear lines after a call has been transferred. Ensure you have this service/feature activated on your line(s). Check under **System Information -> Telephone Lines**.

Remote transfer

Calls that are transferred to a remote or VoIP extension can be transferred back to another extension or voice mailbox.

1. Dial *** *** to place the caller on hold.
2. Enter the extension or mailbox number. If the extension is set to screening calls, dial *** * 4** to complete the transfer. If you want to cancel the transfer, dial *** * 5**. If you have selected *Perform a blind transfer*, the call is automatically transferred and the line is released once the extension or mailbox number has been entered.

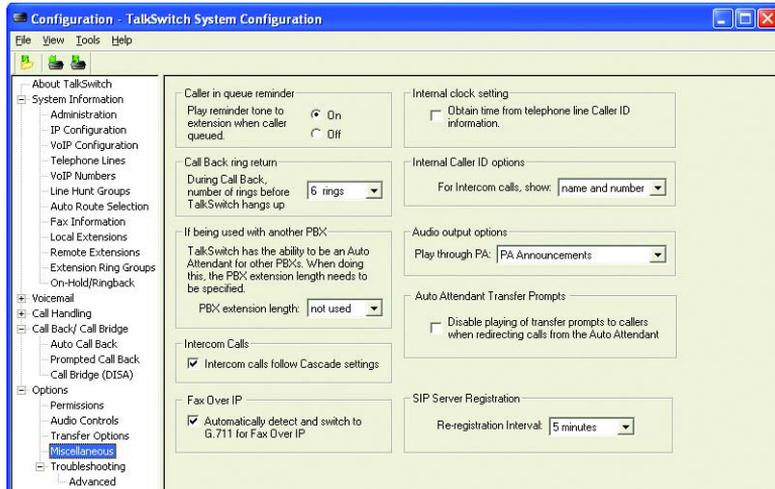
When a call is transferred from one remote extension to another remote extension or VoIP extension, it is always a blind transfer so that the three lines are not tied up.

This feature can only be used when forwarding a call to a remote extension or a VoIP location over two lines. It cannot be used with 3-way calling or Centrex transfer services.

2.6.4 Miscellaneous

All of the following options affect TalkSwitch as a system. Adjust them according to your needs.

Figure 105: Miscellaneous



Caller in queue reminder

When a caller is in queue at an extension that is already in use, a reminder tone plays at the extension to let them know they have a caller in their queue. You can turn this function off by clicking the Off radio button.

Call Back ring return

You can adjust the number of rings during a call back session before TalkSwitch disconnects from call back. Depending on the situation, you may want to give yourself more time to pick up and accept the call back by increasing the number of rings.

If being used with another PBX

PBX extension length

If TalkSwitch is used as an auto attendant with another PBX, you can set the number of digits the PBX uses for its extensions. This allows TalkSwitch to know what digits to pass on to the PBX to complete the transfer.

This feature only works with PBXs that use hook **Flash** as a transfer method. Please contact your PBX manufacturer for transfer details.

Setting up TalkSwitch behind an existing PBX

1. Connect the PBX extension to one of the TalkSwitch line jacks. You may do this with all your TalkSwitch lines (1 PBX extension per line) depending upon how many auto attendant ports you would like on the PBX.
2. Configure incoming calls on your PBX to ring the extensions you have allocated to the incoming telephone lines.
3. Configure the TalkSwitch auto attendants to answer after a specified number of rings.
4. Record your auto attendants. They should list the extensions with the digit 7 in front of them.

Example: If you have chosen the PBX extension length to be 3 digits and John Smith's extension is 111 on the PBX, the auto attendant should say: *"To speak with John Smith, press 7111..."*. When 7111 is dialed, TalkSwitch puts the caller on hold and transfers the caller to extension 111.

Intercom Calls

This setting is enabled by default. Any internal call follows the cascade settings of the extension dialed. If it is disabled, no fall-through actions take place.



Fax Over IP

TalkSwitch supports fax detection and switching to G.711 for transparent fax over IP.

Internal clock setting

TalkSwitch can obtain time information from Caller ID for its internal clock. If you have indicated that you do have Caller ID on any of the incoming telephone lines, this option defaults to the active position. If it is deactivated, TalkSwitch obtains the time information from the time and date set in the *Mode* tab under *Automatic Mode Switching*. The TalkSwitch clock information is used for automatic mode scheduling and for internal Caller ID information when intercom calls are placed in the office.

Internal Caller ID options

TalkSwitch can display name and number Caller ID to each extension during intercom calls. If your Caller ID unit/telephone does not support the name option, you can have TalkSwitch send the number only.

Audio Output Options

TalkSwitch can output different audio to the PA jack. The default setting is to play only PA announcements through the PA jack by dialing ***0** from any extension.

Other options include:

- Voicemail Screening (used in a home environment)
- Demo with a router (used for demonstrating the product)
- Static VoIP Demo (used strictly for demonstrating two TalkSwitch units connected to a hub or a switch and acting as two separate locations.)

Auto Attendant Transfer Prompts

TalkSwitch can disable playing of transfer prompts to callers when redirecting calls from the auto attendant.



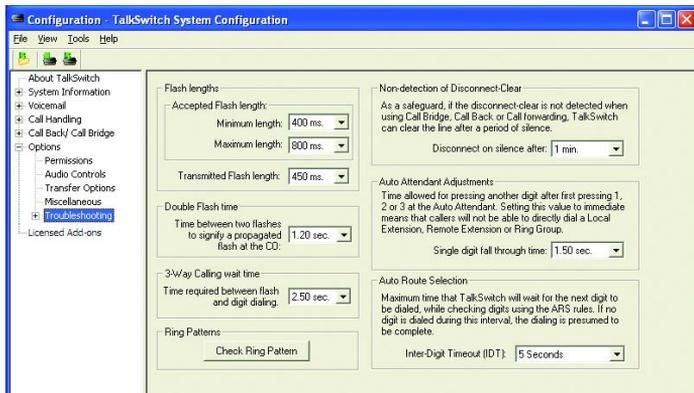
SIP Server Registration

Select the Re-registration Interval between 2 and 60 minutes from the drop-down list. The re-registration interval on your IP phones may differ, but TalkSwitch will use the time specified here. See *Figure 105* on page 110.

2.6.5 Troubleshooting

Do not change the following controls unless you are familiar with the effect they have on the TalkSwitch operation. If your system does not appear to be functioning properly, please contact your dealer for assistance.

Figure 106: Troubleshooting



Flash lengths

Accepted Flash length:

These options control the flash length allowed from a local extension that TalkSwitch considers to be an intentional flash. The range can be adjusted to accept anything from 300 ms to 1 second. The default range for accepting a flash signal at a local extension is between 400 ms and 800 ms. Most phones emit a 600 ms flash signal when the **Flash** or **Link** button is pressed.

Transmitted Flash length:

This option controls the transmitted flash length from TalkSwitch to the incoming telephone lines. TalkSwitch emits a flash when forwarding or bridging calls using the 3-Way Calling/Conference or Centrex transfer service or when transferring calls through an existing PBX system. The value can be adjusted from 250 ms to 1 second. The default setting is 400 ms. If you notice that you are having problems with any of the TalkSwitch features that use 3-way calling, experiment with this value.

Double Flash time

When two flashes are detected, TalkSwitch needs to calculate the time between the flashes to establish if they were intended to propagate a flash at the central office (CO) or to place a call on hold and then quickly retrieve it. The *Double Flash* time values signify the maximum time between two flashes that TalkSwitch recognizes as an intended flash to the CO. When two flashes are detected with a time gap larger than the value displayed, no flash is propagated at the CO. The value can be adjusted from 200 ms to 2000 ms with the default set at 1200 ms.

3-Way Calling wait time

This option adjusts the time required between hook flash and a dial string. Some telephone company 3-way calling/conference services require different time allowances after the flash signal to re-establish a dial tone. The default time is two seconds, so TalkSwitch can let the lines settle after going off-hook before sending the DTMF digits.

Ring Patterns

Click the **Check Ring Pattern...** button and call any extension. After three rings the sequence of the ring pattern displays in a new window. The time is displayed in milliseconds.

Non-detection of disconnect-clear

This option controls the length of time after which TalkSwitch drops a line when silence is heard. This safeguard is useful when a telephone line does not send out the disconnect-clear signal that tells TalkSwitch the call is completed.

Auto Attendant Adjustments

Since TalkSwitch allows callers to dial 100, 200, 300 series extensions, 400 series mailboxes and allows fall-through options for the single digits 1, 2, 3, 4 and 5 at the auto attendant, there needs to be a time limit on how long TalkSwitch waits for another digit before following the single-digit option after the first digit is pressed. There are three ways to make use of this control. Be careful not to isolate your callers from having both choices as listed in (1).

1. With the current setting, callers can dial any local extension or remote extension or ring group, as long as the second digit is pressed within 1.5 seconds of pressing the first digit. Callers can press 1, 2, 3, 4 or 5. When TalkSwitch detects no second digit after 1.5 seconds, it follows the programmed option for that auto attendant.
2. If you don't want to use options 1, 2, and 3 for any of the auto attendants, you can set the option to 4 seconds. This gives callers enough time to dial all three digits of any extension without accidentally falling through to one of the auto attendant options.
3. If you don't want to give callers the ability to dial any of the extensions and only permit them to access single-digit options, set this value to *immediately*. As soon as TalkSwitch detects the first digit, it will follow the specified option for that auto attendant.

Auto Route Selection

The *Inter-Digit time-out* can be set anywhere between 3 and 18 seconds. The default is 5 seconds.

2.6.5.1 Troubleshooting — Advanced

DTMF Detection

Should callers complain about being misdirected when they select options at the auto attendant, the following values need to be adjusted:

Minimum required time to detect a DTMF digit:

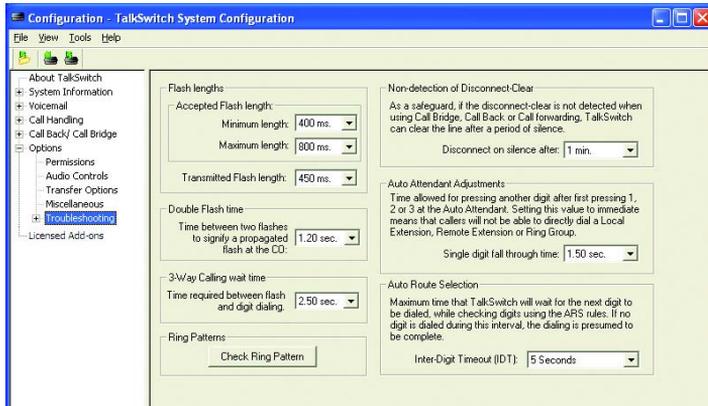
The default setting of 45 ms is adequate for most environments. It may be necessary to lower this value if TalkSwitch appears to have trouble hearing digits when dialed at the auto attendant. Increase the time if the system is accidentally misdirecting callers.

Minimum required time between DTMF digits:

The default setting of 65 ms is adequate for most environments. If there is a problem, increase the value slightly to help the system deal with echoed digits from cell phones.

Example: Calls keep being misdirected to extension 111. This usually indicates that, even though the caller may have dialed 113, TalkSwitch sees 1113 because the 1 was echoed by the cell phone network. Once TalkSwitch captures 3 digits, it ignores the rest, hence the digits 111 are accepted and the 3 is ignored.

Figure 107: Troubleshooting advanced



Line CPC/Disconnect Clear

If calls are accidentally disconnected in mid conversation, it usually indicates a problem with the telephone line. TalkSwitch is always looking for a hang-up signal, called a CPC or Disconnect Clear signal, from the phone company. If the line has power fluctuations or if the line card at the central office is defective, CPC signals in the 250 ms range might come down the line and cause TalkSwitch to hang up. Try increasing this value to correct the problem.

Ringback Control

If an extension seems to ring back as if a caller was on hold, but when answered, no one is there, it could indicate a debounce problem with the phone hook switch. When someone at the extension hangs up, the handset bounces slightly to cause a flash signal that can be seen by TalkSwitch before the hang-up. As a result, TalkSwitch puts the line on hold. To help with this problem, increase this value until it stops ringing back.

Fax Detection

If fax calls are not directed to the fax machine, you may need to set this option to *Detect 1 tone*. If it is set to *Detect 1 tone* and people complain they have accidentally been routed to the fax machine, set this value to *Detect 2 tones*.



VoIP Ports

IP Signalling Port:

The default IP signalling port is 5060. If you need to choose another port, enter a value between 1024 and 16556.

Starting RTP Port:

The starting RTP port for TalkSwitch is 6000. From this starting point, TalkSwitch needs 4 ports for each TalkSwitch unit. RTP ports are evenly numbered from the start port. 6000, 6002, 6004 and 6006 are needed for the first TalkSwitch unit on a LAN and 6030, 6032, 6034 and 6036 are used for the fourth unit.

These are the ports that this TalkSwitch system listens on — as opposed to the port it needs to communicate with at the other end.

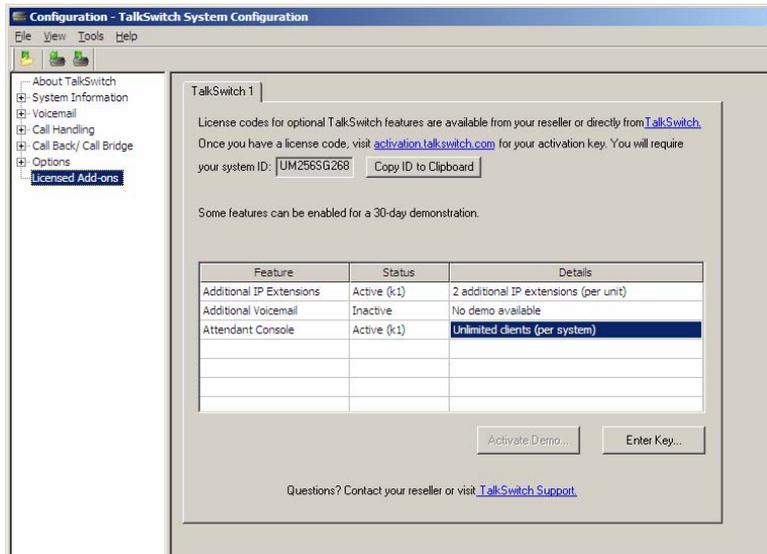
Example: If your IP server at another location uses port 5061 instead of 5060 for signalling, you do not set this port to 5061, you need to include the port number in the address specified in the **VoIP Configuration** section of the software.

Example: Proxy Server Name: 222.234.432.234:5061

2.7 LICENSED ADD-ONS

The **Licensed Add-ons** window provides functionality to enable additional IP Extensions (up to 8 per TalkSwitch unit), enable additional voicemail memory capacity, and allow Attendant Console to connect to the TalkSwitch unit.

Figure 108: Licensed Add-ons



Copy ID to Clipboard button

The unit system ID is required in order to obtain an activation key. Click this button to copy the system ID to the clipboard, which can then be pasted into the appropriate field on the TalkSwitch activation web page.

Activate Demo button

Some licensed add-ons can be activated as time-limited demos. Click this button to activate a demo.

Enter Key button

An activation is obtained after entering a licence code and a system ID on the TalkSwitch activation web page. Click this button to enter the activation key and enable the desired licensed add-on.

Additional IP extensions

It is possible to enable 8 additional IP extensions per TalkSwitch unit. Once enabled, the additional tabs will appear in the following configuration software sections and the IP extensions can be configured the same way you configure the other local extensions:

- **System Information -> Local Extensions**
(see 2.2.7 *Local extensions* on page 56)
- **Voicemail -> Local Ext. Voicemail**
(see 2.3.1 *Mailbox options* on page 70)
- **Call Handling -> Local Extensions**
(see 2.4.5.1 *Local extensions* on page 93)

Additional voicemail capacity

It is possible to increase TalkSwitch unit memory capacity for voicemail, greetings and music on hold by enabling additional capacity in the configuration software. TalkSwitch comes with a default 0.5 hours of capacity and can be upgraded to 2.5 hours, in increments of 1 hour using licence codes (if you need more than 2.5 hours you will need to purchase 4.5 or 9 hour memory cards from your reseller). When adding extensions to your system (per unit), you may find the need to add memory capacity to account for the addition of additional voicemail boxes and greetings. Verify your memory capacity information using Voicemail Manager (see 2.3.3.1 *Voicemail manager* on page 79).

Attendant Console

TalkSwitch Attendant Console is a standalone software application providing display and call control for receptionist tasks. Attendant Console may be enabled as a time-limited demo, or enabled fully with a license code, as described below.

2.7.1 Enabling Add-on demos

It is possible to enable time-limited demos for the addition of IP extensions and Attendant Console.

To enable a demo:

1. In the TalkSwitch configuration software, select **Licensed Add-ons**.
2. In the *Feature* column, click the feature to be enabled as a demo.
3. Click the **Activate Demo** button. A window appears stating the feature has been enabled.
4. Click the **Reboot Now** or **Reboot Later** button. The **Reboot TalkSwitch System** window appears if Reboot Now was selected.

2.7.2 Enabling Add-ons using an Activation Key

In order to enable TalkSwitch optional features, a License Code and the TalkSwitch unit System ID are required.

2.7.2.1 Step 1: Obtaining a License Code

1. Contact your reseller, or go to the Licensed Add-ons section at <http://shop.talkswitch.com> and purchase the feature to obtain the License Code.
2. Record your licence code. It is required to obtain an activation key in order for the feature to be enabled.
3. Perform Step 2 below with your licence code.

2.7.2.2 Step 2: Obtaining the TalkSwitch System ID

1. Select **Licensed Add-ons**. The screen appears displaying available features and activation functions.
2. Record the System ID value present in the *System ID* text box. Alternatively, click the **Copy ID to Clipboard** button to copy the System ID to the clipboard. (if going immediately to <http://activation.talkswitch.com>, the value can be pasted from the clipboard).
3. Go to <http://activation.talkswitch.com>. Licence code and system ID values are required.

4. Fill out the personal information fields and enter both license code and system ID into the appropriate fields.
5. Click the **Submit** button. A page returns with the activation key required in the next step.
6. Perform Step 3 below with your activation key.

2.7.2.3 Step 3: Activating the Licensed Add-on

1. Select **Licensed Add-ons**. The screen appears displaying available features and activation functions.
2. Click the **Enter Key** button. The **Key Entry** window appears.
3. Enter the activation key in the *Activation Key* text box then click the **OK** button. A window appears confirming the feature was enabled.

CHAPTER 3

USING TALKSWITCH

3.1 IN THE OFFICE — RECEIVING CALLS WITH OR WITHOUT THE AUTO ATTENDANT

3.1.1 Receiving calls using the auto attendant

If you don't have a receptionist, the auto attendant helps you handle all your inbound calls. Once an auto attendant has been configured and recorded, it will follow the instructions given in the configuration. For details on adding and configuring auto attendants, see *2.4.2 Auto attendant* on page 83 and its subsections.

When calls are answered by an auto attendant, callers can:

- Dial any local or remote extension
- Dial any VoIP number (250–299)
- Dial any ring group (300–309)
- Dial voicemail directly
- Use a dial-by-name directory
- Choose options 0, 1, 2, 3, 4, 5 (if configured to perform specific actions)
- Transmit a fax
- Go to a different auto attendant
- Remain on the line to follow an action programmed for that auto attendant

TalkSwitch owners and employees can access other options at the auto attendant:

- Call back or call bridge
- Retrieve voicemail
- Enter command mode to make system changes

All these features can be password-protected to avoid unauthorized access.

TalkSwitch also provides the option of ringing specific extensions or ring groups prior to the engagement of an auto attendant. For details about configuring auto attendants, see *2.4.2 Auto attendant* on page 83.

3.1.2 Receiving calls without the auto attendant

If you do not use the auto attendant to answer calls, please note that the call cascade options for the local extensions are not engaged. For calls to go to an extension's mailbox, the call must be transferred manually by someone in the office. Once calls have been answered at a local extension, they can be transferred to any extension or voice mailbox.

3.2 IN THE OFFICE — MAKING AND RECEIVING CALLS USING AN ANALOG PHONE

If you are using an IP phone, see *3.3 In the Office — Making and receiving calls using an IP phone* on page 134 and following pages.

3.2.1 Making calls from a local extension without direct line access

Place intercom calls from one local extension to another local extension, remote extension or extension ring group

You can place intercom calls from one local extension to another local, remote or VoIP extension or to a ring group.

1. Pick up your extension's handset.
2. Dial the number of the extension you want to intercom to. An intercom call is identified by two short rings.

Exceptions: For a remote extension intercom call, the phone rings as usual. If a ring group has a different ring pattern, intercom calls follow the cascade settings of the extension dialed. You can enable this feature in the configuration software under **Options -> Miscellaneous**.

You can perform intercom paging and activate the speakerphone at other local extensions that have TS-80, TS-200, TS-400 or TS-600 phones. To place an intercom call to one of these phones, dial *** 84** and the local extension at the TalkSwitch dial tone.

Place out-of-office calls from a local extension

1. Press **9** or **81-88** to choose a hunt group at the internal dial tone. This hunt group gives you access to an available line to dial out.
2. If your system requires permissions for outgoing calls, enter your access code.
3. Dial the phone number.

Restrictions

You can set restrictions to specific hunt groups for each extension. Your extension can also be configured to access a line directly. When you pick up a handset, you hear the telephone company dial tone and can dial an outside number as usual.

If you have **Direct Line Access** enabled for a local extension, you need to dial **Flash** before dialing local extensions or voice mailboxes.

If you dial your own extension you will be directed to your voicemail. If your voicemail is not set up, you will get a busy signal.

3.2.2 Receiving calls at a local extension

The steps below apply when a regular extension is ringing. If an auto attendant engages before you are able to pick up, the caller can select an option at the auto attendant.

1. Pick up the extension's handset.
2. If the call went through the auto attendant and call screening is enabled in **Call Handling -> Local extension** and **Extension Ring Groups**:
 - a) Press **#** to accept the call.
 - b) Press ***** to re-route the call to its extension's call cascade options.
 - c) Hang up.

You can intercept a call that is ringing at any other local extension.

1. Pick up your handset.
2. At the TalkSwitch internal dial tone, dial *** 9**. This routes the call to your local extension.

You can answer a call ringing at a specific extension.

1. If you know the extension that is ringing, dial *** 7** and the extension number (i.e. *** 7112** to pick up a call ringing at extension 112).
2. If you are in the middle of a call and want to connect to another incoming caller, press **Flash** to place the first caller on hold.
3. Dial *** 9** or *** 7** and the extension number to connect to the second caller.
4. Once you are finished with the second call or would like to return to the first caller, press **Flash 5**.

3.2.3 Placing calls on hold at a local extension

Place calls on hold

1. Press **Flash** (on the TalkSwitch TS-80, TS-200, TS-400 or TS-600 phones).
2. Press **Flash** again to take callers off immediate hold.
If you have multiple callers on hold or in queue at your extension, press **Flash 7** to retrieve callers on a first in/first out basis.

If you press **Hold** on a regular, analog phone, you won't be able to access any of the TalkSwitch features while the caller is on hold. If you have music on hold enabled, the caller does not hear music while on hold.

3.2.4 Transferring calls from an extension

3.2.4.1 Unscreened transfer

You can initiate an unscreened transfer from a local extension to another local extension, remote extension or extension ring group.

1. Press **Flash** to put the caller on hold.
2. Dial the extension you want to transfer to and hang up. The caller is taken off hold when the receiving extension answers. If the transferred call is busy or not answered after the number of rings specified in the *No Answer* setting, or if the extension is busy, one of the following will happen:

The option can be altered under **Options -> Transfer Options** in the configuration software.

Default: The transferred call is handled according to the destination extension's *No Answer* or *Busy* settings.

Option: The call is transferred back to the transferring extension.

3.2.4.2 Screened transfer

You can initiate a screened transfer from a local extension to another local extension, remote extension or extension ring group.

1. Press **Flash** to put the caller on hold and dial the appropriate extension number.
 - a) If the person at the receiving extension wishes to take the call, hang up or dial **Flash 4** to complete the transfer. The caller is immediately taken off hold and connected to the receiving extension. You hear an internal dial tone.
 - b) If the person at the receiving extension does not wish to take the call, press **Flash 5** to return to the caller.
 - c) If you dial an extension that is busy or not answered, you can complete the transfer so that the call is routed according to that extension's call cascade settings.

If you handle many calls, you may want to complete or cancel a transfer without hanging up. **Flash 4** will complete the transfer and provide an internal dial tone. **Flash 5** cancels the transfer and connects you back to the caller.

3.2.5 Transferring calls from a local extension to an external phone

Unscreened Transfer

1. Press **Flash** to put the caller on hold.
2. Select a hunt group (dial **80** for *Same Line Connect*).
3. Dial the outside telephone number.
4. Press **Flash 4** to complete the transfer. Do not hang up to complete the transfer because this disconnects the call.

Screened Transfer

Instead of completing the transfer once you have dialed the outside number, wait until the call is answered and ask the recipient at the extension if he or she wishes to take the call. If the recipient accepts the call, press **Flash 4** to complete the transfer, if not, press **Flash 5** to return to the caller.

If you have the 3-way calling/conference or a Centrex transfer feature enabled on the current line, select the *Same Line Connect (80)* to make use of these features. Everything proceeds as above on the same line the call came in.

3.2.6 Parking and retrieving calls at a local extension

Call park is a feature for placing a call on hold and then retrieving it from another local extension. TalkSwitch has 10 park orbits (500–509).

3.2.6.1 Parking a call

1. Press **Flash** and *** 510**. TalkSwitch selects the first available park orbit (500–509). If you use a TS-80, TS-200, TS-400 or TS-600 phone, press the **Park** button or softkey.
2. You hear a confirmation indicating that the caller has been parked successfully and into which park orbit.

3.2.6.2 Retrieving a parked call at another local extension

The person who parked the call notifies the person at the extension for whom the call is parked (e.g. *“Mary, there is a call parked for you in 500”*).

1. Press *** *** and the appropriate park orbit. If you use TS-80, TS-200, TS-400 or TS-600 phone, press the **Un-park** button or softkey and enter the park orbit. If *Direct Line Access* is enabled at the extension, you must press **Flash** first.

If **Direct Line Access** is enabled at your extension, press **Flash** before retrieving the parked call.

3.2.6.3 Using call park with the paging option

If the external paging option is enabled on TalkSwitch, you may choose to announce a parked call over the PA system by pressing *** 0**. (e.g. *“Mary,*

please pick up 501.”). This tells Mary that she can retrieve the parked call from any extension by pressing *** * 501** at the dial tone.

If **Direct Line Access** is enabled at your extension, press **Flash** before retrieving the parked call.

If you have any combination of TS-80, TS-200, TS-400 and TS-600 phones in the office, you can page an individual extension or a ring group (i.e. press *** 85**, the **Page** button or softkey and the extension or ring group number).

3.2.7 Queuing and retrieving callers

The following sections describe Uniform Call Distribution (UCD). This includes features like call queuing, call waiting, conference calling, etc.

Call queue is a useful feature for placing multiple callers on hold at your extension while you are on an existing call. For details on configuring call queue, please refer to *Call Handling* in section 2.4.5 *Call Cascade* on page 93.

3.2.7.1 Queuing calls to a single extension

If an extension is busy and has *queue call* set for the *Busy* call cascade, incoming callers hear one of the prompts below, followed by music (if enabled) while they are on hold:

If the call came from the auto attendant and the extension has a voice mailbox:

“The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To leave a voicemail message, press 1, and to return to the main menu, press star.”

If the call came from the auto attendant and the extension has no voice mailbox:

“The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To return to the main menu, press star.”

If the call was transferred from another extension and the extension has a voice mailbox, but there is no auto attendant on this line with a main menu to return to:

“The extension you have reached is currently busy. If you wish to continue holding, please remain on the line. To leave a voicemail message, press one.”

If the call was transferred from another extension and the extension has no voice mailbox and there is no auto attendant on this line with a main menu to return to:

“The extension you have reached is currently busy. If you wish to continue holding, please remain on the line.”

While the caller is on hold at your extension, you hear a call waiting beep every 75 seconds. If you want to rotate through queued callers, dial **Flash 7**. The caller you are speaking to is placed on hold, and the first caller in the queue is retrieved. If you have multiple callers queued, press **Flash 7** to retrieve callers on a first in/first out basis.

If call waiting is enabled for the *Busy* action of a local extension, callers hear ringing (or music if enabled) while the local extension hears a call waiting beep and receives the caller ID for the new call. To connect to the new caller, press **Flash 7**. To let the caller fall through to the next level of the cascade, ignore the beep.

3.2.7.2 Queuing callers to an extension ring group

If you want to queue callers to a ring group, configure this action using an auto attendant.

Example: Configure auto attendant 1 to queue callers to ring group 301 for technical support if they have pressed **5**. When a caller presses **5** for technical support, they enter the queue immediately. Every 60 seconds the caller hears the following prompt: *“Your call is important to us. Please remain on the line. To return to the previous menu, press ***#**.”* Extensions that are part of the ring group and are available will start ringing within 10 seconds of hanging up the phone from a previous call. The delay allows time to make a new call or activate *Do not Disturb* as a log out option. After 5 extension rings, it will ring once every 16 seconds. When you pick up, you hear a prompt that provides the option to accept the queued call by pressing **#** or leave the caller in the queue by pressing ***#** and returning to the dial tone to make a new call.

3.2.8 Using call waiting

The call waiting feature is activated under **Call Handling -> Local Extensions** in call cascade section of the *Busy* tab list (*invoke Call Waiting*). When the dialed local extension is busy, the caller hears ringing or music depending on the settings.

If the phone supports call waiting caller ID (type II signaling), you hear a beep followed by a short blip. Depending on your phone, you may hear

something similar to a modem noise. This is normal. During this notification, the caller ID information is displayed on the phone. The caller does not hear the beep. The phone mutes the other side.

If the phone does not support call waiting and caller ID, you hear a beep. When the call waiting notification beep is heard, the person at the extension can dial **Flash 7** to put the current caller on hold and answer the new caller. He or she has the option to dial **Flash 5**. This terminates the current call and connects to the new call. If the person at the extension does not want to take the new call, they can ignore the call waiting beeps and TalkSwitch directs the caller to the next level of the call cascade for that extension (i.e. send the caller to the associated voice mailbox).

3.2.9 Conference calling

You can set up conference calling between 3 local TalkSwitch extensions, two local extensions and one outside caller or two outside callers and one local extension.

3.2.9.1 Two local TalkSwitch extensions and one outside caller

You do not require access to the telephone company's 3-way calling/conference service to use the TalkSwitch conference calling capabilities.

1. Establish a call with an outside party.
2. Press **Flash** to place the outside caller on hold. Dial the number of the local extension you wish to conference with.
3. When the extension is picked up, press **Flash 6** to establish the 3-way call.

3.2.9.2 Two outside callers and one local extension

There are two different methods for this type of conference call. The first method is similar to the one above, using only TalkSwitch for the conference function.

1. Establish a call with an outside party.
2. Place the outside caller on hold and dial the number of the other outside party. Dial **9** or another hunt group before dialing the second phone number unless you have **Direct Line Access** enabled.
3. When the call is answered, press **Flash 6** to establish the 3-way call.

The second method uses the telephone company's 3-way calling/conference service.

1. Establish a call with an outside party.
2. Press **Flash 80** to place the outside caller on hold and dial the number of the other outside party.
3. When the call is answered, press **Flash 6** to establish the 3-way call.

The conference initiator can disengage the second conferenced party at any time by pressing **Flash 5**.

3.3 IN THE OFFICE — MAKING AND RECEIVING CALLS USING AN IP PHONE



3.3.1 Making calls from a local IP extension (IP phone)

Place intercom calls from one local extension to another local or remote extension or to a ring group

An intercom call is usually identified by 2 short rings. The IP phones that we recommend support this distinctive ring feature. You can place intercom calls from one local extension to another local, remote, VoIP location or ring group.

1. Pick up your extension's handset and dial the number of the extension you want to intercom.

Exceptions: For a remote extension intercom call, the phone rings as usual. If a ring group has a different ring pattern, intercom calls follow the cascade settings of the extension dialed. You can enable this feature in the configuration software under **Options -> Miscellaneous**.

You can perform intercom paging and activate the speakerphone at other local extensions that have TS-80, TS-200, TS-400 or TS-600 phones. To place an intercom call to one of these phones, dial *** 84** and the local extension, then press **Dial, Send** or **#** to place the call.

Place out-of-office calls from a local extension

1. Dial **9** or **81-88** to choose a hunt group and dial the phone number.
2. Depending on your IP phone, press **Dial, Send** or **#**.

You can set restrictions to specific hunt groups or block certain types of outbound calls. Your extension can also be configured to access a specific line directly.

If **Direct Line Access** is enabled for a local IP extension, you need to press *** *** before dialing any extensions or voice mailbox.



3.3.2 Receiving calls at a local IP extension

If an auto attendant engages before you are able to pick up, the caller can select an option at the auto attendant. The steps below apply when you pick up an extension that is ringing.

1. Pick up the extension's handset.
2. If the call went through the auto attendant and call screening is enabled in **Call Handling -> Local Extension** and **Extension Ring Groups**:
 - a) Press **#** to accept the call.
 - b) Press *** *** or hang up to re-route the call to its extension call cascade options.
 - c) press **Dial, Send** or **#**.

You can intercept a call that is ringing on any other local extension.

1. Pick up your handset.
2. Dial *** * 9** and press **Dial, Send** or **#** depending on your phone. This routes the call to your local extension.

You can answer a call that is ringing at a specific extension.

1. If you know which extension is ringing, dial *** * 7** and the extension (i.e. *** * 7112** to pick up a call ringing at extension 112).
2. Press **Dial, Send** or **#** depending on your phone.
3. If you are in the middle of a call and want to connect to another incoming caller, press **Hold** to place the first caller on hold.
4. Dial *** * 9** or *** * 7** and the extension.
5. To connect to the second caller, press **Dial, Send** or **#** depending on your phone.
6. Once you are finished with the second call or would like to return to the first caller, press **Cancel** (softkey).



3.3.3 Hold and transfer

3.3.3.1 Hold

1. Press **Hold**.
2. Press the flashing line key that corresponds to the caller on hold to take the caller off hold. If you have multiple callers on hold or in queue at your extension, you can retrieve them on a first in/first out basis.
3. Press **Hold** and **7**, then press **Dial, Send** or **#** depending on your phone.

3.3.3.2 Unscreened transfer

1. Press the **Transfer** or **xfer** key.
2. Dial the appropriate extension number and then press the **Transfer** or **xfer** key. Some IP phones may support *hang-up transfer*. Please check the documentation of the IP phone to see if it supports this feature.

The caller is taken off hold when the receiving extension answers. If the transferred call is busy or not answered after the number of rings specified in the *No Answer* setting, or if the extension is busy, one of the following applies:

The option can be altered under **Options -> Transfer Options** in the configuration software.

Default: The transferred call is handled according to the destination extension's *No Answer* or *Busy* settings.

Option: The call is transferred back to the transferring extension.

3.3.3.3 Screened transfer

To initiate a screened transfer from an IP extension to another local extension, remote extension or extension ring group:

1. Press **Transfer** or **xfer** to put the caller on hold.
2. Dial the appropriate extension number.
 - a) If the person at the receiving extension wishes to take the call, press **Transfer** or **xfer** to complete the transfer. Some IP phones may

support *hang-up transfer*. Please check the documentation of the IP phone to see if it supports this feature. The caller is immediately taken off hold and connected to the receiving extension.

- b) If the person at the receiving extension does not wish to take the call, press **Cancel**. If this softkey is unavailable, press the flashing line button that corresponds to the caller on hold to return to the caller.
- c) If you dial an extension that is busy or not answered, you can complete the transfer and the caller is directed according to that extension's call cascade settings.

3.3.3.4 Transfer from a IP extension to any outside number

Unscreened Transfer

1. Press **Transfer** or **xfer** to put the caller on hold.
2. Dial a hunt group (or **80** for *Same Line Connect*) and the outside telephone number. If **Direct Line Access** is enabled, dial the phone number without a prefix.
3. Press **Transfer** or **xfer** to complete the transfer. Do not hang up to complete the transfer because this disconnects the call.

Screened Transfer

If the person at the receiving extension does not wish to take the call:

4. Press **Cancel** or the flashing line button that corresponds to the call on hold to return to the caller.

If you have the 3-way calling/conference or the Centrex transfer feature enabled on the current line, select the *Same Line Connect* (80) to make use of these features. Everything proceeds as above on the same line the call came in.

3.3.4 Call park — Parking and retrieving callers

Call park is a feature for placing a call on hold and then retrieving it from any other local extension. TalkSwitch has 10 park orbits, 500–509.

3.3.4.1 Parking a caller

1. Press **Transfer** or **xfer**, *** 510** and press **Dial** or **#** to place a caller in the next available park orbit.

2. You will hear a confirmation indicating that the caller has been parked successfully and into which park orbit.

3.3.4.2 Retrieving a parked call

1. Press *** *** and the park orbit.
2. Press **Dial** or **#**. If **Direct Line Access** is enabled at the extension, dial *** *** then *** *** and the park orbit, then press **Dial, Send** or **#**.

3.3.4.3 Using call park with the paging option

If the external paging option is enabled on TalkSwitch, you can announce the call over the PA system by pressing *** 0**. If **Direct Line Access** is enabled, dial *** * * 0** and announce that there is a call parked in a specific park orbit for a co-worker.

Example: “Bob, please pick up 501.” This tells Bob that he can retrieve the parked call from any extension by pressing *** * 501** at the dial tone. The caller is now connected to Bob at that extension. If **Direct Line Access** is enabled at your extension, press **Flash** before retrieving the parked call.

With an IP phone, dial *** * * * 501**.

If you have any combination of TS-80, TS-200, TS-400 and TS-600 phones in the office, you can page an individual extension or a ring group (i.e. press *** 85**, the **Page** button or softkey and the extension or ring group number).

3.3.5 Queuing and retrieving callers

For details on queuing and retrieving callers from a IP extension, please refer to sections 3.2.7, 3.2.7.1 and 3.2.7.2. This feature is used the same way on analog and IP phones except for the slight differences outlined in the next paragraph.

Some IP phones display a notification on the screen while the caller is on hold at your extension. If you want to rotate through queued callers while you are on the phone, dial **7** and press **Dial, Send** or **#**. The caller you are speaking to is placed on hold, and the first caller in the queue is retrieved. If several callers are queued, dial **7** and press **Dial, Send** or **#** to retrieve them on a first in/first out basis.

3.3.6 Using the TalkSwitch call waiting feature

If the call waiting feature is activated as a *Busy* option in the call cascade routing for a local extension and the dialed extension is in use, the caller hears ringing or music depending on the music on hold settings. The person at the extension hears the following:

- If the phone supports call waiting caller ID, you hear a beep followed by a short blip.
- If the phone does not support call waiting caller ID, you hear a beep. When the call waiting notification beep is heard, the person at the extension can pick up another line and dial **7** and press **Dial, Send** or **#** to put the current caller on hold and answer the new caller. To return to the previous call, press the flashing line key that corresponds to the previous call. On most IP phones, this will put the current call on hold and connect to the other caller.
- To terminate the current call and go back to the other caller on hold, hang up, then go off-hook and press the button that corresponds to the flashing line of the caller on hold. If the person at the extension does not want to take the new call, she or he can ignore the call waiting beeps and TalkSwitch directs the caller to the next level of the call cascade for that extension (i.e. send the caller to the associated voice mailbox).



3.3.7 Conference calling with TalkSwitch

3.3.7.1 Two TalkSwitch local extensions and one outside caller

You do not require access to the telephone company's 3-way calling/conference service to use the TalkSwitch conference calling capabilities.

1. Establish a call with an outside party.
2. Press **Conf** or **Hold** to place the outside caller on hold. Dial the local extension you wish to conference with.
3. When the extension answers, press **Conf** to establish the 3-way call.

3.3.7.2 Two outside callers and one local extension

There are two different methods for this type of conference call. The first method is similar to the one above, using only TalkSwitch for the conference function.

1. Establish a call with an outside party.

2. Press **Conf** or **Hold** to place the outside caller on hold.
3. Pick up another line and dial the number of the other outside party.
Dial **9** or another hunt group before dialing the second phone number.
4. When the call is answered, press **Conf** to establish the 3-way call.

The second method uses the telephone company's 3-way calling/conference service.

1. Establish a call with an outside party.
2. Press **Hold** to place the outside caller on hold.
3. At the dial tone, enter **80**, dial the number of the other outside party and press **Dial, Send** or **#**.
4. When the call is answered, press **Transfer** to establish the 3-way call.



3.4 MAKING AND RECEIVING CALLS USING VOIP

TalkSwitch supports inter-branch connectivity via PSTN and VoIP. Once the system is properly configured, calls made between locations are using VoIP with broadband connections. TalkSwitch has 50 VoIP location codes (250-299) that can be assigned to any VoIP number. These location codes are dialed from an extension or from an auto attendant. TalkSwitch automatically finds the VoIP ports for placing these calls.

Other calls are made over VoIP by dialing hunt group **88**. Activate VoIP numbers in this hunt group first. Dialing **88** at the internal dial tone tells TalkSwitch to hunt all VoIP resources for an available VoIP line. In order to reach a remote location, a VoIP phone number has to be assigned to the VoIP number you want to call.

Example: You have 3 branch locations (New York, Miami, San Francisco).

Location 1: New York

Phone numbers assigned to 4 VoIP lines: 250-261 (12 phone numbers)

Location 2: Miami

Phone numbers assigned to 4 VoIP lines: 270-273 (4 phone numbers)

Location 3: San Francisco

Phone numbers assigned to 4 VoIP lines: 280-283 (4 phone numbers)

If someone in the New York office wants to reach someone in the Miami office, they can dial one of the phone numbers associated to Miami (270-273). TalkSwitch automatically finds an available VoIP number and

connects to the Miami location. The TalkSwitch at the Miami location handles the call according to the configuration in the *Call Handling* section for the VoIP numbers.

3.5 USING PHONES CONNECTED IN PARALLEL TO TALKSWITCH

Phones connected in parallel (not connected to TalkSwitch but connected to the same line) with TalkSwitch always ring at least once before TalkSwitch starts ringing its extensions or the auto attendant answers the call. TalkSwitch uses the first ring to identify the incoming ring pattern before passing it on to the extensions or the auto attendant (TalkSwitch determines if the call is a distinctive ring and detects any caller ID information). A parallel phone can interrupt a telephone call on any of the incoming telephone lines.

Making and receiving calls on phones connected in parallel to TalkSwitch

Make calls on parallel-connected phones as if TalkSwitch were not sharing the telephone line with them.

If you answer on a parallel-connected phone before the auto attendant engages, TalkSwitch takes no further action. If a call is answered on a parallel-connected phone after the auto attendant engages, you can turn off the auto attendant message by pressing *** * 0** at one of the TalkSwitch extensions, or let it time-out on its own, in which case you will hear: "Good-bye".

Transferring calls on parallel-connected phones

Calls answered on a parallel-connected phone can be transferred to any TalkSwitch extension or voice mailbox. To transfer the caller, press *** ***. After you hear 2 beeps, dial the extension and hang up. You cannot announce a caller to the transferred extension but if you stay on the line, you are in a conference call state. You can hang up at any time.

If you transfer a call from a parallel-connected phone and then hang up, the extension rings the number of times specified for that extension. If it is not answered within that number of rings, the call is handled according to that extension's call cascade actions.

When a phone in parallel with TalkSwitch is off-hook, you notice that the corresponding line light flickers to indicate the line is busy. TalkSwitch does not allow extensions to interfere on a call that has the line in use.

You can change the transfer prefix to ***#** instead of ****** if the default prefix conflicts with another device or service. To change the transfer prefix, open the configuration software and change the setting in the **Options -> Transfer Options** window.

3.6 MODEMS AND TELEPHONE LINE ACCESS

If your modem is attached to TalkSwitch as an extension, you can configure it to access your incoming telephone lines in one of two ways.

1. Enable **Direct Line Access** to one of your incoming telephone lines. You don't have to dial **9** or any other hunt group to access an outside line at the extension where your modem is connected. Your modem will work the same way it always has. You don't have to change your dial-up settings for the modem. See *2.2.5 Automatic Route Selection and Toll Restriction* on page 45 for more details on configuring **Direct Line Access**.
2. If you use a telephone and modem on the same extension, do not enable **Direct Line Access**. A TalkSwitch extension with **Direct Line Access** immediately connects to one of your incoming telephone lines when you pick up the telephone's handset, preventing you from accessing other system features without first pressing **Flash** on an analog phone or ****** on a IP phone.

You can have your modem access a specific incoming telephone line using a hunt group.

Change your modem's dial properties to include the intended hunt group (**9** or **81-88**) followed by the letter **w**, before the dial-up networking telephone number (e.g. **9w789-1234**). The letter **w** after your dial-out access number tells your modem to dial the access number 9, wait until it detects a dial tone, and then dial your dial-up networking number.

If your modem has problems dialing out, use a comma (,) to create a 2 second pause instead of the **w** (wait for dial tone).

Because TalkSwitch units are digital systems, modem calls will only operate at a maximum speed of approximately 28 Kbps.



3.7 OUT OF THE OFFICE — RECEIVING CALLS WITH CALL FORWARDING (ANALOG AND IP)

3.7.1 Three ways to forward calls

1. Automatic call forwarding

The auto attendant can forward office calls to your call forwarding telephone number (i.e. remote extensions). Use the auto attendant message to prompt callers to select your remote extension number(s). (e.g. *"To speak to John Smith, press 211."*)

2. Manual call forwarding

If someone in your office has answered a call at a local extension, the call can be transferred to one of the call forwarding (remote extension) numbers.

- a) Put the call on hold at the local extension.
- b) Dial the remote extension number (e.g. 211) to which you want the call forwarded.
- c) Hang up an analog phone or press **Transfer** or **xfer** on a IP phone.

The call is handled according to the call cascade options of that remote extension.

3. Conditional call forwarding

If the auto attendant answers a call and the caller selects an extension or ring group number that is busy or is not answered, the auto attendant can forward the call to a remote (out of office) extension number. Conditional call forwarding is configured in the TalkSwitch software using the call cascade's *No answer* or *Busy* settings for each local extension, remote extension and ring group (see 2.2.7 *Local extensions* on page 56 through to 2.2.9 *Extension ring groups* on page 64).

3.7.2 Transferring calls from a remote extension

You can forward calls from a remote extension without using Centrex or 3-way calling services. Calls can be transferred to any local or remote extension, ring group or voice mailbox. This feature also applies to VoIP numbers.

1. Press *** * *** to place a call on hold at a remote extension anytime during a conversation. You hear the same dial tone as at a local extension and have the following three possibilities:

- a) Press *** *** to retrieve the call placed on hold.
- b) Dial any local extension, remote extension, voice mailbox, VoIP location or ring group.

If the system is configured in the TalkSwitch configuration software under **Options -> Transfer Options** in the *Remote Transfer* section to *Perform a blind transfer*, the system plays the prompt “Call Transferred — Goodbye.” and hangs up after the extension or mailbox is dialed.

If the system is configured in the *Transfer Options* section to allow call screening, dial *** * 4** to complete the transfer or *** * 5** to cancel the transfer and return to the caller.

3.7.3 Screening options for forwarded calls

The screening options for call forwarding are configured in the TalkSwitch software using the call cascade *Answered* options in the remote extension window.

Forwarding calls with screening

When *play accept/reject prompt* is configured in the *Answered* call cascade option under **Call Handling -> Remote Extension**, TalkSwitch plays a pre-recorded prompt to callers asking them to hold and then dials your call forwarding number. When you answer the forwarded call, you hear a prompt saying: “You have a forwarded call. To accept the call, press **#**. To reject the call, press *** ***.” If you reject the call, it is routed according to the call cascade option specified next to *If a call is rejected from this ext.:*

When *play caller's name first* is configured in the *Answered* call cascade option, TalkSwitch plays a pre-recorded prompt to callers asking them to record their name at the sound of the tone. TalkSwitch asks the caller to hold and dials the call forwarding telephone number. When you answer the forwarded call, TalkSwitch plays a pre-recorded prompt saying: “This is call forward, you have a call from...” and plays the recording of the caller's name. To accept the forwarded call, press **#** on your telephone keypad, re-route the call to its call screening call cascade options by pressing *** ***.

When either screening option is used, music on hold is enabled and you are not using the 3-way calling feature (*Same Line Connect*) to forward calls, the caller hears music while TalkSwitch is waiting for the remote extension to accept the call.

If a forwarded call is not answered or the line is busy, TalkSwitch performs the following, based on how the call was originally forwarded:

- If the call was automatically forwarded, it follows the remote extension's *No Answer* or *Busy* call cascade action.
- If the call was manually forwarded, it follows the remote extension's *No Answer* or *Busy* call cascade action. This action can be changed to ring the extension back to the person who performed the transfer.
- If the call was conditionally forwarded, it follows the call cascade action in the *No Answer* or *Busy* for the original extension selected.



3.7.4 Calls over VoIP with IP phones and Gateways

If teleworkers are connected to the system via a certified compatible IP device (i.e. gateway or IP phone), there are several ways for them to interact with TalkSwitch.



Making calls

Once the phone or gateway is registered with the TalkSwitch unit acting as the SIP server, calls can be made as follows:

1. At dial tone, dial a phone number associated to any of the other locations.
2. At dial tone, dial a phone number associated to one of the VoIP numbers on TalkSwitch. Ensure that an auto attendant is programmed to answer this line. Once the auto attendant has answered, dial an extension, VoIP location or voice mailbox. To place calls through the regular phone lines, you must activate and configure a call bridge account (see *2.5 Call Back/Call Bridge* on page 102). To access regular phone lines dial **9** and enter the password for call bridge.

Currently, TalkSwitch does not place calls on hold or transfer calls when you call other locations, since the call does not actually go through TalkSwitch.

Receiving calls

Once the phone or gateway is registered with the TalkSwitch unit acting as the SIP server, calls are received directly from other registered gateways or through the system.

1. If the call is routed via TalkSwitch, you can transfer the caller dialing *** *** and any extension or voice mailbox.
2. If the call came directly from a IP gateway or phone, you can transfer the call if your phone or service provider supports this feature.

3.8 USING THE VOICEMAIL SYSTEM

Each TalkSwitch unit comes with 30 minutes of voicemail recording capacity. This time is shared with auto attendant greetings. The capacity can be increased by adding TalkSwitch memory cards. You can add 1, 2, 4.5, or 9 hours of storage to each unit by using internal memory expansion activation keys, or TalkSwitch memory cards. When you insert the memory card, the system configures it automatically. Contact your TalkSwitch dealer to purchase memory cards.

3.8.1 Activating voice mailboxes

There are three types of voice mailboxes in the **Voicemail** folder of the configuration software.

Local Ext. Voicemail

The local mailboxes are associated with local extensions. All local extensions are activated with a respective mailbox by default. You can also activate the extension as an announcement or you can de-activate it.

Remote Ext. Voicemail

The remote mailboxes are associated with the remote extensions. If you activate a remote extension, it is activated as mailbox by default. You can also activate the extension as announcement or you can de-activate it.

General Voicemail

Activate these mailboxes in the *General Voicemail* window. They are not associated with particular extensions. You can also activate the extension as announcement or you can de-activate it.

If you want incoming calls to go straight to a voice mailbox after a specified number of rings, program an auto attendant to answer and provide a fall through to voicemail.

Callers can reach voice mailboxes, call cascade, be transferred from a local extension or have announcements played through the auto attendant.



To leave a message from a regular local extension:

1. Dial ***#** and the mailbox number.

To leave a message from a local IP extension:

1. Dial ***#** and the mailbox number and press **Dial, Send** or **#**.

To transfer a caller directly to voicemail from a regular phone:

1. Press **Flash** *****, the mailbox number and hang up.



To transfer a call to a voice mailbox from an IP phone:

1. Press **Transfer** or **xfer** *****.
2. Enter the mailbox number.
3. Press **Transfer** or **xfer** to complete.
4. Press **Dial, Send** or **#**.

When a caller presses **#** after leaving a message or after 2 minutes have elapsed, TalkSwitch plays a prompt asking callers if they wish to keep the message, listen to the message or re-record the message. Callers can press **0** and the system directs the call according to the settings of that mailbox.

3.8.2 Retrieving messages and accessing a voice mailbox

The first time you access your voice mailbox, you are prompted to assign a password, record a greeting and record your name for the dial-by-name directory.

For the general mailboxes, there are no directory listings and you are not prompted to record a name. If you wish to have your name in the company directory, use the mailbox associated with its extension as the directory is related to the extension's mailbox. When your mailbox has been set up, you access it through the prompt instructions.

When a new message is left in a local extension mailbox, TalkSwitch plays a stutter dial tone when the handset is picked up. If the phone supports a frequency-shift keying (FSK) message waiting indicator (not applicable to IP phones), the light flashes. The TS-80, TS-200, TS-400 and TS-600 phones support a message waiting counter. The display on the phone shows the number of new messages stored in the mailbox(es) associated with that extension.

1. Dial ***** ***** **#** from a local extension or ***** ***** and the mailbox number at the auto attendant if you are dialing from an outside location. From an IP phone, press **Dial, Send** or **#**.
2. Local extensions can transfer employees to access voicemail by pressing **Flash** ***** ***** and entering the mailbox number. From an IP phone, press **Transfer** or **xfer** + mailbox + **Dial, Send** or **#**.
If **Direct Line Access** is enabled, be sure to press **Flash** before trying to

access your mailbox. From a Grandstream or Polycom IP phone, press **MSG** or **Messages** button.

If you have a line configured to go straight to voicemail, dial **8** during the greeting to access messages from outside phones.

a) **Press 1 to listen to messages.**

By default, TalkSwitch starts with newest message before it plays the older ones. While the message is playing or after the message has finished, you can dial the following:

1	to rewind 10 seconds
11	to rewind to beginning of the message
3	to skip ahead 10 seconds
33	to skip to the end of the message
5	to listen to the time and date stamp
6	to forward the message
7	to delete a message
9	to save a message
* 	to back up one level
# 	to skip to next message, leaving new messages as new.

b) **Press 2 to change greeting options.**

This allows you to record a new personal voicemail greeting. The default greeting is: *"The extension you have reached is unavailable at this time. Please leave a message after the tone".*

Remember: While recording your greeting or announcement, callers can press ***** to return to the auto attendant, if the call came from the auto attendant. When they are done recording, they can press **#** for more options. You may want to give your callers these options for their convenience. In local extension mailbox options, **0** can be configured as an option for additional routing.

c) **Press 3 to change personal options.**

- change your current password
- turn the auto date and time stamp on/off
- turn the phone number on/off

- set up or change pager and remote phone notification

By default, the auto date and time stamp is on. This information is played at the end of each message. If you have caller ID service, the phone number is displayed with the time and date.

d) **Press 4 to record your name for the dial-by-name directory.**

If you haven't recorded your name for the directory or if you wish to change the recording, press **4**. The dial-by-name directory is accessed at the auto attendant according to your configuration. Callers are directed to enter the first 3 letters of the employee's last name. The system finds the exact or closest match and plays the recorded name with the extension number, giving callers the option to connect to that extension.

You must record your name in order for the extension to be included in the dial-by-name directory.



3.8.3 Recording an announcement on a regular or an IP phone

An announcement is a recorded message providing information to callers. It can be recorded using your phone as a voicemail message or loaded as an audio file. The system hangs up after playing an announcement. Callers have the option of pressing ***** during an announcement to return to the auto attendant. Callers can also press **0** to perform other actions if this is configured in the voicemail options.

You might include in the announcement a mention of the selections a caller may press and leave a section of silence at the end to give the caller time to make a selection.

To record an announcement, access a voice mailbox and select option **2** to change greeting options as described in *3.8.2 Retrieving messages and accessing a voice mailbox* on page 147. To load a pre-recorded announcement or to configure **0** as an option to be selected, see *2.3.1 Mailbox options* on page 70. To configure call handling options for announcements, see *2.4 Call Handling* on page 81.

3.8.4 Pager and cell phone notification

When a new message is left in a voice mailbox, it can notify a pager, cell phone or a remote number.

You can configure the system to notify you for every new message or for only the first new message received since the last time you accessed new messages. For more information, see *2.3.2 Voicemail notification* on page 72.

If you use remote phone notification, you can accept or reject the notification call when you answer.

- Press **#** to listen to messages.
- Press ***** to postpone listening to messages.

For details on configuring pager and remote phone notification, see *2.3.2.1 Dialed notification* on page 72.

3.9 MUSIC ON HOLD

TalkSwitch can play music to callers when they are on hold, parked or queued. The music played is provided by the source you have connected to the music jack or from a sound file (8 KHz, 8 bit, mono, u-law, .wav format) stored on the unit(s). When a sound file is loaded on one of the TalkSwitch units, it is duplicated to all units on the LAN.

Music on hold and call forwarding to remote extensions

If you are using the telephone company's 3-way calling/conference service (*Same Line Connect*) to activate the TalkSwitch call forwarding feature, callers do not hear your music while they are forwarded to a remote extension. When calls are forwarded, the caller is put on hold by the telephone company's central office switch rather than by TalkSwitch and the caller hears silence while being transferred.

A system prompt indicates that callers hear silence while the system is locating their party. This prompt explains to callers why there is silence for an extended period of time before connecting to the remote extension. This prompt cannot be disabled.



If you have two or more units on a LAN, we recommend that you load a .wav file to the system of units. If you use an external music source, supply a music source to each TalkSwitch unit. 1/8" audio splitters are available at most electronics stores. See *1.4 Unit back panel* on page 3 for more details.

3.10 MODE SWITCHING OPTIONS

Manual mode switching

Use telephone keypad commands at a local extension or at the auto attendant to switch modes.

1. Dial **#** at the local extension dial tone or at the auto attendant to enter command mode.
2. Enter a password if required.
 - a) Dial **30 #** to check the current mode.
 - b) Dial **31 #** to activate mode 1.
 - c) Dial **32 #** to activate mode 2.
 - d) Dial **33 #** to activate holiday mode.

Automatic mode switching using the time scheduler

Use the configuration software to activate the TalkSwitch time scheduler feature. You can configure TalkSwitch to switch between modes automatically (see *2.4.1 Modes* on page 81).

If you switch modes manually, TalkSwitch switches to the proper mode during the next scheduled mode change.

Ensure that the TalkSwitch configuration software tool is closed at the time of the scheduled mode switch. The mode cannot change if the configuration software is open.

3.11 OUT OF THE OFFICE — MAKING CALLS WITH CALL BACK AND CALL BRIDGE

Working together, call back and call bridge act as your personal long-distance operator. Whether you are across town or around the world, these two features allow you to place calls from your office incoming telephone line(s).

You can make a call to TalkSwitch, access an incoming telephone line connected to TalkSwitch and enter the number you want to dial. This is especially useful when you are out of the office with your cell phone and need to make a long-distance call. You can avoid the long-distance cell rates by making a local call to TalkSwitch and access your office savings plan through call bridge.

3.11.1 Using call bridge

1. Dial into an incoming telephone line.
2. When the auto attendant answers, select a hunt group (**9, 81-88** or *Same Line Connect 80*).
3. Enter your account password. If you are using *Same Line Connect* and your line supports the 3-way calling/conference service, you are prompted to enter the phone number.
4. When you have completed your call, do one of the following:
 - a) Make another call by pressing **# #**.
TalkSwitch disconnects you from the call in progress. You can dial another number or redial the same number.
 - b) Activate the auto attendant by pressing **# ***.
TalkSwitch ends the call bridge session and activates the auto attendant. When the auto attendant message begins to play, you have the following options:
 - Press **#** or *** 55** to enter command mode
 - Dial **6** to change/enter call back settings
 - Dial a local or remote extension or a ring group. This option allows you to contact someone in your office, check your voicemail, etc.
 - c) End the call bridge session by hanging up. This disconnects you from TalkSwitch. To ensure that TalkSwitch disconnects at the end of a call bridge session, press **# #** before hanging up.

The call bridge telephone number you dial can be local, long-distance, toll free (800 and 888) or international. Enter the telephone number as you would in your office. For long-distance calls, include the 1, country code and area code.

3.11.2 Using call back

The call back feature allows you to initiate TalkSwitch to call you at a specified telephone number. This gives you access to the following:

- Call bridge
- Local extensions, remote extensions and ring groups
- Voicemail

- Configuration settings

Using call back involves 3 steps:

1. Call your TalkSwitch telephone number to activate call back.
2. Answer and accept call back. TalkSwitch calls you back within 30 seconds.
3. At the auto attendant you can choose to perform call bridge, dial an extension, access and retrieve voicemail or configure TalkSwitch.

Activate call back

There are two ways to activate call back.

1. Using prompted call back activation from your out-of-office location:
 - a) Dial a TalkSwitch telephone number.
 - b) Let the line ring until the auto attendant answers.
 - c) Dial **6** on the telephone keypad.
 - d) If you have password protection on the account, enter your password.
 - e) TalkSwitch offers a series of prompts:
You can enter **1** and hang up to initiate the call back.
If you need to enter a new prompted call back number, enter **2** and follow the prompts.

For prompted call back, TalkSwitch always dials the last prompted call back number entered. If you do not have the auto attendant answer a line, a generic TalkSwitch auto attendant will answer after 15 rings. You can then dial **6** to access the call back settings.

2. Using auto call back activation from your out-of-office location:
 - a) Dial one of the call back telephone numbers that has been configured for auto call back activation.
 - b) Let the line ring at least once but no more than three times and hang up before the call is answered.
 - c) Within 20 seconds, TalkSwitch calls your auto call back number.

It is important to remember that all call back settings must be configured to use auto call back. Please ensure that the auto attendant for the line using call back does not answer before 4 rings.

Accepting the call back

There are two ways to accept a call back from TalkSwitch.

1. Answer the call and press **#**.
If required, enter your password. This option is used when you have a direct line to your call back telephone number.

The *Use Announced message* option allows the call back from TalkSwitch to reach you when you are in a hotel or an office where calls are intercepted by a receptionist or switchboard operator. When the call back is answered, TalkSwitch begins playing your pre-recorded message (e.g. *“Please forward this call to Bob Smith in room 312”*). The message is repeated for 2 minutes. When the call is forwarded to you, press **#** and enter your password on the telephone keypad to accept the call back. If the call back is not accepted within the initial 2-minute period, TalkSwitch disconnects the call back. When you accept the call, you hear the auto attendant. You can select to check your voicemail, ring an extension, use call bridge or enter command mode.

CHAPTER 4

CALL DETAIL RECORD (CDR) LOGGING

This chapter covers the setup and retrieval of records generated by calls through the phone system.

The *Store to File on TalkSwitch* option saves a record of calls on the unit. These records can be accessed via a web interface, as described in this chapter.

4.1 ENABLING CALL DETAIL RECORD (CDR) LOGGING

Units can report the CDRs in a file on the unit itself.

If multiple units are networked on a LAN, the CDR data is stored on unit 1.

1. Connect TalkSwitch to your computer and open the TalkSwitch configuration software.
2. Click **Tools -> Call Logging Output**.
3. Select **Store to File on TalkSwitch**.

Figure 109: Call log output options



4. File -> Save to TalkSwitch

Figure 110: Save settings



When storing the CDR to a file on TalkSwitch, approximately 6000 records are stored in a circular buffer. This means that after about 6000 records, the oldest ones will be deleted and new ones are created. On a very busy system, 6000 records may only cover a couple of days, but on a system with less traffic, 6000 records may cover more than a month.

4.2 RETRIEVING DATA

The following method is available for accessing CDR data on the TalkSwitch:

Web Interface — *Store to File on TalkSwitch*. While the TalkSwitch configuration software displays two options for **Call Logging Output**, only the available option can be selected.

4.2.1 Web interface — Store to File on TalkSwitch

You can access the CDR information through a web interface as follows:

1. Enter the unit IP address in the *Address* field of a web browser.

Address	<input type="text" value="http://192.168.1.200"/>
---------	---

Select **System Information -> IP Configuration** in the configuration software to find the unit's IP address.

2. Enter the system name in the *User Name* field and the administrator password in the *Password* field.

Figure 111: Log in window

User Name	<input type="text" value="TalkSwitch"/>
Password	<input type="password" value="*****"/>

Select **System Information -> Administration** in the TalkSwitch configuration software to find the username.

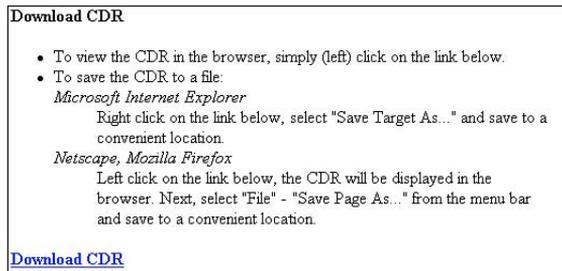
3. Click on the link **Call Detail Record (CDR) Management**.

Figure 112: Initial CDR web interface



4. You may either right click on **Download CDR** to save the information to your PC or left click on the link to view the current information. Browser specific information is available on the website when you scroll down.

Figure 113: Download CDR



Once the CDR file has been downloaded, you can clear the buffer/file by selecting the link under **Clear CDR**. The link will take you to a confirmation page. Select **Yes** to proceed or **No** to cancel.

Figure 114: CDR web interface



4.3 ANALYZING THE DATA

We recommend that you clear the text file immediately after it is downloaded. Restart saving data to a new file. This allows easier viewing and analyzing of the captured information.

The data can be viewed either through the web interface or downloaded into a spreadsheet (e.g. Excel) as a delimited text file. The log is divided into 13 columns identified by numbers:

1. Call Type	6. Event Duration	10. Line
2. Log Number	7. Connection	11. Account Number
3. Event Counter	8. Phone Number	12. Carrier Code
4. Date	9. Name from Caller ID	13. PIN Number
5. Time		

Each column contains specific information related to the current state of the call. The following is a list of the columns with a description:

1. **Type** — This column can contain the following characters:
 - a) **I** — Inbound call
 - b) **T** — Transfer state
 - c) **O** — Outbound call
 - d) **B** — Bridged call. For example, a call in on one line that is forwarded over another line. This can include calls forwarded to remote extensions.
 - e) **X** — Blocked call
 - f) **A** — Account number assigned to the call with the same log #.
2. **Log #** — This column indicates log number for the call. For the duration of the call, the log number remains the same. The event number will increment with each change of state. The first digit of the log number indicates on which unit the event originated.
3. **Event** — This column tracks various state changes of the call. For example, when an auto attendant answers, the counter might be 1. When the caller selects an extension, the counter increments to 2, etc.
4. **Date** — This column shows the date of the call (MM/DD/YYYY).

5. **Time** — This column contains the time that the call entered a specific state. The time is displayed in 24-hour format.
6. **Duration** — This column indicates the total time the call was in this state/event.
7. **Connection** — This column indicates where the call was for each event. This includes auto attendants, voicemail, extension ringing or queuing. *RCO* indicates that multiple extensions are ringing.

Exxx = Connected to extension xxx

(xxx = 111-118, 121- 128, 131-138, 141-148)

Axxx = Connected to auto attendant 00x (x= 1 to 9)

Rxxx = Ringing at extension xxx

(xxx = 111-118, 121-128, 131-138, 141-148)

Qxxx = Queued at extension xxx

(xxx = 111-118, 121-128, 131-138, 141-148, 300-309)

Mxxx = Connected to voice mailbox xxx

(xxx = 111-148, 211-248, 410-449)

COxx = Forwarded out on line xx. (See 10. *Line* below for more details).

8. **Phone Number** — This column displays the phone number of the inbound or the outbound caller if available.
9. **Name** — This column displays the name of the caller if available.
10. **Line** — This column indicates on which unit and line the call came in or out. The format of this information is unit and line number. Lines 1–4 are PSTN lines and 5–8 are VoIP lines.
Example: 11 — unit 1, line 1
23 — unit 2, line 3
16 — unit 1, VoIP 2
11. **Account Number** — This column contains an account number that the user can assign after a call. This allows you to group calls together that may have been placed over time to calculate the total amount of time spent with a particular customer. The account number can be a numerical code up to 15 digits long. To assign an account number after a call has been completed, dial *88 + <Account Number> + #.

If **Direct Line Access** is enabled on an extension:

- a) Analog phone — press **Flash**, then dial *88 + <Account Number> + #.
- b) IP phone — press *, *, then dial *88 + <Account Number> + #.

12. **Carrier Code/Access Code** — This column shows the phone number of the preferred long-distance provider.
13. **PIN or User Code** — This column shows the user's access code to dial out using a specific carrier.

The data can be imported into any call management software or an Excel spreadsheet as a delimited text file. After the import, the data can be grouped and tallied based on the criteria set forth.

Example:

1. Total line usage for a day
2. Number of inbound calls
3. Wait times
4. Average call times
5. Number of calls per line

CHAPTER 5

VOIP INFORMATION

5.1 INTRODUCTION TO VOIP

TalkSwitch units can be hybrid PSTN/VoIP phone systems. The addition of VoIP to the TalkSwitch product line adds the ability to use a broadband IP connection for voice calls.

VoIP also enables the use of IP phones as extensions on your TalkSwitch system. Connect an IP phone to your LAN and TalkSwitch can configure it to have the same powerful features as regular analog phones, plus exciting new features that break the boundaries of possibility for the office phone. See *1.9.2.3 Connecting an IP phone* on page 18 for information on connecting IP phones.

TalkSwitch units can be used in many different applications. The most popular VoIP configurations are multi-branch, teleworker and service provider integration.

TalkSwitch uses the industry-standard Session Initiation Protocol (SIP) to make and manage VoIP calls. The term SIP is often used in place of VoIP.

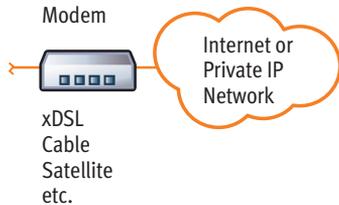
To connect to a VoIP network or VoIP service to make and receive SIP calls, each location must be able to establish a high-speed connection to the Internet or private network. The following steps should ensure success for typical network connection methods.



5.2 OPTIMIZING YOUR IP NETWORK FOR VOIP

5.2.1 The broadband connection

Figure 115: Broadband



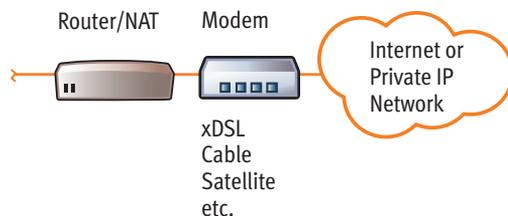
For optimal quality of VoIP calls on your system, we recommend that you subscribe to a high-speed broadband connection from your service provider. 'Lite' broadband is not suitable for simultaneous voice and data traffic. 'Lite' broadband typically runs 128 Kbps downstream and 64 Kbps upstream. The limiting factor for a broadband connection is the upstream bandwidth to the Internet Service Provider (ISP).

Example: If you have a 1 Mbps ADSL connection from your service provider, it likely means you have an upstream bandwidth of approximately 380 Kbps. Keep in mind that not the full 380 Kbps are available at all times. You can estimate about half of the upstream bandwidth as being available. In this case, you could safely support two simultaneous G.711 calls (100 Kbps/calls) or four simultaneous G.726 calls (50 Kbps/calls) or 10 simultaneous G.729 calls without factoring in other data traffic on the network at the same time. With additional traffic on the Internet or private IP network, the number of simultaneous calls supported is reduced.

G.711, G.726 and G.729 are VoIP codecs. For more information on codecs, see *Codec Options (button)* on page 173.

5.2.2 The router/NAT/firewall

Figure 116: Router

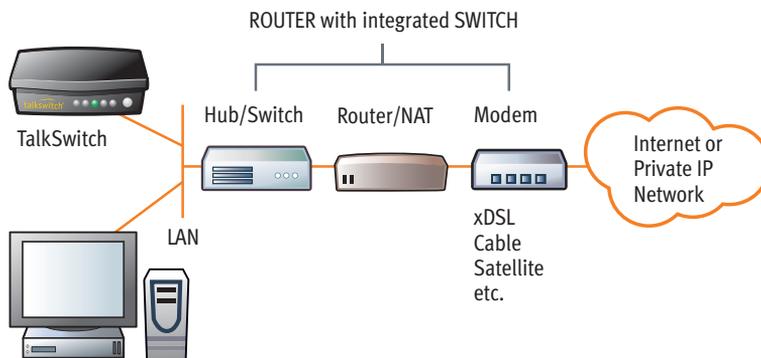


In a typical small office environment, there are several computers connected to an Ethernet switch, sometimes integrated into a router. A router handles the connection between two or more IP networks. Routers look at the destination IP addresses of the packets passing through and route them accordingly. In a network that shares data and voice, it is critical that the voice traffic has priority over the data packets. Some routers support Quality of Service (QoS) functionality for this purpose. These routers are identified as devices that support QoS, VoIP or prioritization.

5.2.3 Connecting to a LAN and IP network

Ensure the TalkSwitch or gateway is connected to a Local Area Network (LAN), equipped with an Ethernet switch and router or to a router with an integrated switch. A PC and the router/switch should be connected to the LAN for configuring the TalkSwitch unit.

Figure 117: LAN



To share voice and data properly, your network requires an Ethernet switch. A switch isolates the Ethernet ports from each other and only allows traffic intended for the device connected to that port. This ensures that data traffic doesn't compete with voice traffic. The most successful way to separate voice and data traffic is to install a router using different subnets for voice and data.

5.2.3.1 Confirm sufficient network capacity for VoIP

Ensure that the LAN in each location has a reliable high-speed broadband connection to the Internet. The quality of VoIP calls, especially on mixed voice and data networks, depends on high data-transfer rates across the

network. 'Lite' broadband connections are not suitable for simultaneous voice and data traffic.

To test the VoIP call capacity based on your Internet connection speed, perform the test available on our website at: http://www.talkswitch.com/voip/voip_test.asp.

5.2.3.2 Confirm router/firewall path for voice data

In order to pass through a router's firewall, voice data received by your network must be mapped to the appropriate TalkSwitch unit. TalkSwitch uses UPnP (Universal Plug and Play) to automatically map firewall ports to forward VoIP data. If your router supports UPnP, ensure that it is enabled (see your router documentation for configuration details).

If a router does not support UPnP, you will need to configure the router to map specific ports to TalkSwitch. The following describes which ports to map to TalkSwitch:

- To allow SIP Signaling data to be forwarded to TalkSwitch, firewall port 5060 must be mapped to the TalkSwitch unit known as the local proxy. By default, this is the lowest numbered TalkSwitch on a LAN. The IP configuration window of the TalkSwitch software indicates with an asterisk which TalkSwitch unit is the local proxy. SIP Signaling uses UDP. **Example:** in a 4-unit network, firewall port 5060 (UDP) would need to be mapped to the IP Address of the TalkSwitch unit ID #1.
- To allow voice traffic to be forwarded to TalkSwitch, corresponding ports must be mapped to each TalkSwitch unit. The following table shows the firewall ports to map to each TalkSwitch unit on a LAN.

Firewall ports* required open for VoIP communication with TalkSwitch

Description	Type	Unit 1	Unit 2	Unit 3	Unit 4
Voice traffic	UDP	6000–6006	6010–6016	6020–6026	6030–6036

*Default port numbers. For instructions on re-assigning these numbers, contact TalkSwitch Technical Support.

- If you plan on configuring TalkSwitch remotely, you will need to map port 9393 (Type: TCP) to the TalkSwitch unit that is being configured. Configuration is done using TCP.

TIP: For information on configuring a router for port forwarding, visit <http://www.portforward.com/routers.htm>.

Prioritize your voice traffic

If your Internet connection is shared between computers and TalkSwitch (i.e. data and voice), it is critical that the voice traffic has priority over the data traffic. Many routers support Quality of Service (QoS) routing for this purpose. If your router supports QoS, ensure that it is enabled (see your router documentation for configuration details).

DDNS Support

If you are using Dynamic Domain Name Service (DDNS) to manage the IP address of the SIP server, ensure that your router(s) support the specific DDNS service that you are using.

5.3 SETTING UP A VOIP NETWORK

A TalkSwitch VoIP network can be set up in 4 steps:

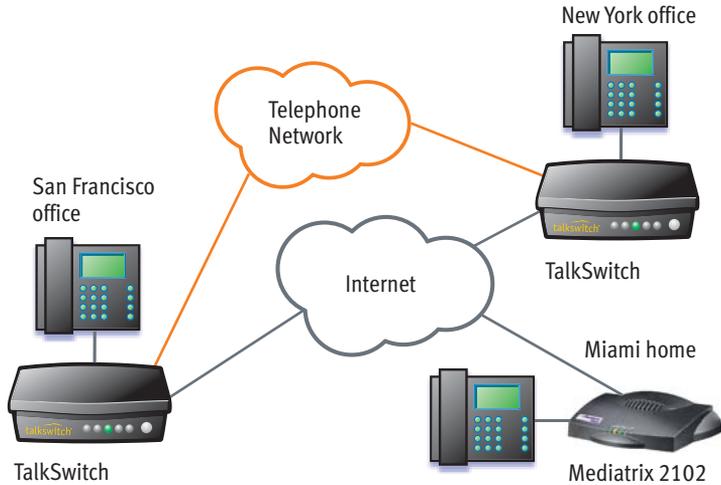
1. Connect a TalkSwitch or SIP-compatible gateway at each location.
2. Select a TalkSwitch to use as the SIP network server.
3. Assign phone numbers to each VoIP location.
4. Configure TalkSwitch to make and manage VoIP calls.

5.3.1 Connect TalkSwitch or SIP-compatible gateways

In order to create a VoIP network, a VoIP-equipped TalkSwitch or a VoIP compatible gateway, such as a Mediatrix® 2102, is required at each location. Each location in a multi-location network can support up to 4 TalkSwitch units, with a maximum of 32 local extensions (analog and/or IP phones) and 16 VoIP lines per location.

Example: *Figure 118* on page 166 shows a multi-location network where two locations are equipped with TalkSwitch systems and a teleworker is equipped with a Mediatrix 2102 device. TalkSwitch and the Mediatrix 2102 route VoIP calls over the Internet, while local calls are routed over the regular (traditional) telephone network.

Figure 118: Multi-location network with VoIP calling



5.3.2 Select which TalkSwitch to use as the SIP network server

A multi-location SIP network requires a SIP server to manage calls across the network. The SIP server contains registration information for all SIP devices in the network. When making or receiving a VoIP call, a device will register with the SIP server, which will then route the call to the appropriate destination.

TalkSwitch units have a built-in SIP server to manage a VoIP network. Though every networked TalkSwitch unit has the capability, you must designate only *one* TalkSwitch unit in a network as the SIP server.

For VoIP calls made through a VoIP service provider, the service provider will act as the SIP Server.

5.3.2.1 Which location and unit should be the SIP server?

Any TalkSwitch unit on the network can be designed to be the SIP server.

To simplify network operation and reliability, we recommend that you choose a location that has a *static*, rather than a *dynamic* public IP address. A static public IP address does not change, providing a consistent connection address for each location to communicate with the SIP server.

If you do not have a static IP address, you can obtain a Fully Qualified Domain Name (FQDN) from a Dynamic Domain Name Service (DDNS) provider. A DDNS matches your dynamic IP addresses to your FQDN, so that your TalkSwitch VoIP network operation is not affected by IP address changes.

One DDNS provider is DynDNS.

Visit <http://www.dyndns.com/services/dns/dyndns/> for more information.

Configuring TalkSwitch to act as the SIP server is described in 5.4.2.1 *TalkSwitch Profile* on page 170.

Example: Building on the previous example, the TalkSwitch system in the San Francisco office is designated as the SIP server. When a location initiates a call, the TalkSwitch in San Francisco will resolve the destination address for the call and coordinate a connection to route the call to its destination.

5.3.3 Assign phone numbers to each VoIP location

You can now assign phone numbers to locations in your VoIP network.

Each TalkSwitch unit with VoIP lines can support up to 12 VoIP phone numbers. Unique call handling scenarios can be configured for each VoIP number. VoIP phone numbers can be 3 or more digits in length.

TalkSwitch supports **50 VoIP location numbers** (250 to 299) that **permit direct** dialing from any extension or auto attendant in the network. If needed, numbers outside of this range can also be added. See 5.4.3 *VoIP Numbers* on page 176 for details. Each VoIP phone number assigned in a multi-location network must be unique (i.e. only used for one location).

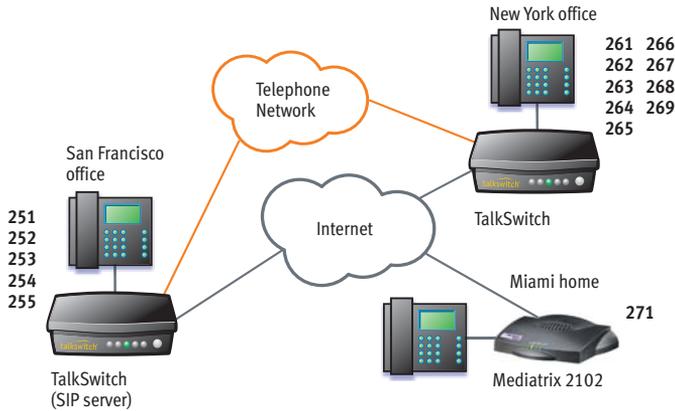
For VoIP calls made through a VoIP service provider, the service provider will assign the VoIP numbers to be used for their service.

To assist in assigning VoIP numbers for your network, use the 5.6 *VoIP network administration form* on page 186.

Assigning VoIP phone numbers in your TalkSwitch configuration is described in 5.4.3.1 *Assign VoIP phone numbers* on page 177.

Example: Continuing with the previous example, five VoIP location numbers are assigned to VoIP lines in San Francisco (251 to 255), nine are assigned to VoIP lines in New York (261 to 269), and one is assigned to the teleworker office in Miami (271). These phone numbers can be dialed directly from an extension or from an auto attendant from any location within the network.

Figure 119: TalkSwitch network configuration



5.4 VOIP CONFIGURATION

5.4.1 Configure TalkSwitch IP addresses

Each TalkSwitch system in a multi-location network must be configured with both local IP and public IP address information. These addresses are used to direct VoIP calls to the appropriate location. Each TalkSwitch unit in a system will have a different local IP address, and share a common public IP address.

5.4.1.1 Set the TalkSwitch local IP address

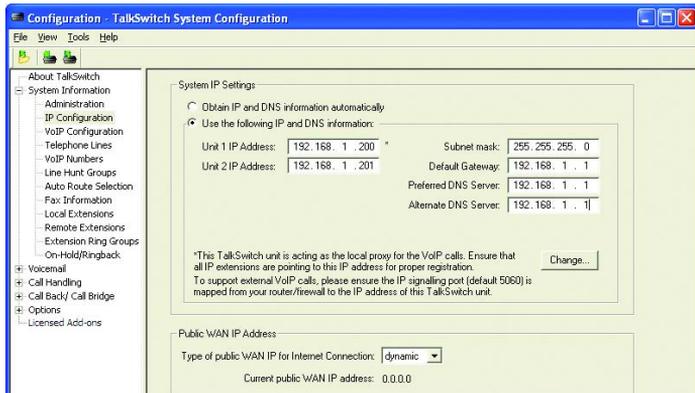
The following steps describe how to set the local IP address(es) of a TalkSwitch system:

1. Open the TalkSwitch configuration software by double-clicking the **TalkSwitch** icon on your Desktop or, from the Windows **Start** menu, select **Programs**, the **TalkSwitch** folder, and click **TalkSwitch**.
2. In the *TalkSwitch System Configuration* window, select **System Information -> IP Configuration** from the menu at the left.
3. TalkSwitch is factory configured to use DHCP (Dynamic Host Configuration Protocol) to automatically obtain its system IP settings. If the location has a DHCP server, which is typically a function of the router, the fields on the screen will automatically be filled in with the correct IP information. Click *Use the following IP and DNS information to*

'lock in' these settings and ensure they are saved as the system IP settings.

4. If the location does not have a DHCP server, the *System IP Settings* fields will not be filled in. Click *Use the following IP and DNS information* and enter the *IP address(es)*, *Subnet mask*, *Default Gateway*, *Preferred DNS Server*, and *Alternate DNS Server* information in the appropriate fields. For details on these settings, refer to 2.2.2 *IP Configuration* on page 39.
5. If there are multiple units in a system, enter the IP addresses of all units in the system (Unit 2 IP Address, Unit 3 IP Address, Unit 4 IP Address).

Figure 120: Local IP configuration



5.4.1.2 Set the TalkSwitch public IP address

The public IP address of a TalkSwitch system is assigned by your Internet Service Provider. Depending on the type of service, you will either have a dynamic or static IP address. If you are unsure which type you have, contact your Internet Service Provider.

The following steps describe how to specify the type of public IP address of a TalkSwitch system:

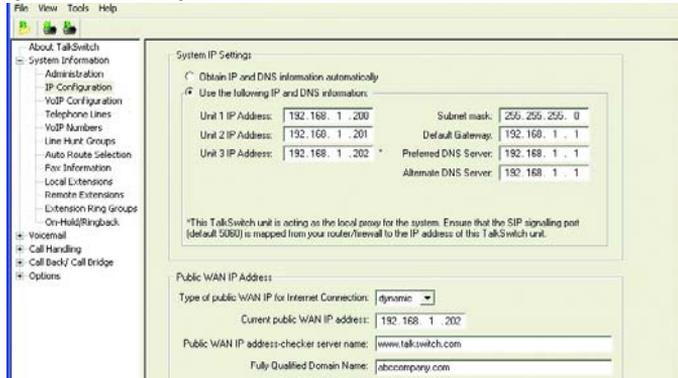
1. Specify the *Type of Public WAN IP for Internet Connection* by selecting either *dynamic* or *static* from the drop-down list. For private networks, please ensure the WAN IP address is the same as the Local IP address.

If this TalkSwitch system will act as the **SIP server** for a network, we recommend having a static Public IP address.

- If this TalkSwitch system will act as the SIP server for a network and has a dynamic Public IP address, enter the *Fully Qualified Domain Name* (FQDN) associated with the Public IP address. If the address is static, you can leave this field blank.

You can obtain a FQDN from a Dynamic Domain Name Service (DDNS) provider such as DynDNS (visit <http://www.dyndns.com/services/dns/dyndns/>). Check your router documentation to confirm which DDNS your router can support.

Figure 121: IP Configuration

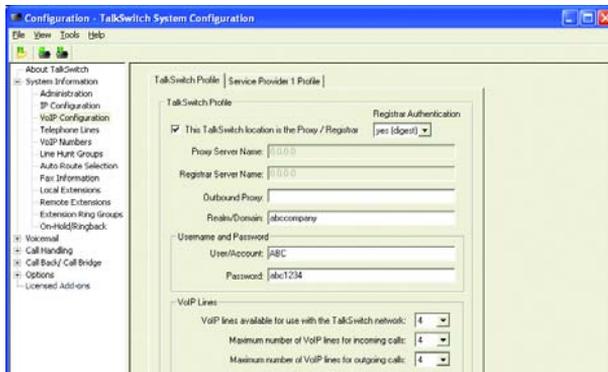


5.4.2 Configure TalkSwitch Profile

5.4.2.1 TalkSwitch Profile

Select **System Information -> VoIP Configuration**

Figure 122: TalkSwitch profile



TalkSwitch Profile tab

Use this tab to configure a TalkSwitch VoIP network.

TalkSwitch has a built-in IP server proxy/registrar/redirect. This facilitates the configuration and maintenance for multi-branch and telework applications.

A multi-location TalkSwitch system requires that one (and only one) unit be configured to act as the SIP server (Redirect Proxy/Registrar). For information on the SIP server, see 5.3.2 *Select which TalkSwitch to use as the SIP network server* on page 166.

This TalkSwitch location is the Proxy/Registrar:

If this location is designated to be the proxy/registrar server, click this checkbox. When it is enabled and other IP devices are registered with this system, any calls made within the group contact this device to resolve the destination location. Once the destination is resolved, the call is made directly from the source to the destination location.

The TalkSwitch registrar supports *digest* authentication. The digest mechanism is a challenge/response protocol in which the client presents its credentials in response to a challenge from the server. This method of authentication is very secure. To enable authentication, select *yes (digest)* from the drop-down list under registrar authentication. Using authentication will allow only authorized devices to access the network.

Proxy Server Name:

If this location is acting as the proxy server, it is filled in automatically.

If this location is not the proxy server, enter the IP address or domain name associated to the proxy server. If the server is using a different port number than the default, specify the port number after a colon at the end of the domain or IP address.

Example: 222.234.432.234:5061

Registrar Server Name:

If this location is acting as the registrar server, it is filled in automatically.

If this location is not the registrar server, enter the IP address or domain name associated to the registrar server. If a TalkSwitch is acting as the server, both the proxy server location and the registrar server location fields should have the same information. If the server is using a different port number than the default, specify the port number after a colon at the end of the domain or IP address.

Example: VoIP.domain.com:5061

Outbound Proxy:

If TalkSwitch is being provisioned with a VoIP service provider, enter the outbound proxy associated with the service provider.

Realm/Domain:

If this location is the proxy/registrar, you can assign a realm/domain name for your network. (e.g. abccompany.com). The *Realm/Domain* field may be required when TalkSwitch is being provisioned with a VoIP service provider. In this case, enter the realm/domain associated with the service provider. If this unit is not the proxy/registrar, it must have the same realm as the registrar server.

User Name and Password section

User/Account:

If this location is the proxy/registrar and you have enabled authentication, assign a user/account name to this unit. This account will be required by all other locations for registration purposes.

If this unit is not the proxy/registrar and if authentication is required, it must have the same user/account name as the registrar server.

Password:

If this location is the proxy/registrar and you have enabled authentication, assign a password to this unit. This password is required by all other locations for registration purposes.

If this unit is not the proxy/registrar and authentication is required, it must have the same password as the registrar server.



VoIP Lines

TalkSwitch offers the flexibility to share VoIP numbers with the TalkSwitch network and a service provider. You can reserve lines for use with one or the other depending on your requirements.

VoIP lines available for use with the TalkSwitch network:

By default, there are no restrictions on VoIP number use between the TalkSwitch and service provider profiles. If you need to give priority of availability to calls being placed in/out of this location on the TalkSwitch VoIP network, select an upper limit for the number of lines to use for each profile.

Maximum number of VoIP lines for incoming calls:

The maximum number of VoIP numbers available is determined by how many lines are available (shared) with the TalkSwitch VoIP network. For business reasons, you might want to make all lines available for incoming

calls and restrict the number of lines used for outgoing calls. It is recommended that at least one line be allocated for incoming calls.

Maximum number of VoIP lines for outgoing calls:

The maximum number of VoIP numbers available is determined by how many lines are available (shared) with the TalkSwitch VoIP network. For business reasons, you might want make all lines available for incoming calls and restrict outgoing calls so that no incoming calls are getting missed. It is recommended that at least one line be allocated for outgoing calls.

Codec Options (button)

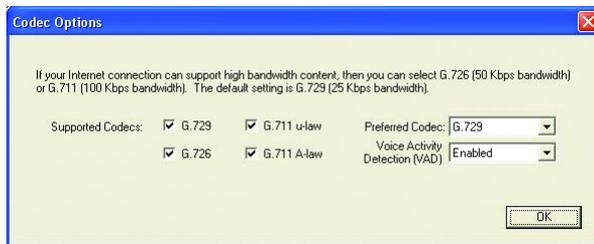
A codec is a method of compressing and decompressing audio signals for communication across digital networks. TalkSwitch supports 3 standardized VoIP codecs:

- G.729 (default — low bandwidth)
- G.726 (better voice quality — medium bandwidth)
- G.711 (best voice quality — high bandwidth)

One of these codecs must be selected as the *Preferred Codec*. By default G.729 is used to accommodate the maximum number of concurrent calls.

Voice Activity Detection (VAD) analyzes an audio stream to detect the presence or absence of speech and distinguishes speech from background noise. VAD can be used to reduce voice bandwidth when no speech is detected and reduce unwanted noise transmission.

Figure 123: Codec

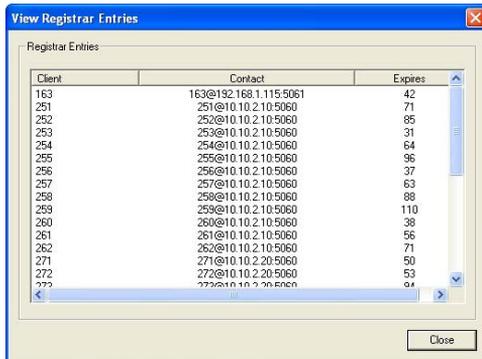


View Registrar Entries (button)

If this location is the designated SIP server (proxy/registrar), click this button to view which systems are registered with the SIP server. A window opens listing all TalkSwitch systems and gateways registered with this system. *Client* is the phone number assigned to a VoIP line. *Contact* is the information used by the SIP server to contact the client. It is comprised of

the phone number, public IP address and part number. *Expires* is the amount of time, in seconds, that the client is registered with the SIP server before having to re-register.

Figure 124: Registrar entries



The screenshot shows a window titled "View Registrar Entries" with a table of registrar entries. The table has three columns: Client, Contact, and Expires. The data is as follows:

Client	Contact	Expires
163	163@192.168.1.115:5061	42
251	251@10.10.2.10:5060	71
252	252@10.10.2.10:5060	85
253	253@10.10.2.10:5060	31
254	254@10.10.2.10:5060	64
255	255@10.10.2.10:5060	96
256	256@10.10.2.10:5060	37
257	257@10.10.2.10:5060	63
258	258@10.10.2.10:5060	88
259	259@10.10.2.10:5060	110
260	260@10.10.2.10:5060	38
261	261@10.10.2.10:5060	56
262	262@10.10.2.10:5060	71
271	271@10.10.2.20:5060	50
272	272@10.10.2.20:5060	53
272	272@10.10.2.20:5060	94

View Registration Status (button)

Click this button to see if this system is registered with the designated SIP server. A window opens listing all registered system VoIP phone numbers. *Client* is the phone number assigned to a VoIP line. *Status* is the registration status (registered or not registered).

Figure 125: Registration status



The screenshot shows a window titled "Registration Status" with a table of registration status. The table has two columns: Client and Status. The data is as follows:

Client	Status
251	Registered
252	Registered
253	Registered
254	Registered
255	Registered
256	Registered
257	Registered
258	Registered
259	Registered
260	Registered
261	Registered
262	Registered

5.4.2.2 Service Provider profile

Service Provider Profile tab

TalkSwitch can register with VoIP service providers to make and receive calls using their service. Visit www.talkswitch.com/support to access a list of service providers that have been certified interoperable with TalkSwitch.

Service Configuration Guides are also available for download. They contain all the specific necessary information to complete your configuration.

Information in those documents should be used instead of this guide. If there is no configuration guide for your service provider, follow the instructions below.

Figure 126: Service Provider profile



Service Provider Name:

Enter the name for your VoIP service provider. The name will be displayed elsewhere in the configuration software in areas related to VoIP service.

Proxy Server Name:

Enter the IP address or domain name associated with the proxy server, exactly as specified by your service provider. If the server is using a different port than 5060, specify the port number after a colon at the end of the domain or IP address.

Example: 222.234.432.234:5061

Registrar Server Name:

Enter the IP address or domain name associated with the registrar server, exactly as specified by your service provider. If the server is using a different port number than 5060, specify the port number after a colon

at the end of the domain or IP address.

Example: VoIP.domain.com:5061

Outbound Proxy:

Enter the outbound proxy (if required) associated with the service provider.

Realm/Domain:

Enter the realm/domain (if required) associated with the service provider.

VoIP Lines

The descriptions for the VoIP lines under the *TalkSwitch Provider Profile* tab below are the same as described in *VoIP Lines* on page 172.

- VoIP lines available for use with this Service Provider
- Maximum number of VoIP lines for incoming calls
- Maximum number of VoIP lines for outgoing calls

Codec Options (button)

See *Codec Options (button)* on page 173 for a description.

View Registration Status (button)

See *View Registration Status (button)* on page 174 for a description.



5.4.3 VoIP Numbers

Each VoIP-capable TalkSwitch unit supports up to 12 VoIP phone numbers. You can configure unique call handling scenarios for each VoIP number.



You must associate each VoIP number with either the TalkSwitch profile or the service provider profile. Any VoIP number may use any VoIP line (i.e. trunk) to make or receive a call.

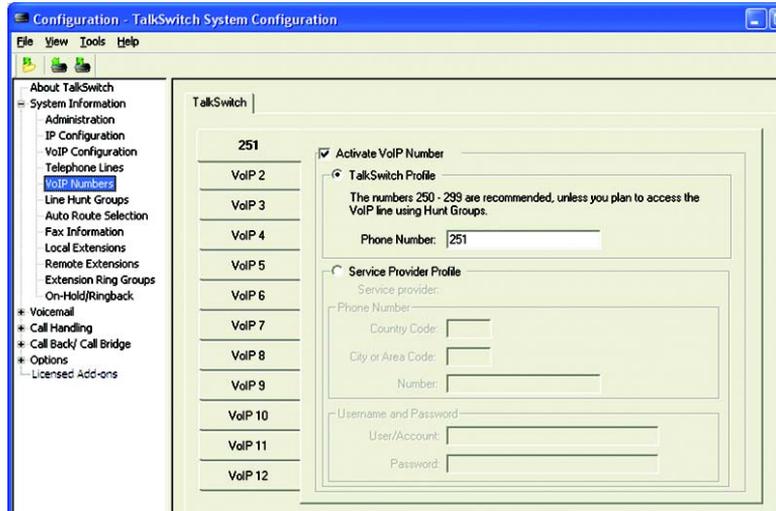
Example: You might assign a general number for customers to reach the main auto attendant greeting and configure up to 11 other numbers so that the inbound calls will ring straight through to specific extensions in the office. A VoIP-capable TalkSwitch unit can support up to four simultaneous VoIP calls. VoIP numbers can be three or more digits long.

Each VoIP phone number assigned in a multi-location network must be unique (i.e. only used for one location). For convenience, TalkSwitch supports 50 VoIP location numbers (250 to 299) that permit direct dialing from any extension or auto attendant in the network.

If needed, you can also add numbers outside of that range and access them using hunt groups. You can assign these numbers when the 3-digit VoIP numbers have already been assigned or reserved. To do this:

1. Under **System Information -> VoIP Numbers**, select an available VoIP number and assign it to a unique number (i. e. 4004).
2. Select **System Information -> Line Hunt Groups**.
3. Ensure that you have at least one line hunt group configured (i.e. 88) to *Line Type: TalkSwitch VoIP Numbers*.
4. Callers can now dial **88** from any extension to get a VoIP dial tone and then enter **4004** to reach the VoIP location.

Figure 127: TalkSwitch profile in System Information



For VoIP calls made through a service provider, the service provider will assign the VoIP numbers to be used for their service. To assist in assigning VoIP numbers for your network, use the *5.6 VoIP network administration form* on page 186.

5.4.3.1 Assign VoIP phone numbers

TalkSwitch Profile section

All VoIP numbers are inactive when the system is first configured.

1. In the *TalkSwitch System Configuration* window, select **System Information -> VoIP Numbers**.
2. Select *VoIP 1* and check the *Activate VoIP Number* box, the default profile for entering a phone number is the *TalkSwitch Profile*.
3. Phone numbers 250-299 have special functionality with TalkSwitch. They can be dialed directly from any extension or auto attendant and are routed automatically via VoIP to the destination with the number assigned to one of its ports. Part of the role of an administrator is to keep track of all phone numbers assigned to each location to prevent duplicate numbers in multiple locations. We recommend that you use the numbers 250–299. It will facilitate calling from extensions and auto attendants.

Ensure that each number assigned is unique in the network.

For each TalkSwitch unit with VoIP lines on the network, you can assign up to 12 VoIP phone numbers, increasing the number of inbound VoIP call handling options available.

4. Repeat the above steps for VoIP numbers 2-12.
5. For systems with multiple units, repeat the process for each TalkSwitch unit (click on the tabs labeled *TalkSwitch 1*, *TalkSwitch 2*, *TalkSwitch 3*, and *TalkSwitch 4*).

5.4.3.2 Configure call handling for VoIP numbers

Configuring call handling for VoIP numbers customizes TalkSwitch to fit the unique needs of your business. These call handling options are similar to the options available for your incoming telephone lines. Perform the following steps to configure VoIP call handling options for each networked TalkSwitch unit with VoIP lines.

1. In the *TalkSwitch System Configuration* window, select **Call Handling -> VoIP Numbers**.
2. Select your first VoIP number (It will be labeled with the number you assigned in *Assign VoIP Phone Numbers* above).
3. From the drop-down list next to *On an incoming call during mode:*, select the mode to configure (for example, *Mode 1*).
4. From the drop-down lists, select *Ring extensions only*, *Play auto attendant*, or *Go to voice mailbox*.

- For *Ring extensions only*, select the extensions from the list (by default they are all selected).
 - For *Play auto attendant*, select which auto attendant to play (for example, 1), the number of rings to take place before the auto attendant answers (for example, after 2 rings), and the extensions to ring before the auto attendant answers. If you want the auto attendant to answer all calls immediately without ringing extensions, select *immediately* rather than the number of rings.
 - For *Go to voice mailbox*, select the mailbox number (for example, 111) and the number of rings to occur before the call goes to the voice mailbox.
5. Repeat Steps 3 and 4 for each mode.

Figure 128: Call handling



6. Repeat this process to configure the VoIP numbers 2-12.
7. For systems with multiple units, repeat the process for each TalkSwitch unit (click on the tabs labeled *TalkSwitch 1*, *TalkSwitch 2*, *TalkSwitch 3*, and *TalkSwitch 4*).

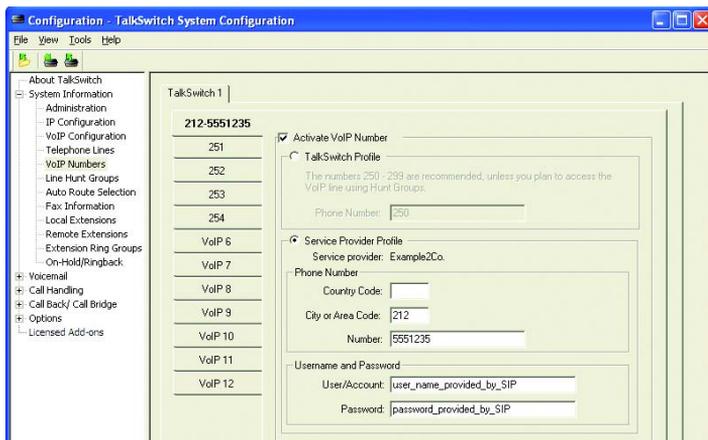
5.4.3.3 Assign service provider phone numbers

1. In the *TalkSwitch System Configuration* window, select **System Information ->VoIP Numbers**.
2. Select 1 and check the *Activate VoIP Number* box, the default profile for entering a phone number is the *TalkSwitch Profile*. Select *Service Provider Profile*.

3. Enter one of the phone numbers, exactly as provided by your VoIP service provider. If your service provider requires a country code, enter it, otherwise leave it blank.
4. Enter the *User/Account* information and *Password* for this particular phone number.
5. Repeat the above steps for VoIP numbers 2-12 as is required.

For systems with multiple units, repeat the process for each TalkSwitch unit (click on the tabs labeled *TalkSwitch 1*, *TalkSwitch 2*, *TalkSwitch 3*, and *TalkSwitch 4*).

Figure 129: Service provider profile in System Information



5.5 FAQ

How does VoIP work?

Voice over IP (VoIP), sends calls over the Internet or any Internet Protocol (IP) network by converting voice traffic into data packets. These packets are routed over the IP network the same way as other data, then reassembled at the point of arrival and converted back to voice.

Does a VoIP call sound like a regular phone call?

Under the right conditions, VoIP calls sound as good or better than regular phone calls. Voice quality is affected by numerous factors, including how it is compressed and decompressed for delivery over the Internet and how packets are processed. A few steps can help ensure call quality.

- **High-speed connections on both ends of the call**
Good quality VoIP requires a broadband connection to support more simultaneous VoIP calls. Please visit our site at http://www.talkswitch.com/voip/voip_test.asp to determine what your Internet connection can support. Good quality VoIP is not possible over a dial-up modem connection.
- Choose the right codec for your location
 1. The default codec is **G.729** (8 Kbps), using approximately 25 Kbps bandwidth upstream and 25 Kbps bandwidth downstream for each call. G.729 provides very good call quality while minimizing bandwidth usage.
 2. The **G.726** (32 Kbps) codec is a better quality solution compared to the G.729 codec. However, it requires more bandwidth per call. A G.726 call typically requires 50 Kbps bandwidth upstream and 50 Kbps bandwidth downstream for each call.
 3. The **G.711** (64 Kbps) codec provides the best voice quality. The trade-off is the bandwidth requirement. G.711 calls typically require up to 100 Kbps bandwidth upstream and 100 Kbps bandwidth downstream.

What happens if the power goes out or if the IP network to VoIP fails?

To ensure a reliable network connection, all elements of the VoIP network should be connected to back-up power supplies (UPS). These elements might include LAN switches, routers, firewalls, broadband connection devices (i.e. cable modems, DSL modems), and VoIP devices. If the power goes out at the Internet Service Provider, then no VoIP calls can be made. Calls can still be placed over the incoming telephone lines.

Can a firewall prevent VoIP calls from passing through?

The purpose of a firewall is to control what kind of traffic enters and leaves your network. TalkSwitch is designed with embedded applications to help traverse firewalls properly. To allow VoIP calls to pass through your firewall, you may need to use the port forwarding feature on your firewall.

TalkSwitch uses the following ports for VoIP:

Format	Type	Unit 1	Unit 2	Unit 3	Unit 4
RTP (voice):	UDP	6000-6006	6010-6016	6020-6026	6030-6036
SIP (signaling):	UDP	5060	This port is mapped to only the SIP server unit.		

What is SIP?

The Session Initiation Protocol (SIP) is a signalling protocol used for establishing sessions in an IP network. A session could be a simple 2-way telephone call or it could be a collaborative multi-media conference session.

Over the last couple of years, the Voice over IP community has adopted IP as its protocol of choice for signalling. IP is an RFC standard (RFC 3261) from the Internet Engineering Task Force (IETF), the body responsible for administering and developing the mechanisms that comprise the Internet.

The IETF's philosophy is simplicity: specify only what you need to specify. IP just initiates, terminates and modifies sessions. In this guide, the term IP is often used in place of VoIP.

TalkSwitch phone systems use the Appello® SIP Stack for VoIP communications. The Appello® SIP Stack was developed by TalkSwitch, a division of Centrepont Technologies Inc.

What is a VPN?

VPN stands for Virtual Private Network. It is a means of having the advantages of a private network on a shared public infrastructure like the Internet.

Can a VPN help to carry data securely over the Internet?

A VPN provides security by using encryption/decryption. Using a process called tunneling, private data is encrypted and encapsulated before being sent across the network. The packets of information are sent to their destination. Only the ones with the correct protocol information can enter a company's network.

A VPN is most useful for providing employees with remote access to the company network without compromising security. Encryption can have a negative impact on the call quality because of the increased overhead on the network connection. IPSec adds approximately 10% overhead to VPN traffic.

What's the difference between a public IP address and a private IP address?

A public IP address is a globally unique number that identifies a device on the Internet. If you want someone on the Internet to connect to you, they must know your public address. Also known as your real or external address.

Private IP addresses are typically assigned to devices on a Local Area Network (LAN) and are not routable outside the LAN. Private IP addresses are usually in the range 192.168.x.x, 172.16.x.x or 10.x.x.x. These IP addresses are used where you have multiple computers all sharing the same Internet connection.

To access the Internet, a computer or VoIP device must have an IP address. So what do you do if your ISP (Internet Service Provider) has provided you with only one IP address but you have more than one device that requires Internet access? The solution is to give each of the devices on the LAN a private IP address. A router makes them work by performing Network Address Translation (NAT — See the next topic for details).

Addresses of the form 192.168.x.x, 172.16.x.x or 10.x.x.x are not recognized on the Internet and can only be used for private networks. For example, if you tell someone on the Internet to connect to you using a 192.168.x.x address, it will not work. Instead, you must provide your real/public IP address.

What is NAT and how does it affect VoIP?

Network Address Translation (NAT) allows multiple devices to share the same external IP address to access resources on the Internet. NAT is typically used to allow all the devices in a subscriber's local area network to access the Internet through a router with a single public IP address assigned by the Internet Service Provider.

If a VoIP device is sitting behind a NAT, the private IP address assigned to it is not usable for communications with the entities outside the private network. The VoIP device must substitute the private IP address information with the proper external IP address/port in the mapping chosen by the underlying NAT to communicate with a particular public peer address/port. TalkSwitch can automatically check for the current public IP address and substitute the private IP address with the public address so that VoIP traffic is properly routed through the NAT.

What's the difference between a static IP address and a dynamic IP address and what effect does this have on VoIP?

A static IP address is an address that is permanently assigned to a device. Typically, a static IP address has to be assigned by the network administrator or Internet Service Provider (ISP).

A dynamic IP address is an address that is temporarily assigned to devices by a DHCP or PPP server that maintains and assigns a pool of IP addresses.

When deploying VoIP devices, it's best to have a static public IP address. This prevents destinations from becoming unreachable for periods of time when the IP address changes. But since this can be an expensive option, TalkSwitch supports the ability to handle dynamic public IP addresses for the Internet connection. TalkSwitch automatically checks for the current public IP address and substitutes the private IP address with the public address, so that VoIP traffic is properly routed through the NAT.

What is a IP Proxy and Registrar?

The basic IP architecture is client/server in nature. The main entities in the IP server are the proxy, redirect server and the registrar.

TalkSwitch has the IP server fully integrated for easy deployment and maintenance of your VoIP network. If you are deploying a multi-site configuration which can comprise of TalkSwitch units, certified third-party gateways and certified third-party IP phones, one TalkSwitch can act as the IP proxy and registrar. The other IP end-points will register with this TalkSwitch and all calls will be negotiated through this TalkSwitch.

When you first set up your VoIP device (TalkSwitch unit, gateway or phone), it will register itself with the TalkSwitch that is configured as the IP server so that calls can be directed to your location.

When you initiate a VoIP call, your device (TalkSwitch unit, gateway or phone) will contact the TalkSwitch that is configured as the IP server to obtain the contact information for the destination you are trying to reach. Once the information is received, the call is established directly between your location and the destination. The IP server is no longer involved in the call.

If the location of the IP proxy and registrar has a dynamic IP address, you have to use the dynamic DNS since the IP server location becomes unreachable as soon as the IP address changes.

What is Dynamic DNS?

Dynamic DNS is a Domain Name Service used with dynamic IP addresses. DNS is used to allow devices to find other devices on the Internet by name rather than by IP address. Dynamic DNS allows you to use a Fully Qualified Domain Name (FQDN) to locate a device with a dynamic IP address. The FQDN does not change. There are companies on the Internet who can provide you with a DNS address, even if your IP address changes. Such companies are www.dyndns.org, hn.org or www.no-ip.com. Please consult with them how to use their service. The basic idea is that you will receive an address like myname.homeip.net which points to your IP address (i.e. 209.210.230.140). This is important for locations where the IP server resides and where the public IP address is dynamic. All other locations will point to this IP server's domain instead of an IP address.

What is the drawback of using dynamic DNS?

When DNS records are updated, they have to be sent all over the world so that everyone will be aware of the changes. Basically, one DNS server connects to another and swaps information. In many cases, the DNS servers are updated very quickly. Updates should not take longer than 15 minutes to propagate throughout the DNS system. It is possible that propagation could

take longer due to unforeseen traffic but it shouldn't take much more than an hour at the most.

How often will my public IP address change?

This depends on the policies of your Internet Service Provider (ISP, the company that provides your Internet access). It can be a matter of days, weeks or even months before your IP address changes. A power loss or reboot to your cable or DSL modem usually results in the assignment of a new IP address.

TalkSwitch has the technology that helps keep this information up to date so that the system can function properly after an IP address change has been detected and updated.

What is a port number?

In an IP address like 64.26.209.103:5060, the port number is 5060, the number after the colon.

Port numbers are required so that a computer or device can support multiple applications or streams of simultaneous data communication. Each stream or application uses a unique port number. You can think of it like mailing a letter to a friend in an apartment building at a specific address. You need to specify the apartment number as well as the address of the building, so that the mail can reach the correct final destination.

5.6 VOIP NETWORK ADMINISTRATION FORM

Use this form to keep track of phone numbers assigned to units that are part of a TalkSwitch VoIP network.

SIP server location _____

Equipment at SIP server location: _____

VoIP phone numbers assigned to SIP server location: _____

Telephone numbers assigned to unit

Location name	Equipment	VoIP phone number(s)

We trust that your **TalkSwitch phone system** will provide exceptional features, performance and value to your business. Should you have any further questions, please contact your authorized TalkSwitch reseller . If you purchased directly from TalkSwitch, visit **www.talkswitch.com/support** or call **1.866.393.9960** (North America).

We welcome your feedback, comments and suggestions. Please e-mail us at **info@talkswitch.com** or write us at **TalkSwitch, 1545 Carling Avenue, Suite 510, Ottawa, ON Canada K1Z 8P9.**

Thank you for choosing TalkSwitch

CHAPTER 6

TROUBLESHOOTING AND SUPPORT

6.1 TROUBLESHOOTING

Solutions to the following problems are described in this chapter:

- Problems that may occur during configuration
- Problems that may occur while using the TalkSwitch features

Some problems might be due to physical connections such as loose cables. Before you start to analyze possible configuration issues. See *1.15 Verifying the connections* on page 29.

6.1.1 Problems that may occur during configuration

I am not able to retrieve settings from TalkSwitch.

- Reboot TalkSwitch.
On the TalkSwitch front panel, turn the power button off and back on.
- Make sure you do not have any communications programs running at the same time you are using the TalkSwitch configuration software. These can include Palm Pilot, Hot Sync, TalkWorks, digital camera software or others. These programs tend to occupy COM ports, making them unavailable for other programs.

I am unable to configure TalkSwitch using a touchtone phone.

1. You cannot configure TalkSwitch using a phone sharing the same line in parallel. You have to configure TalkSwitch using one of the extensions plugged into the back panel of TalkSwitch.

2. You cannot configure TalkSwitch using touchtone keypad commands while the TalkSwitch configuration software is open. If the software is closed and you hear the prompt *"I'm sorry, TalkSwitch is currently being configured"*, reboot TalkSwitch by turning the power off and then back on. Wait until the lights stop flashing and the **Data** light is on solid.
- If you are using a local extension, check if the extension has **Direct Line Access** to your incoming telephone lines. If **Direct Line Access** is enabled, press Flash on an analog phone or *** * *** on a IP phone to receive an internal dial tone. Press **#** to enter command mode. Enter the correct password, if you have password protection.
- From an out-of-office phone, call into TalkSwitch and wait for the the auto attendant. Dial **#** or *** 55** to enter command mode. Enter the correct password, if you have password protection.

6.1.2 Problems that may occur while using the TalkSwitch features

6.1.2.1 The auto attendant

The auto attendant does not play when calls come in.

- Check if the line light on an incoming call is flickering while the phone is ringing. If not, replace the phone cord.
- Make sure you have an auto attendant message recorded.
- Check the mode TalkSwitch is currently running. Open the configuration software. In the lower right part of the **About TalkSwitch** window, check what it says for *Current Mode*. Select **Call Handling -> Telephone Lines**.
- *On an incoming call during mode:* change the mode if necessary. Set the auto attendant to answer *immediately*.

The auto attendant message is broken up or very faint.

- Open the configuration software. Select **Options -> Audio Controls**. Adjust the *Line Controls* volume setting.
- The quality of the microphone in the telephone handset that you used to record your auto attendant can affect the quality of your recording. Try recording the auto attendant from another extension/telephone.

The auto attendant answers calls before any of the extensions ring.

If TalkSwitch users wish to have their local extensions ring before the auto attendant picks up incoming calls, do the following:

- Open the configuration software.
Select **Call Handling -> Telephone Lines**.
Click the checkbox(es) in the section *Extensions to ring on an incoming call before the Auto Attendant answers*.
Increase the number of rings.

The auto attendant is transferring calls to the wrong extension.

- Check if the extension jacks of your devices (phones, faxes) are plugged into the corresponding jacks on the TalkSwitch back panel (i.e. extension 111 is plugged into E1).

The auto attendant answers calls, but does not transfer them to the extensions.

- Make sure your extensions are plugged into the extension jacks on the back panel of TalkSwitch (instead of your telephone wall jacks).
- Open the configuration software.
Select **Call Handling -> Auto Attendant**.
In the *At the Auto Attendant, perform the following actions* section, check if the options that the caller can select are pointing to the correct extensions.

6.1.2.2 Music on hold

Callers hear the “One moment please” message and then total silence when the auto attendant transfers their call to an extension.

- There is a problem with the music source. Check all connections and the power.

Callers hear only silence when put on hold at an extension.

- Make sure you use the Flash or **Link** button on an analog phone, or the **Hold** button on a IP phone, to put callers on hold. If you use the **Hold** button on an analog phone, callers will be placed on hold at the phone itself and not through TalkSwitch.
- You have enabled the music on hold feature, but have not attached an audio device to the music jack on TalkSwitch.
- You have enabled play the *Play music from file* feature but have not loaded a .wav music file to TalkSwitch.

See 1.4 *Unit back panel* on page 3 for more details on the Music on Hold port.

6.1.2.3 Call routing with local extensions and home phones

Callers are disconnected when transferring calls from one extension to another.

- If you are using the hook switch rather than the **Flash** button to put callers on hold, you may be holding the hook switch down too long and disconnect.

I can't use my conference/3-way calling feature from the phone company.

- You need to make sure you have checked the box indicating that you have 3-way calling or transfer and clear on the appropriate lines under **System Information -> Telephone Lines** in the configuration software.

I'm unable to place intercom calls from a local extension.

- If the local extension has been set up for **Direct Line Access** to your telephone lines, you need to press **Flash** on an analog phone, or *** *** on a IP phone, before intercom calling.

6.1.2.4 Answering and fax machines

The answering machine and/or fax machine picks up calls before they can be answered by telephone extensions.

- Check your settings under **Call Handling -> Telephone Lines** in the configuration software. Your fax machine and/or answering machine extension may be set to answer calls at the same time as your telephone extensions. To fix this problem, remove the fax or answering machine extension from the list of extensions to ring on an incoming call.

Incoming faxes are not automatically detected and routed to the fax machine.

- ensure that you selected the correct extension for the field labeled *if a fax call is detected*, under **Call Handling -> Auto Attendant** in the configuration software.
- Not all fax machines emit a CNG tone that TalkSwitch can detect and route. Therefore, you may consider incorporating an additional method of fax routing. This could include the use of a dedicated fax line or a distinctive ring number.
- Many fax machines/modems will time-out or disconnect after 25 seconds on average if they have not connected with another fax machine/modem. If you clicked the checkbox next to *After the Auto Attendant has finished playing and no selection has been made within*, under **Call Handling -> Auto Attendant**, the fax call may time-out before it reaches your fax extension. If you have a long auto attendant message, you may want to shorten it or use another fax routing option.

- The level of the auto attendant is too loud and is overpowering the CNG tone, affecting the detection. Try lowering the volume level of the auto attendant.

6.1.2.5 Other possible local extension problems

I am unable to access TalkSwitch lines with a local extension.

- Check your **Direct Line Access** settings under **System Information -> Local Extension** in the configuration software. If a local extension has not been given **Direct Line Access** to your incoming telephone lines, you must dial **9** or **81-88** to access a line.
- Check if *Activate Automatic Route Selection* is enabled under **System Information-> Auto Route Selection**. ARS can re-direct or block calls.

My PC (internal or external) modem cannot dial out.

- If **Direct Line Access** to incoming telephone lines is not enabled on the extension where your modem is connected, you will have to include a line hunt group prefix (**9** or **81-88**) before the dialing the modem number.

One of the local extensions (telephones) does not ring.

- Some telephones require more voltage in order to ring. These phones are usually the older 'Bell' phones with mechanical ringers. These phones should not be used as TalkSwitch extensions.
- Make sure the telephone cord you are using between the phone and TalkSwitch is working properly.
- Ensure that the ringer is turned on.

Extension(s) ring, but there is no caller.

- This can occur when a caller hangs up after the auto attendant has begun to play. In some areas, the telephone company's disconnect/clear signal is weak. The auto attendant may not be able to pick up the disconnect signal as soon as a caller hangs up. The auto attendant will not receive a response after playing its message and the call will fall through to your settings for *After the Auto Attendant has finished playing and no selection has been made within*, under **Call Handling -> Auto Attendant**,

When I try to access voicemail, I hear a busy tone.

- Check your **Direct Line Access** settings under **System Information -> Local Extensions** in the configuration software. If a local extension has been given **Direct Line Access** to your telephone lines, you must first press **Flash** on an analog phone, or *** * *** on a IP phone, before dialing the mailbox. The same applies for dialing other extensions to initiate intercom calls.



6.1.2.6 Multiple TalkSwitch units connected to the same LAN

I can't reach an extension across the LAN.

- Make sure the other unit is connected to the LAN and powered on.
- Pick up a local extension belonging to the other unit. If you hear a dial tone, check your LAN connections. On the rear panel, the top LED of the three LEDs next to the LAN port should be on. If there is no dial tone, make sure the unit is powered on.

The configuration does not show the other TalkSwitch unit(s).

- Check your LAN connections. Are you able to communicate with another extension on the other unit?
- Check the unit ID of each unit. If the lights on one or more units are flashing, there is a conflict with the unit ID numbers assigned to each TalkSwitch unit. Connect a phone to each TalkSwitch unit and lift the handset. It will tell you immediately if a unit ID number needs to be assigned. Follow the instructions. You can dial **# 00 #** to check the unit ID of the unit you are connected to. Click the *System Information* link in the TalkSwitch configuration software.

After recording an auto attendant greeting, I can't play it back.

- After recording an auto attendant on one of the TalkSwitch units, it will copy it to other TalkSwitch units on the LAN. During this process, you cannot listen to that particular auto attendant. Try again in a few minutes.

Calls are not always reaching my voicemail.

- When the system is in high use, all channels across the LAN might be in use, so that calls cannot be transferred to extensions or voicemail. If this situation happens frequently, consider optimizing the way the system uses the lines under **System Information -> Line Hunt Groups** in the configuration software. Click the checkbox next to *For all Hunt Groups, hunt lines on same unit first*. Also, ensure that frequently used extensions are connected to units with enough lines out to reduce LAN traffic.

Example: You have two TalkSwitch units, six incoming telephone lines and 12 employees. The Six lines have the hunt/rollover service from the phone company. Suggested setup: Plug the four highest traffic lines into unit 1. Also, connect the eight highest phone usage employee extensions to unit 1. Connect the remaining lines and extensions to unit 2.

Under **System Information -> Line Hunt Groups** in the configuration software, make sure you have the *For all Hunt Groups, hunt lines on same unit first* checkbox enabled. This arrangement keeps most of the traffic on unit 1 and any outbound calls made from any extension will go out on the same unit the extension is connected to. This also applies for

inbound calls that are forwarded to remote extensions. The call will be forwarded out on a line belonging to the same unit that the call came in on. The call will only be forwarded on the other TalkSwitch unit if no incoming telephone line on that first unit is available.

Calls across the LAN are lower in volume.

- Check the line gain levels under **Options -> Audio Controls** in the configuration software. You may need to increase the line gain levels on one or more lines to compensate for high loss lines. If these controls are set incorrectly and your lines are problematic, you may hear an echo when calls are made across the LAN.



Problems that may occur with VoIP

I hear a re-order tone when I try to dial a VoIP location.

- Confirm the number dialed is actually part of the VoIP group (i.e. registered with the same proxy/registrar server as your TalkSwitch, IP gateway or IP phone is registered with.
- Check that TalkSwitch has all proper settings in the IP configuration and VoIP configuration pages.
- Verify that the number you are dialing is registered with your TS proxy registrar. Go to **System Information -> VoIP Configuration** and click the **View Registrar Entries** button.
- Contact our customer support department by e-mail:
www.talkswitch.com/supportsupport@talkswitch.com.

Callers complain the sound is distorted or choppy.

- Your broadband connection may not have enough upstream bandwidth to support many simultaneous VoIP calls. Under **Options -> Audio Controls**, you may need to disable the G.711 and G.726 codecs and only use G.729.
- You may have too much data traffic on your Internet connection at the same time you are trying to make voice calls. We suggest that you use a router that supports QoS (Quality of Service) for VoIP.

When I call someone or they call me, voice is only heard in one direction.

- The cause of this problem is usually a result of a router being misconfigured with respect to port mappings. Ensure all required VoIP ports are mapped to your TalkSwitch unit. Also, ensure you use a static private IP address when connected to your router as this can also affect port mappings from the firewall/router.
- Use a router that supports the UPnP.

For more VoIP information, visit our support section at <http://www.talkswitch.com/support>.

6.2 SUPPORT

If you are having problems with the configuration or operation of TalkSwitch, please read this section. If you don't find the solution to your problem here, check the FAQ section located in the support area at our website at <http://www.talkswitch.com/support>. Select your TalkSwitch model and click **Continue**.

If you require further assistance, do one of the following:

1. Contact your TalkSwitch dealer.
2. Contact our customer support department by e-mail: support@talkswitch.com.
3. Call a technical support specialist at TalkSwitch:
1-866-393-9960 (continental North America) or **613-725-2466**
weekdays between 9:00 a.m. and 8:00 p.m. EST.

Please have the following information available when you call:

1. Your e-mail address and telephone number.
2. The serial number located on the bottom of TalkSwitch.
3. Your date of purchase.
4. The number of telephone lines that you have connected to TalkSwitch.
5. The telephone company services that you are using with TalkSwitch.
6. Your PC's operating system (e.g. Windows 95, 98, ME, 2000, XP).

APPENDIX A

FUNCTIONS AND COMMANDS

About functions and commands

The following tables list numbers, functions and commands you can enter. The first part lists all the possible analog, IP, remote and ring group extensions as well as VoIP numbers and general voice mailbox numbers. The following tables list functions that can be entered from local and outside phones and at the auto attendant. The last table is a list of commands you can enter in command mode.

Extension numbers

	Unit 1	Unit 2	Unit 3	Unit 4
Local analog/IP extensions	111-118	121-128	131-138	141-148
Remote extensions	210-219	220-229	230-239	240-249
Extension Ring Group	300-309	There are 10 ring groups for the entire system.		

Other 3-digit numbers

VoIP location numbers	250-299	There are 50 external VoIP numbers.		
General voice mailboxes	410-419	420-429	430-439	440-449

Functions you can access from local extensions

Keys	Function performed (Flash for Analog phone, Hold for IP phone)
Flash/Hold	If you are connected to a caller, Flash/Hold puts the call on hold and you hear the internal dial tone.
Flash/Hold	Pressing Flash/Hold again at the internal dial tone retrieves the most recent call on hold.
Flash/Hold at external dial tone	When you are connected to an outside line, pressing Flash/Hold switches you to the internal dial tone.
 On a regular phone, press Flash . On an IP phone, press Hold .	
* * * * * and mailbox number (local, remote or general voicemail)	Dials into a local, remote or general voice mailbox to leave a message.
* * * * * and mailbox number (local, remote or general voicemail)	Accesses voicemail of a local, remote or general voice mailbox. You can retrieve messages, change greetings, password, etc.
* * * #	Accesses the mailbox(es) associated with your extension.
Flash/Hold 4	Completes a transfer and returns to the internal dial tone.
 On a regular phone, press Flash 4 . On an IP phone, press Hold 4 followed by Dial, Send or # to complete the transfer.	
* 0	External PA access to make an announcement through the attached external PA system.
Flash/Hold 5	Disconnects the current caller and reconnects to the last caller on hold. A conference-call initiator can disengage the second conferenced party by pressing Flash 5 .
Flash/Hold	There are 10 parking spots for the entire system. Press Flash/Hold to place a call on hold and dial
* 500-509	* 500-509 to assign a parking spot (park orbit).
* 510	Parks a call automatically in the first available parking spot, starting at 500. Press Flash/Hold and * 510 to place a caller in the next available park orbit.
* *	Retrieves a parked call.
500-509	Dial from a local extension to pick up a parked call.

Functions you can access from local extensions

Keys	Function performed (Flash for Analog phone, Hold for IP phone)
Flash/Hold 6	Conference call for 3 parties. Press Flash to put the first caller on hold, dial another extension or external number and press Flash 6 .
* 60	Disables <i>Do Not Disturb</i> at the current extension.
* 61	Enables <i>Do Not Disturb</i> at the current extension.
* 62	Toggles <i>Do Not Disturb</i> at the current extension.
* 7 and extension	Call pick-up at a specific extension. Dial from dial tone to pick up a call ringing at another extension.
Flash/Hold 7	Press Flash 7 to cycle through queued calls on a first in/first out basis.
80	Accesses <i>Same Line Connect</i> when transferring an outside caller to another outside number.
9, 81–88	Dial to access an outside line.
 On an IP phone, press Dial , Send or # after your entry for the following options:	
* 80	Enable play music source through PA output.
* 84 and extension	Hands-free intercom call to an extension, automatically answered in speaker mode, enabling instant hands-free 2-way communication (for supported sets).
* 85 and extension or ring group number	Page a phone at an extension or the phones of a ring group.
* 88 and account code	Attach account code to Call Detail Record (CDR) for last call .
* 9	Call pick-up of any ringing line.
# or * 55	Enters command mode.

Functions you can access from outside phones

The list below shows the digits you can dial after the auto attendant answers.

Keys	Function performed
0	Performs the action defined in the auto attendant configuration (see <i>2.4.2 Auto attendant</i> on page 83 for details).
3-digit local ext.	Rings at the local extension (i.e. 111). See <i>Extension numbers</i> on page 195.
3-digit remote ext.	Rings at the remote extension (i.e. 211). See <i>Extension numbers</i> on page 195.
300–309 Extension Ring Groups	Rings all the extensions in the ring groups. There are 10 ring groups for the entire system.
1, 2, 3, 4 and 5	Performs the action defined in the auto attendant configuration (see <i>2.4.2 Auto attendant</i> on page 83 for details).
6	Enters call back configuration (see <i>2.5 Call Back/Call Bridge</i> on page 102 for details).
7 and extension	Transfers a caller on hold to another phone system used in conjunction with TalkSwitch. As soon as the call connects to the other system, it disconnects from TalkSwitch.
9 and 81–88	Accesses line hunt groups for call bridge and is protected with the administrator password.
80	Accesses the call bridge feature using <i>Same Line Connect</i> . Requires 3-way calling/conference on the line.
* * * and extension (local, remote or general voicemail)	Dials into a local, remote or general voice mailbox to leave a message.
* * * and mailbox number (local, remote or general voicemail)	Accesses voicemail of a local, remote or general voice mailbox. You can retrieve messages, change greetings, password, etc.

Functions you can access from command mode

The list below shows the digits you can dial while in command mode from a local extension or an outside phone.

Keys	Function performed
00	TalkSwitch tell you the Unit ID of the unit you are connected to.
0 and ID number	Assigns a Unit ID (1 to 4).
30	TalkSwitch tells you which mode it is currently using.
3 and mode number	Switch modes (1 or 2).
4 and auto attendant number	Record auto attendant (1 to 9).
04 and auto attendant number	Erase auto attendant messages (1 to 9).
5 and auto attendant number	Play back auto attendant messages (1 to 9).
6 and message number	Record auto call back announced messages (1 to 4).
65	Record prompted call back announced message.
06 and message number	Erase call back announced messages (1 to 4).
065	Erase prompted call back announced message.
7 and message number	Play back auto call back announced messages (1 to 4).
75	Play back prompted call back announced message.
90	Set to Serial connection mode (default).
91	Set to USB connection mode.
Remote extension number, 8 , and phone number	Set a remote extension phone number.
Remote extension number, 980 , and 0	Turn off <i>Same Line Connect</i> for a remote extension.
Remote extension number, 980 , and 1	Turn on <i>Same Line Connect</i> for a remote extension.

Keys	Function performed
Remote extension number, 9 , and hunt group	Set the hunt group to connect for a remote extension.
Local extension number and 61	Enable <i>Do Not Disturb</i> at the local extension.
Local extension number and 60	Disable <i>Do Not Disturb</i> at the local extension.

APPENDIX B

TALKSWITCH AND TELEPHONE COMPANY CALLING SERVICES

Many of the calling services offered by your local telephone company are compatible with and enhanced by TalkSwitch's capabilities.

Caller ID (Call Display) — Displays the telephone number and name associated with an incoming call.

TalkSwitch is capable of regenerating the caller ID and passing the information to its local extensions when incoming calls are transferred by the auto attendant. TalkSwitch regenerates the caller ID and passes it only to the selected local extension. If a local extension is configured to ring at least twice before the auto attendant answers, TalkSwitch will pass the caller ID information to that extension. If a call is transferred from one extension to another, the caller ID information will also be passed on. TalkSwitch can use the caller ID's time stamp to set its internal clock.

Distinctive Ring — The ability to have more than one telephone number that callers can dial to reach the same telephone line. Distinctive ring is also referred to as: Ident-A-Ring; Ident-A-Call; Ring Master; Teen Service; Double Number; Personal Ringing; Ring Plus; Smart Ring Service; Special Ring; or Customized Ringing.

TalkSwitch can detect distinctive ring telephone numbers and use them to route incoming calls automatically. TalkSwitch gives you the ability to use two distinctive ring telephone numbers on all lines. Distinctive ring numbers on TalkSwitch lines also have access to the auto attendant and all other TalkSwitch call routing features.

The TalkSwitch distinctive ring call routing features can be used in a number of ways:

- Routing fax/modem calls
- Routing voice messaging
- Separating home office calls
- Using auto call back

Each distinctive ring number has a distinctive ring cadence or ring sequence that allows you to identify the number that was dialed.

Voicemail (Call Answer) — A service which gives callers the opportunity to leave a message when your phone line is busy or unanswered. TalkSwitch can work in conjunction with your telephone company's voicemail service.

Call Waiting — If you are already on the phone, the call waiting service will emit a soft beeping sound to let you know you have another call waiting to be answered.

To talk to the other caller when you hear a call waiting beep, press the **Flash** button twice. To return to the other caller, you will need to press the **Flash** button twice again. This service is not recommended, since TalkSwitch cannot answer the second call while the first call is already being routed by the auto attendant. You may want to consider adding the hunt/rollover service to your lines instead.

3-Way Calling/Conference — The 3-way calling/conference service allows you to add a third party to a call already in progress.

The 3-way calling/conference service enhances TalkSwitch capabilities whether you have one or more telephone lines. Using a 3-way calling/conference service gives you access to the TalkSwitch call forwarding, call back and DISAcall bridge features.

TalkSwitch can work with a service to forward calls to remote extensions or other external phone numbers while using same line connect. You can use this service as an option for bridging calls to external numbers when you call into a unit from outside. If you are having difficulties with line volume levels when you forward calls to remote extensions or when you are using DISAcall bridge, forward calls using 3-way calling.

Centrex/Plexar Transfer — The Centrex transfer service is similar in function to the 3-way calling/conference service. The primary difference is that two external parties can still remain in conversation after you hang up. TalkSwitch can disconnect after the transfer if you have this service and

have enabled the *Transfer and clear* option under **Option -> Miscellaneous** in the TalkSwitch configuration software. TalkSwitch can clear the line after forwarding a call to a remote extension, if your telephone line supports this feature. If you enable this feature without having the service, callers are disconnected when a unit attempts to forward their call to the remote extension.

Using one telephone line with Centrex transfer service gives you access to the TalkSwitch call forwarding, call back and DISAcall bridge features.

Hunt or Rollover — When using two or more lines, calls coming in on a busy line can be rolled over to an available line in a hunt pattern that you can set. There are two types of hunting — sequential and circular. Sequential hunting starts at the number dialed and ends in the last number in the assigned group. Circular hunting hunts all the lines in the hunting group, regardless of the starting point. (It will only circle once through the hunt group).

TalkSwitch has the same functionality available on all lines and can work in conjunction with the hunt and rollover services.

Call Hold and Flash

TalkSwitch detects **Flash** from regular phones to put calls on hold. A flash signal is produced by pressing the **Flash** or **Link** button on your telephone (Link is another name for flash). If you do not have these buttons on your telephone, you can produce a Flash signal by quickly depressing and releasing the telephone's hang-up hook.

Single-Flash — is pressing the **Flash** or **Link** button once. You can also depress and release the telephone's hang-up hook once.

Double-Flash — is pressing the **Flash** or **Link** button twice. You can also depress and release the telephone's hang-up hook twice. When using your telephone's hang-up hook to initiate a double-flash, do not try to initiate the second flash too quickly or you may disconnect your caller.

Press and release the hang-up hook once, count to 2, then initiate the second flash .

When would you use double-Flash?

1. If TalkSwitch is connected to any other system/service that uses flash to activate its features (e.g. a Key Telephone System, PBX, or telephone line with the Centrex service from your phone company).
2. If you are using any of your local telephone company services that are activated with flash (e.g. call waiting, 3-way calling/conference, etc.).

What does single and double-Flash do?

TalkSwitch and the above-mentioned systems/services use flash to activate their calling features. TalkSwitch therefore needs a way to tell which (or whose) features you want to access when you press the **Flash** button.

Using a single-**Flash** : places the caller on hold. (The flash signal is not passed to the CO line, solving problems associated with flash causing a hang-up at some CO).

Using a double-**Flash** : the first flash places the caller on hold, and the second flash (if pressed within 2 seconds) tells TalkSwitch to pass a flash to the CO line to activate the intended CO or PBX feature.

APPENDIX C

TALKSWITCH AND POWER INTERRUPTIONS

TalkSwitch settings and configurations

In the event of a power failure, your TalkSwitch settings and configurations will not be lost.

The TalkSwitch configuration is stored in non-volatile memory. Non-volatile memory means that your feature settings (configurations) remain in the TalkSwitch memory despite interruptions in the power supply.

Calling features

In the event of a power failure, TalkSwitch features or capabilities, including VoIP, will be unavailable except for extension port E4. Calls can be made or received via a direct connection to Line 1 on the same unit. Since the connection is direct, calls must be dialed without a hunt group. When the power is restored, all features and capabilities will be re-established.

Mode scheduling and power interruptions

When the power supply to TalkSwitch has been interrupted, TalkSwitch's internal clock will continue to run for up to 12 hours. If you have enabled the automatic mode scheduling feature and it has been longer than 12 hours, TalkSwitch will not function properly until the internal clock is reset via a caller ID call or with the TalkSwitch configuration software. When power is restored to TalkSwitch, it will check the time and determine the mode it should be running according to the configuration and set itself to that mode.

To check the current mode of operation:

1. Dial **#** at a local extension to enter command mode and enter a password if required.
2. Dial **30 #**. TalkSwitch will report the current mode.

APPENDIX D

SAFETY AND REGULATORY INFORMATION

Safety Precautions

Before using TalkSwitch, please review and ensure the following safety instructions are adhered to:

1. Read, follow and retain instructions — All safety and operating instructions should be read, followed and retained for future reference before operating the equipment.
2. Heed warnings — All warnings on the equipment and in the operating instructions should be adhered to.
3. Temperature — Do not operate in environments where the temperature is below 0°C or 32°F. If the product was shipped in cold weather, please let the system warm up at room temperature for at least two hours before plugging it in to a power outlet.
4. Moisture — Do not place TalkSwitch in a high-humidity environment.
5. Heat — Never place TalkSwitch near heat sources such as radiators, floor registers or in direct sunlight. Do not stack units on top of each other.
6. Power supply — The equipment should only be connected to an approved power supply of the type described in the operating instructions or marked on the equipment. Use only the power adapter supplied with TalkSwitch.
7. Damage requiring service — Do not attempt to service the TalkSwitch yourself. Unplug TalkSwitch and refer servicing to a licensed technician when:
 - The plug or power cord has been damaged.
 - TalkSwitch has been exposed to moisture.

Call Redirection & Service Provider Billing Advisory

Use of the Automatic Route Selection, Toll Restriction and Call Detail Recording features does not imply any guarantee whatsoever by regulatory authorities, your telephone service provider(s), TalkSwitch or its distributors and resellers, with regard to the accuracy of these features and that the use of such a features may not be considered by a telephone company in any disputes which may arise regarding the accuracy of any subscriber's telephone account.

Important Notices

CE Compliance

This equipment has been tested and found to comply with the following standards:

EN55022:1998 + amendment A1:2000 + amendment A2:2003, Class B

EN55024:1998 + amendment A1:2001 + amendment A2:2003

EN60950-1:2001, first edition

EN61000-3-2:2000

EN61000-3-3:1995+ amendment A1:2001

CISPR22:1997 + amendment A1:2000 + amendment A2:2002



This equipment is marked with the CE symbol, indicating compliance with CE standards.

FCC and IC Warnings

This equipment complies with ACTA TIA/EIA/IS-968A and Part 68 of the Federal Communications Commission (FCC) rules in the United States. It also complies with regulations RSS210 and CS-03 of Industry Canada and Science Canada. This equipment also complies with Part 15 of the FCC Rules, as well as ICES003 of Industry Canada. Operation is subject to the following two conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation of the device.

FCC Part 15

This equipment has been tested and found to comply with the limits for a "CLASS B" Digital Device pursuant to Part 15 of the FCC rules. These limits are designed to provide a reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with these instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment to an outlet on a circuit different to that which the receiver is connected.
- Consult the dealer or an experienced Radio/TV technician for help.

This CLASS B Digital apparatus meets all requirements of the Canadian interference-causing equipment regulations.

Changes or modifications not expressly approved by TalkSwitch could void the user's authority to operate the equipment.

ACTA TIA/EIA/IS-968A, FCC Part 68 and IC CS-03

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

A label is located on the underside of the base unit containing both the FCC registration number and Ringer Equivalency Number (REN) or the IC registration number and Load Number. You must, upon request, provide this information to your local telephone company.

The REN is used to determine the quantity of devices that may be connected to the telephone line. Excessive RENs on the telephone line may result in

the devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed five (5). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company.

This equipment is compatible with inductively-coupled hearing aids.

RoHS Compliance

This equipment conforms with the Reduction of Hazardous Substances (RoHS) Directive 2002/95/EC. The RoHS directive provides for restriction or elimination of the following substances:

Lead, mercury, cadmium, hexavalent chromium, polybrominated biphenyls (PBB), and polybrominated diphenyl ether (PBDE).

RoHS and related legislations are initiatives to reduce the environmental issues related to the manufacture of electronic equipment.

Service

If you are experiencing difficulty with your Model: CT-TS001.1 telephone equipment, please contact your reseller for repair and/or warranty information. See *Appendix E: TalkSwitch One-Year Limited Warranty and Return Policy* for more information.

If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is corrected.

This equipment cannot be used on public coin telephone services provided by the telephone company. Connection to party line service is subject to state tariffs.

The FCC requires that you connect your cordless telephone to the nationwide telephone network through a modular telephone jack (USOC, RF11C, RJ11W, or RJ14).

Your telephone company may discontinue your service if your equipment causes harm to the telephone network. They will notify you in advance of disconnection, if possible. During notification, you will be informed of your right to file a complaint to the FCC.

Occasionally, your telephone company may make changes in its facilities, equipment, operation, or procedures that could affect the operation of your equipment. If so, you will be given advance notice of the change to give you an opportunity to maintain uninterrupted service

APPENDIX E

TALKSWITCH ONE-YEAR WARRANTY AND RETURN POLICY

TalkSwitch limited warranty

Condition: The original end-user customer acknowledges that he or she has read the limited One-Year Warranty and Warranty Disclaimers and understands the same and agrees to be bound by its terms and conditions.

TalkSwitch, a division of Centrepont Technologies Inc. (“TalkSwitch”) warrants to the original end-user customer (“Customer”) that each new TalkSwitch product shall be substantially free from defects in materials and workmanship under normal use for a period of one (1) year from the date of the original purchase (proof of purchase required). The exclusive remedy and entire liability under this warranty will be for repair or replacement on a like-for-like basis at TalkSwitch’s option.

The above warranty shall not apply to product defects resulting from (a) improper maintenance or installation; (b) misuse, neglect or accident; (c) damage from moisture or corrosive environments; (d) use of the power adapter with supply voltages other than that for which it is specified; (e) high voltage surges; (f) electrostatic discharges; (g) operation outside the product’s specification; or (h) failure to follow product instructions.

The warranty is void where (a) the serial number has been altered, removed or effaced; or (b) the product has been used in an application, country, region, locality, or connected to any network, other than those for which the product was intended to operate.

TalkSwitch shall not have any obligation to repair or replace product until the Customer returns defective product to TalkSwitch. Any replacement product may be either new or like-new, and may contain remanufactured parts, equivalent to new in performance.

EXCEPT FOR THE EXPRESS WARRANTIES SET FORTH ABOVE, AND TO THE EXTENT PERMITTED BY APPLICABLE LAW, THE PRODUCTS AND STORAGE MEDIA ARE SUPPLIED ON AN “AS IS” BASIS AND THERE ARE NO WARRANTIES, REPRESENTATIONS OR CONDITIONS WHETHER EXPRESSED OR IMPLIED, WRITTEN OR ORAL, ARISING BY STATUTE, OPERATION OF LAW, COURSE OF DEALING, USAGE OF TRADE OR OTHERWISE, WITH RESPECT TO THE PRODUCT. TALKSWITCH SPECIFICALLY DISCLAIMS ANY IMPLIED WARRANTIES OR CONDITIONS OF MERCHANTABILITY, SATISFACTORY QUALITY, DURABILITY, FITNESS FOR A PARTICULAR PURPOSE, AND NON-INFRINGEMENT.

IN NO EVENT SHALL TALKSWITCH BE LIABLE FOR ANY SPECIAL, INCIDENTAL, CONSEQUENTIAL, INDIRECT OR SIMILAR DAMAGES, INCLUDING ANY LOST PROFITS, LOST DATA, OR OTHER INCIDENTAL OR CONSEQUENTIAL OR PUNITIVE DAMAGES, WHETHER BASED ON CONTRACT, TORT (INCLUDING WITHOUT LIMITATION, NEGLIGENCE), OR ANY OTHER LEGAL THEORY, WHETHER OR NOT ADVISED OF THE POSSIBILITY OF SUCH DAMAGES, OR WHETHER SUCH DAMAGES ARE FORSEEABLE. IN ANY EVENT, THE TOTAL AGGREGATE LIABILITY OF TALKSWITCH FOR DIRECT DAMAGES SHALL NOT EXCEED THE PURCHASE PRICE OF THE PRODUCT.

This warranty gives specific rights, and Customer may have other rights subject to jurisdiction. To the extent any part of this limited warranty statement is inconsistent with such local law, that part shall be deemed modified to be consistent with such local law.

DOC Compliance

This digital apparatus does not exceed the Class B limits for radio noise emissions for digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

The FCC Statement is located in *Appendix D: Safety Precautions and Regulatory Information*.

Warranty Service

To be eligible to obtain warranty service you must retain proof of purchase and contact your reseller for warranty details. If you fail to provide proof of purchase, your reseller and TalkSwitch, a division of Centrepont Technologies Inc. (“TalkSwitch”) will not be able to provide you with any kind of warranty service.

To obtain warranty service, products must be sent to your reseller or a service facility designated by TalkSwitch. The original end-user purchaser shall pay all shipping charges, duties and taxes for products sent to TalkSwitch for warranty service. TalkSwitch shall pay for return of the product to the end-user purchaser.

If purchased from a reseller:

Contact your reseller and obtain a Warranty Return Authorization (WRA) number if required.

If purchased from TalkSwitch:

Please contact the Technical Support department at TalkSwitch to obtain a WRA number — (866) 393-9960 toll free in continental North America or (613) 725-2466. Please have your model and serial number available when you call.

If the product is being shipped from outside of Canada, the following statement must be included on the package:

Canadian Goods Being Returned to Canada.

Failure to obtain a WRA number and include the necessary information will result in the package being refused.

APPENDIX F

SPECIFICATIONS

System	240vs	244vs	280vs	284vs	480vs	484vs
PSTN analog trunks	2	2	2	2	4	4
Analog extensions	4	4	8	8	8	8
VoIP trunks	0	4	0	4	0	4
Maximum extensions* (analog + IP)	12	12	16	16	16	16
USB interface	Standard USB Type B					
LAN interface	RJ-45 Ethernet (10BaseT)					
Memory	30 mins to 150 mins internal* 1, 3, 4.5 and 9 hour memory cards					
Music input	Phono jack (1/8" mono) or stored .wav file					
PA output	Phono jack (1/8" mono)					
Power source	120 VAC, 60 Hz — In/16 VAC, 1.5 A — Out					
Power-failure line to extension jacks	Extension 4 connects to Line 1.					

* requires purchase of License Code

VoIP Features (models with VoIP trunks only)

- Embedded SIP server and user agents
- Integrated registrar
- Fax tone detection
- RTP
- G.711 μ -law/A-law, G.726 (32 Kbps), G.729a
- G.168 echo cancellation
- VAD, silence suppression
- Adaptive jitter buffer
- Packet loss concealment
- Can use external SIP servers as required

Analog extension interface

Trunk type	Loop start
Connector type	RJ-11, 2-pin
Interface impedance	600 Ω
Loop range	0-600 Ω
Hook Flash supported	Yes
On-hook voltage	35 volts
Off-hook loop voltage source	35 volts
Off-hook loop currant	23 mA to 40 mA
Dial tone level	-2.4 dBm
Ringing voltage	90 volts RMS
Ringing frequency	20 Hz - True sine wave
Total ringing load	5 REN

Analog Trunks

Connector type	RJ-11, 2-pin
Trunk type	Loop start
Interface impedance	Configurable, 600 ohms, 600 complex, 900 ohms, 900 complex
Ringing impedance	3,900 ohms
Ringing sensitively	45 volts RMS @ 20 Hz
REN	0.7B
Longitudinal balance	69 dB
Signal level	-10 dB
Distinctive ring supported	Yes
Caller ID support	FSK, all lines

Dimensions (H x W x L)

Metric	51.6 mm x 203.2 mm x 213.4 mm
Imperial	2.03" x 8" x 8.4"

Weight	240vs	244vs	280vs	284vs	480vs	484vs
Metric (kg)	0.65	0.67	0.72	0.74	0.79	0.81
Imperial (lbs)	1.43	1.46	1.58	1.62	1.74	1.77

Regulatory Approvals



ACTA TIA/EIA/IS-968A and FCC Part 68, Industry Canada CS-03,
FCC Part 15B, Industry Canada ICES-003.

Power adapter complies with UL and CSA
Reduction of Hazardous Substances (RoHS) Directive 2002/95/EC

APPENDIX G

COPYRIGHT AND LICENSING NOTICES

TalkSwitch copyright © 2007 — All Rights reserved.

TalkSwitch is a division of Centrepoint Technologies Inc.

TalkSwitch software is protected under copyright law and international treaties. Unauthorized reproduction or distribution of TalkSwitch software, or any portion of it, in any form, may result in severe civil and criminal penalties, and will be prosecuted to the maximum extent of the law.

TalkSwitch phone systems use the Appello® SIP stack for VoIP communications. The Appello® SIP stack was developed by TalkSwitch — a division of Centrepoint Technologies Inc.

TalkSwitch and Appello are registered trademarks of Centrepoint Technologies Inc.

Elements of TalkSwitch software are protected under additional copyrights listed below, and subject to the terms and disclaimers listed below.

A) Copyright © 1998 by the Massachusetts Institute of Technology.

Permission to use, copy, modify, and distribute this software and its documentation for any purpose and without fee is hereby granted, provided that the above copyright notice appear in all copies and that both that copyright notice and this permission notice appear in supporting documentation, and that the name of MIT not be used in advertising or publicity pertaining to distribution of the software without specific, written prior permission. MIT makes no representations about the suitability of this software for any purpose. It is provided “as is” without express or implied warranty.

B) Copyright © 2000-2003 Intel Corporation. All rights reserved.

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

- Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.
- Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.
- Neither name of Intel Corporation nor the names of its contributors may be used to endorse or promote products derived from this software without specific prior written permission.

THIS SOFTWARE IS PROVIDED BY THE COPYRIGHT HOLDERS AND CONTRIBUTORS "AS IS" AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL INTEL OR CONTRIBUTORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

C) Copyright © 2001-2003 Swedish Institute of Computer Science. All rights reserved.

Redistribution and use in source and binary forms, with or without modification, are permitted provided that the following conditions are met:

- Redistributions of source code must retain the above copyright notice, this list of conditions and the following disclaimer.
- Redistributions in binary form must reproduce the above copyright notice, this list of conditions and the following disclaimer in the documentation and/or other materials provided with the distribution.
- The name of the author may not be used to endorse or promote products derived from this software without specific prior written permission.

THIS SOFTWARE IS PROVIDED BY THE AUTHOR "AS IS" AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL THE AUTHOR BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE.

D) Copyright © 1991-1992, RSA Data Security, Inc. Created 1991.

All rights reserved.

License to copy and use this software is granted provided that it is identified as the "RSA Data Security, Inc. MD5 Message-Digest Algorithm" in all material mentioning or referencing this software or this function. License is also granted to make and use derivative works provided that such works are identified as "derived from the RSA Data Security, Inc. MD5 Message-Digest Algorithm" in all material mentioning or referencing the derived work. RSA Data Security, Inc. makes no representations concerning either the merchantability of this software or the suitability of this software for any particular purpose. It is provided "as is" without express or implied warranty of any kind. These notices must be retained in any copies of any part of this documentation and/or software.

E) Copyright © 2004 Magnus Damm <damm@opensource.se>

All rights reserved.

Permission is hereby granted, free of charge, to any person obtaining a copy of this software and associated documentation files (the 'Software'), to deal in the software without restriction, including without limitation the rights to use, copy, modify, merge, publish, distribute, sublicense, and/or sell copies of the software, and to permit persons to whom the software is furnished to do so, subject to the following conditions:

The above copyright notice and this permission notice shall be included in all copies or substantial portions of the software.

THE SOFTWARE IS PROVIDED "AS IS", WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED, INCLUDING BUT NOT LIMITED TO THE WARRANTIES OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE AND NONINFRINGEMENT. IN NO EVENT SHALL THE AUTHORS OR COPYRIGHT HOLDERS BE LIABLE FOR ANY CLAIM, DAMAGES OR OTHER LIABILITY, WHETHER IN AN ACTION OF CONTRACT, TORT OR OTHERWISE, ARISING FROM, OUT OF OR IN CONNECTION WITH THE SOFTWARE OR THE USE OR OTHER DEALINGS IN THE SOFTWARE.

F) Copyright © 1991, 1999 Free Software Foundation, Inc.

3rd-Party IP Phone configuration library.

Copyright © 2006 Centrepont Technologies, Inc.

This library is free software. You can redistribute it and/or modify it under the terms of the GNU Lesser General Public License as published by the Free Software Foundation, either version 2.1 of the License, or (at your option) any later version. This library is distributed in the hope that it will be useful, but WITHOUT ANY WARRANTY; without even the implied warranty of MERCHANTABILITY or FITNESS FOR A PARTICULAR PURPOSE. See the GNU Lesser General Public License for more details. You should have received a copy of the GNU Lesser General Public License along with this library. If not, write to the Free Software Foundation, Inc.,

59 Temple Place, Suite 330, Boston, MA 02111-1307 USA.

APPENDIX H

HOME/OFFICE WIRING GUIDE

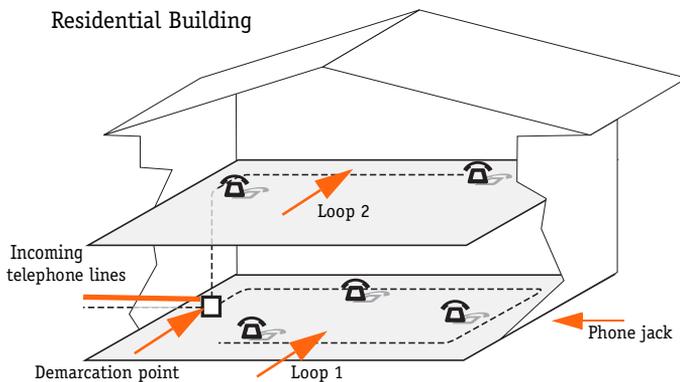
Basic residential phone wiring

Looped wiring

Loop wiring is common in most residential houses. The demarcation point (see next page) breaks the incoming phone lines into 'loops' that can run the length of the entire house, but are often broken into smaller loops that serve different areas of the house.

All houses have at least one phone line, however it is also common to have two or more incoming phone lines. Whether you are wiring single, dual or multiple lines, you will still be using phone cord with four wires. How each phone jack is wired to the loops will determine if the phone jack is on Line 1 or Line 2.

This document describes single- and dual-line wiring configurations. Many of the processes can also be applied to multi-line wiring.



Depending on how the jack is wired, the phone can be plugged into Line 1, Line 2, or both.

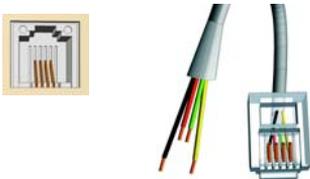
Single-line wiring



For a single incoming phone line, all phones in the building will usually be wired using the RED and GREEN wires of the phone cord. The other two wires, BLACK and YELLOW, are not used.

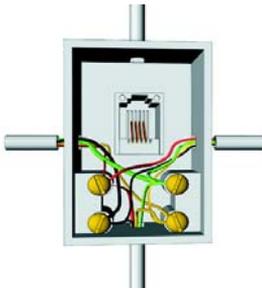
The demarcation point connects the incoming RED and GREEN pair to various loops within the building. It is probable that the building is also wired with a BLACK and YELLOW pair, however these are not connected to any wall jacks.

Dual-line wiring



Two incoming phone lines are denoted in pairs, usually the RED and GREEN are used as Line 1, and the BLACK and YELLOW are Line 2. All four wires are looped throughout the building so that each jack has access to Line 1 and Line 2. How each jack is connected to the RED, GREEN, BLACK, and YELLOW wires determines if it is connected to Line 1, Line 2, or both.

Demarcation Point



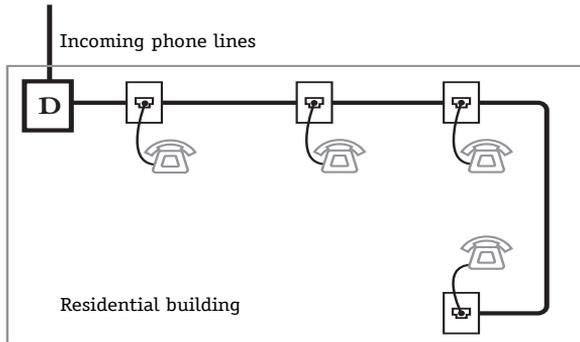
The demarcation point is the point of interconnection between the telephone company facilities and your building.

Location: It is usually a box located inside the building, within 15 feet of the incoming phone wiring. In newer houses, it may be located near the fuse box.

Wiring: The box will contain at least four posts used to branch the incoming wiring to the internal phone wiring. The RED post should have all the RED wires attached to it, the GREEN should have all the GREEN wires attached etc.

This demarcation point shows two lines (four wires) wired as two loops.

Home and home office wiring options prior to adding TalkSwitch



All phones are connected together in a single loop (or in multiple loops)

The phones are all wired to Line 1. If you have two or more incoming lines, the phones can be wired to Line 1, Line 2 or even both lines for dual line phones.

Prior to adding a TalkSwitch, all phones in the home ring in unison when a call comes in. Home phones do not have TalkSwitch functionality and rely on phone company features for any enhancements.

Choose a wiring method (Option 1 or Option 2 below) that best suits your needs.

OPTION 1 — TalkSwitch phones separate from home phones

Advantages

- Easy to set up. May not require any wiring changes when used with one incoming line.
- You can still use some of TalkSwitch's features from the home phones.

Disadvantages

- Your home phones and office phones are not fully integrated together. For example, calls to the office cannot be forwarded to the home phones.
- Home phones don't have access to voicemail and other TalkSwitch features.
- All phones ring at the same time for incoming calls.

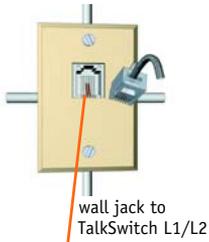
LEGEND:

	Telephone Lines		Phone Cord (RJ-11)		Fax Machine
	Demark or MPoP		Home Phone		TalkSwitch
	Extension Block		TalkSwitch Extension		TalkSwitch Lines
	Single Line Wall Jack (RJ-11)				

Connecting to phone lines

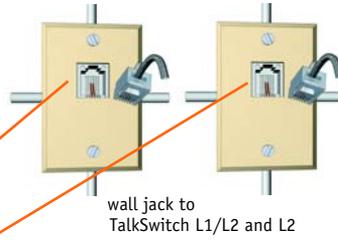
Using one phone line

single-line phone cord — 2 pins



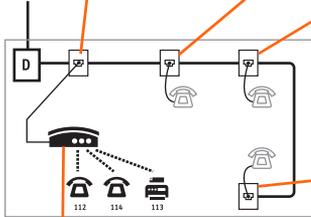
Using two phone lines

single-line phone cords — 2 pins



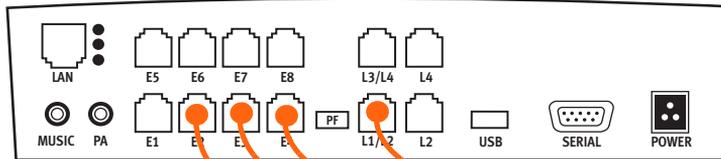
Using a dual-line phone jack

dual-line phone cord — 4 pins



Connecting to TalkSwitch rear panel

If you have two single lines, you will need to connect to jack L1/L2 and jack L2.



To office extensions, phones and/or fax machines using single-line RJ-11 cabling

This shows a single or a dual line from wall jack to TalkSwitch L1/L2

LEGEND:

Telephone Lines	Phone Cord (RJ-11)	Fax Machine
Demark or MPoP	Home Phone	TalkSwitch
Extension Block	TalkSwitch Extension	TalkSwitch Lines
Single Line Wall Jack (RJ-11)		

OPTION 2 — TalkSwitch controls incoming lines and phones

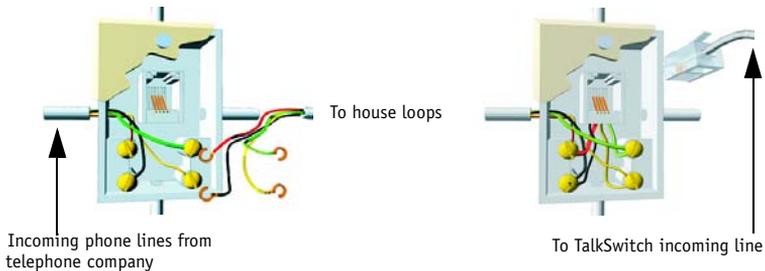
Advantages

- Home phone system and office phone system are integrated together.
- Calls to the home can be forwarded to the office and vice versa.
- The home phones have all the call handling capability as the office phones including voicemail, call cascade, mode scheduling, call forwarding etc.
- Suitable for one or two incoming phones lines.

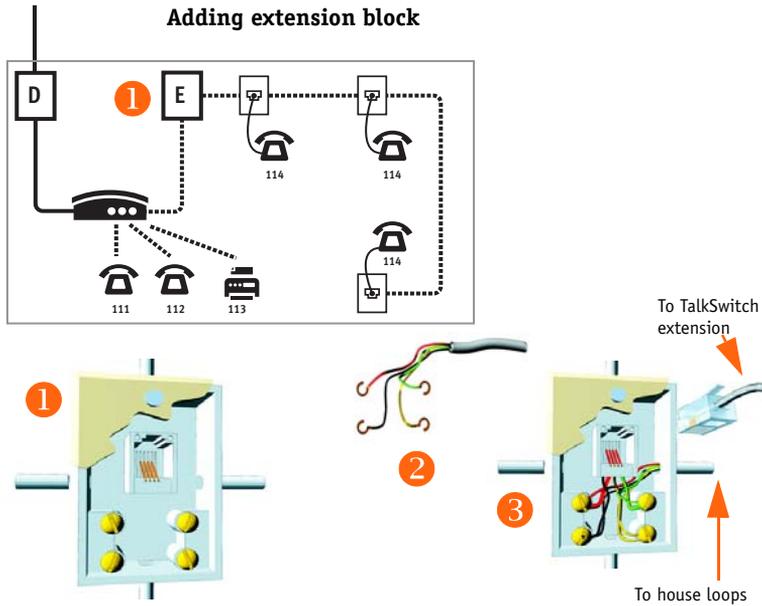
Disadvantages

- Requires a little more wiring, and additional cabling.

Re-wiring demarcation point



1. Remove the wiring to the house loops.
 2. Disconnect any additional house loops that may also be connected. Only one loop wiring is shown for clarity.
 3. Tighten the screws connecting the incoming wires to their respective posts.
- Use an RJ-11 (single line or dual line) to connect to the TalkSwitch. Plug the RJ-11 into jack L1/L2.

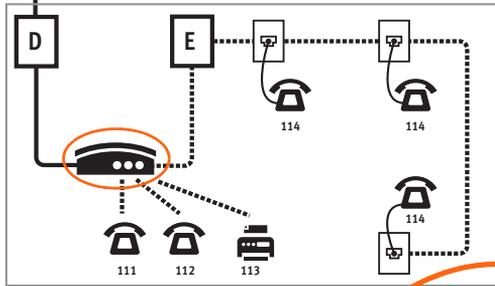


1. Add an extension block next to the previous demarcation point.
2. Take the previous house loop wiring and connect them to the four posts of the extension block.
3. Use an RJ-11 (single line) to connect from the TalkSwitch. Plug the RJ-11 into a TalkSwitch extension, jack E1 through E8.

LEGEND:

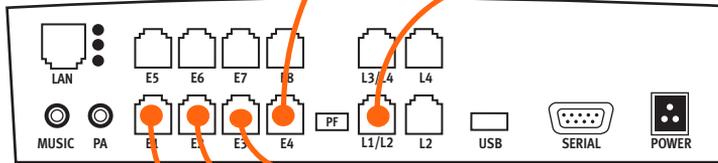
	Telephone Lines		Phone Cord (RJ-11)		Fax Machine
	Demarc or MPoP		Home Phone		TalkSwitch
	Extension Block		TalkSwitch Extension		TalkSwitch Lines
	Single Line Wall Jack (RJ-11)				

Connecting to TalkSwitch



To extension block, use a single-line phone cable from jack E4

Connecting to TalkSwitch rear panel



From demarcation point, use a dual-line phone cable from the demarcation point to jack L1/L2

Ext 113: To fax machine (optional) from jack E3

Ext 112: To phone from jack E2

Ext 111: To phone from jack E1

To office extensions, phones and fax machines

RJ-11 cabling can go up to 100 feet from the TalkSwitch without affecting voice quality. You may extend the lines farther although you may suffer voice quality.

Cabling can be run along baseboards, moldings, threaded between walls, and hidden under carpets. Keep the cords in corners and along edges to help hide it, and avoid high traffic areas.

LEGEND:

	Telephone Lines		Phone Cord (RJ-11)		Fax Machine
	Demark or MPoP		Home Phone		TalkSwitch
	Extension Block		TalkSwitch Extension		TalkSwitch Lines
	Single Line Wall Jack (RJ-11)				

1. Use dual-line phone cable to connect from the demarcation point to TalkSwitch. You will need to purchase the dual-line phone cable (available at most hardware stores) or wire it and crimp it yourself. Do not use the supplied RJ-11 cable as it is only a single-line cable.
2. Use the single-line RJ-11 phone cord supplied with TalkSwitch to connect from the TalkSwitch extension to the extension block.
3. Connect the office phones to the TalkSwitch using RJ-11 cabling supplied with TalkSwitch or the cables supplied with your phone.

GLOSSARY

*** Star/asterisk sign**

The character at the bottom left of your touchtone telephone number pad.

Number/pound sign

The character at the bottom right of your touchtone telephone number pad.

3-way calling/conference call

Connecting three parties together for one telephone conversation.

TalkSwitch offers its own 3-way conference feature the same as the 3-way conference service offered by telephone companies.

Analog (see also digital)

Analog devices (phones, answering machines, etc.) transmit and receive sound in waves. The telephone lines in most homes are analog.

Call back

If call back is activated, it instructs TalkSwitch to call you back at a preset out-of-office telephone number. With call back you can access the TalkSwitch long-distance plan while you are away. TalkSwitch offers automatic call back and prompted call back.

Call bridge (see DISA)

Call forward

Call forward is used to forward calls to one of the remote extensions.

Caller ID

A service provided by telephone companies, transmitting information about calls (caller's name, telephone number, time of call, etc.).

Call waiting

A service provided by telephone companies, allowing you to accept a second

incoming call on a telephone line that is already in use. Call waiting allows you to toggle from one caller to the other.

Centrex/Plexar

Central office exchange service. The telephone company offers PBX features on a 'pay-per-service' basis. This service may not be available from all telephone companies. Centrx/Plexar call switching happens at a local telephone company instead of your company's premises.

CNG

A tone emitted by most fax machines to initiate contact with another fax machine.

CO

Central office of a telephone company.

COM port

Communications port. If you connect your TalkSwitch unit to your PC with a Serial cable, the COM port communicates with your TalkSwitch. It creates an internal virtual pathway to transmit data to and from the Serial port and the software. Most PCs have four COM ports.

Communications software

Software to send and receive data over telephone lines via modems.

Compression

Reduces the data size. It saves data storage space, bandwidth and transmission time.

Conference call

Connecting three or more parties for one phone conversation. They can be internal or external TalkSwitch parties or remote callers.

Configuration

The way the system is set up. The configuration software defines the settings of the TalkSwitch system. These settings determine what each TalkSwitch feature will do and when and how it will do it.

Connector

The end of a cable that plugs into a port or interface to connect one device to another.

Digit

A whole number from 0 to 9. The numbers on your telephone keypad are digits.

Digital (see also analog)

Digital devices (phones, answering machines, etc.) convert analog waves to

digital signals for data storage or transmission over digital lines. The digital signal is converted back to analog waves that you can hear.

DISA (call bridge)

Direct Inward System Access. Allows you to make long-distance calls from anywhere in the world over your PBX using an access number and an authorization code.

Distinctive ring

A service provided by some telephone companies to distinguish two different telephone numbers using the same telephone line (i.e. fax machine).

TalkSwitch can detect one, two and three ring bursts per ringing cycle to provide different handling for incoming calls.

Driver

The software that controls a device (printer, disk drive, modem, etc.). The driver acts as a translator between the device and the programs that use it.

DTMF

Dual-tone Multi-frequency. It is the system used by touchtone telephones. DTMF assigns a specific frequency (tone) to each key so that it can be identified by a microprocessor.

Dual-line telephone

A telephone allowing you to make and receive calls on two separate telephone lines.

Extension

See *Local extension* or *Remote extension*.

FCC

Federal Communications Commission. The FCC is the federal regulatory agency for the United States, responsible for the regulation of interstate and international communications by radio, television, wire, satellite and cable.

Firmware

Firmware is a combination of software and hardware. Microchips that have data or programs recorded on them are called firmware.

Flash button

A button on your telephone. Pressing the **Flash** button has the same effect as pressing the **Link** button or pressing and releasing the hang-up hook on a telephone. All of these actions send a signal to TalkSwitch to put calls on/off hold.

Icon

A small picture representing a program, an action or a feature in the configuration software. When you click on an icon, the program, action or feature is activated.

IETF

Internet Engineering Task Force. Develops and promotes Internet standards like TCP/IP and SIP protocols.

Intercom

A dedicated voice service within a specified user environment. TalkSwitch offers intercom capabilities between its local extensions.

Interface

Connects two separate entities. A user interface is the part of a program that you see on your computer screen and that connects you to the computer.

IP address

Internet Protocol. The standard protocol used by devices to communicate over a computer network. The IP address is a unique number or name that identifies the device.

IP extension

An extension that has its own IP address utilizing the Internet protocol in connection with VoIP.

IP phone

See IP extension.

ISDN

Integrated Services Digital Network.

ISP

Internet Service Provider. Facilitates access to the Internet for end-users. Most ISPs are telephone companies. Their services include domain name registration, web hosting, leased line access, etc.

Keypad

The push buttons on a telephone.

Keypad commands

Combinations of digits used to configure TalkSwitch from a touchtone telephone.

LED

Light Emitting Diode. An device that lights up when electricity is passed through. TalkSwitch has five LEDs on its front panel letting you know which lines are in use and three on the back panel beside the LAN port.

Line 1, 2, 3, or 4

A jack or plug-in on the back panel of TalkSwitch which can accept an RJ-11 plug from a telephone line or PBX system. You can plug in a single-line telephone cable into this jack and it becomes a usable TalkSwitch line.

Link button

The **Link** button has the same effect as the **Flash** button or pressing and releasing the hang-up hook on most telephones. On some telephones, the function of the **Link** button is internal to the telephone and pressing it has no effect on TalkSwitch.

Local extension

A local extension is any telephone or analog device (answering machine, fax, etc.) plugged into one of the extension jacks on the back panel of your TalkSwitch.

Modem

Acronym for *MOD*ulator-*DEM*odulator. A modem is a device that converts digital information from a computer into analog signals for transmission over telephone lines and vice-versa.

Modes

TalkSwitch supports different time and day modes of operation, providing different call management depending on the current mode of operation.

Monitor (TalkSwitch monitor)

A feature included with the configuration software. The monitor provides a graphical representation of telephone traffic as it flows through the system.

Network

A group of two or more computer systems linked together.

PBX

Private Branch eXchange. A PBX is a private telephone network used within an enterprise. The PBX controls access to telephone lines and services. It allows many users to share a smaller number of telephone lines using extensions and have access to additional call handling benefits (call hold, intercom calling, etc.). TalkSwitch is a small PBX.

Phone load

see *Ringling equivalency*.

POP3

The post office protocol (POP) is a standard Internet protocol for retrieving e-mail from a remote server. Most subscribers to an Internet service provider have e-mail accounts using POP3 (3 is the version number).

Prompt

Recorded instructions played to a caller by TalkSwitch. Prompts are also used to provide instructions when you configure TalkSwitch using a telephone keypad.

PSTN

Public Switched Telephone Network. Your incoming phone line is on a PSTN.

Registered jacks

Telephone and data plugs registered with the FCC.

Remote extension

A call forwarding number that can be a telephone number in any location, set up as a virtual extension of the TalkSwitch unit. The remote extension is assigned an extension number and has a voice mailbox.

REN

Ringer Equivalency Number. A number determined in accordance with the US code of federal regulations. It represents the ringer loading effect on a line.

RJ-11

Registered jack 11 is used to connect single dual-wire telephone lines. It is used with telephones, cordless phones, fax machines and modems.

RJ-14

Registered jack 14 usually connects dual-line telephone devices and supports four conductors.

RJ-45

Registered jack 45 supports eight conductors and is generally used for networking applications.

RTP

Real-time Transport Protocol. It defines a format for delivering audio and video files over the Internet.

Routing

Determining the path that a message or call has to take over circuits. TalkSwitch routes incoming calls according to the configuration.

Serial communication

A type of electronic communication requiring data bits to be sent one after the other, unlike parallel communication. Modems rely on serial communication to send data over telephone lines.

Shortcut key

A key combination that causes a specific command to be executed. Shortcut keys combine the Ctrl or Alt keys with other keys. (e.g. Ctrl and C is the shortcut key to copy selected items).

SIP

Session Initiation Protocol. A signaling protocol, developed by the Internet engineering task force (IETF).

SIP server

A server using the SIP protocol to manage calls between SIP devices.

Serial port

Serial ports are physical plug-ins. They allow two devices to exchange data.

SMTP

Simple Mail Transfer Protocol. The standard for e-mail transmission across the Internet. SMTP is a text-based protocol.

Terminal adapter

A device that allows analog devices to be used on digital lines (ISDN). Terminal adapters are similar to modems.

TX/RX

Transmit and receive data.

UDP

User Datagram Protocol. An Internet protocol which allows applications on a computer network to exchange short messages (datagrams).

UPnP

Universal Plug and Play. Allows networked devices to connect seamlessly in home and corporate environments using Internet-based communication standards.

USB

Universal Serial Bus. USB connection supports more bandwidth than a Serial or parallel connection.

Virtual

The term 'virtual' refers to something that is not physically plugged in.

Voicemail (call answer)

A service provided by TalkSwitch or telephone companies giving callers the opportunity to leave a message when your phone is busy or unanswered.

VoIP

Voice over IP. Sending voice over an Internet connection.

VPN

Virtual Private Network. A communication network used within a company or organization to communicate over a public network. VPN message traffic is carried on Internet infrastructure using standard protocols.

INDEX

A

- Additional IP extensions 121
- Administration
 - system password 37
- Announcements
 - configuring 70, 81
 - recording 149
- Audio controls 111
 - extensions 111
 - lines 111
- Auto attendant 83
 - adding and configuring 84
 - additional features 89
 - automatic fax detection 89
 - erasing messages 86
 - loading messages 86
 - receiving calls with... 125
 - receiving calls without... 126
 - recording messages 86
 - routing options 87
 - troubleshooting 188
 - working with messages 85
- Automatic route selection 45
 - permissions 108

C

- Call back/call bridge 102
 - auto call back 103
 - prompted call back 105
- Call bridge 107
- Call cascade 93
 - examples 100
 - extension ring groups 98
 - local extensions 93

- remote extensions 96
- Call conference 133
 - SIP extension 139
- Call detail record (CDR) 155
 - enabling 155
 - retrieving data 156
- Call forward 143
 - screening options 144
 - to a remote extension 63
 - to a ring group 66
 - to remote extension (music on hold) 150
- Call handling
 - auto attendant 84
 - local extension 127
 - local extensions 93
 - modes 81
 - remote extensions 96
 - telephone lines 90
 - VoIP numbers 176
- Call queue
 - caller in queue reminder 114
 - cycling through queued calls 197
 - queue at ring group 88
 - queue call (call cascade) 95
 - queuing and retrieving 131
 - to a single extension 131
 - to an extension ring group 132
- Call routing
 - auto attendant routing
 - options 87
 - call cascade 93
 - troubleshooting 190
- Call transfer

- configuring transfer options 112
- from a remote extension 64
- from an analog extension 128
- same line connect 63
- screened transfer 129
- settings (music on hold) 70
- to a remote extension 63
- to a ring group 66
- troubleshooting 190
- unscreened transfer 128

Caller ID

- local extensions 58

Carrier codes 48

Configuration software 1

- installing 4
- IP configuration 39
- pull down menus 32
- screen navigation 35
- system requirements 1
- troubleshooting 187
- upgrading 5

Connecting devices 15

- audio for music on hold 21
- fax machine 19
- incoming telephone lines 15
- local extension phones 16
- modem 18
- output to public address 21

Connecting incoming lines 15

D

Direct line access

- configuring 60

Distinctive ring

- fax machine 19

E

Extension ring groups

- call cascade 98
- call handling option 88
- configuring 65
- different uses 65
- queuing callers to... 132

Extensions

- numbering conventions 23

F

Fax machine

- automatic detection 19
- dedicated fax line 19

- distinctive ring 19
- troubleshooting 190

H

Hardware unit

- back panel connectors 2
- connecting fax machines 16
- connecting telephones 16
- connecting to a PC 5
- front panel lights 1
- setting or changing unit id 22

Help and troubleshooting 187–194

- auto attendant 188
- call routing/transfer/
intercom 190
- configuration 187
- contacting TalkSwitch 194
- fax machines 190
- local area network 192
- local extensions 191
- music on hold 189

Holiday mode 82

Hunt group access

- local extensions 61

I

Incoming lines

- audio controls 111
- call detail record 158
- configuration 42
- connecting 15
- connecting devices/wiring 15
- dedicated fax line 19, 52
- diagnostics 28
- front panel lights 2
- keeping track of... 24
- line hunt groups 44
- rear panel jacks 4
- troubleshooting 116
- using phones connected in
parallel 141

Intercom

- troubleshooting 190

IP configuration 39

L

Licensed add-ons

- additional IP extensions 121
- additional voicemail
capacity 121

- Attendant Console 121
 - enabling demos... 122
 - enabling... 120
- Line hunt groups 44
- Local area network 163
 - connecting 10
 - IP configuration 39
 - LAN connector 4
 - networking units 21
 - optimizing for VoIP 162
 - optimizing the system 25
 - router/NAT/firewall 162
 - special considerations 35
 - troubleshooting 192
 - unit IDs 23
- Local extensions
 - activating/configuring 56
 - adjusting ringing sequence 92
 - call cascade 93
 - configuring additional IP extensions 58
 - configuring for analog phone 56
 - configuring for IP phone 57
 - connecting telephones 16
 - keeping track of... 24
 - troubleshooting 190, 191

M

- Memory capacity
 - increasing... 26
 - memory cards 26
- Miscellaneous options 114
- Modem
 - connecting 18
- Modes 81
 - holidays 82
- Music on hold 66
 - adjusting volume 69
 - internal audio file 68
 - troubleshooting 189

N

- Notification options
 - configuring 73

R

- Remote extensions 61
 - call cascade 96
 - configuring 61
 - forwarding calls to... 63

- transferring calls from... 64

T

- Telephone company services 43
- Telephone lines
 - configuring for modes 90
 - phone numbers 43
- Telephones
 - selecting phone type 59
 - single line phone 16
 - two line phone 17
- Toll restriction 45
- Transfer options 112
 - transfer from a home phone 112

U

- Uniform call distribution (UCD) 131
- Unit ID 195

V

- Voicemail 70
 - additional voicemail capacity 26, 121
 - dialed notification 72
 - email notification 75
 - global settings 77
 - mailbox options 70
 - management 79
 - message waiting light 74
- VoIP 161–185
 - admin form 186
 - broadband connection 162
 - configuration screen 168
 - FAQ 180–185
 - local area network 163
 - local IP address 166, 168
 - optimizing your network 162
 - public IP address 169
 - router/NAT/firewall 162
 - service provider profile 175
 - setting up a network 165
 - SIP server 166
 - TalkSwitch profile 170
- VoIP numbers 92, 176