USER GUIDE
RELEASE 3.24

CT.TS005.001101.UK

For use in
United Kingdom and Ireland

talkswitch®

Power

Line 1 Line 2 Data Line 3 Line 4

talkswitch

A N S W E R S   W I T H   I N T E L L I G E N C E®

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PREFACE

Thank you for purchasing the TalkSwitch Phone System. This user guide provides instructions for installing, configuring and using your TalkSwitch. Please read this section before you start installing the product.

Important Safety Information

Power & Lightning Surges

Surge protection devices are recommended in areas where lightning or power surges are likely to occur. Avoid using corded telephones during a lightning storm. There may be a risk of electric shock. Do not install a TalkSwitch system during an electrical storm. If your TalkSwitch was supplied with telephone line lightning surge protectors to met local regulatory requirements, please install the devices in accordance with the instructions provided.

Power Adaptor & Power Cords

Use the power adaptor and power cords provided with the TalkSwitch. Other power adaptors may damage the unit. Check that the power adaptor voltage rating matches the local supply. If they do not match, please contact your TalkSwitch reseller for advice.

Temperature & Environment

Operate TalkSwitch in a location where the temperature is between 0°C and 40°C.

- Do not expose it to a heat source or direct sunlight and avoid damp, dusty locations.
- Do not expose it to vibration, hazardous chemicals or explosives.
- Do not locate the TalkSwitch unit where it is subjected to electrical noise generating devices such as motors, fluorescent lamps or radio transmitting devices.
- Do not stack TalkSwitch units directly on top of each other.

- Allow the TalkSwitch to reach room temperature before installation after shipping or storage to avoid problems caused by condensation.

Maintenance

Do not attempt to repair a TalkSwitch unit yourself.

Advisory Information

Power Failure

When a power failure occurs, Line 1 connects to Extension 4 on each unit to enable incoming and outgoing calls over PSTN telephone lines. Consider using an Uninterrupted Power Supply (UPS) and surge protection for applications requiring uninterrupted operation or in areas that are subject to frequent power outages.

Emergency Service Access

Ensure that you do not block extension access to Emergency Services when you set up the Automatic Route Section and Toll Restriction features.

PBX Fraud and Misuse

We recommend that you use password security for the administrator. Change the password frequently to prevent unauthorized users from making calls through your PBX.

TalkSwitch Usage

The model of TalkSwitch you have purchased is designed and supplied to meet the technical standards for use in specific countries and for connection to the major telephone service providers in those countries. It is subject to regulatory certification and compliances as detailed in the appendices. Export and use outside the intended countries is not recommended or supported by TalkSwitch.

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.

Waste Disposal

Please dispose of the product in compliance with local regulations for electrical and electronic waste material.

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CHAPTER 1: TALKSWITCH INSTALLATION

CHAPTER 2: TALKSWITCH CONFIGURATION

CHAPTER 1

TALKSWITCH INSTALLATION

1.1 INSTALLATION

Open the system package, which will contain:

- The TalkSwitch Unit

- AC Power Adaptor
  110V socket mount adaptor or 230V with power cord
  (subject to market model)

- USB cable (supplied only with models
  without a LAN port)

- RJ-45 LAN cable (supplied with all models
  with a LAN port)

- Telephone cables (one per telephone line
  port)

- CD containing configuration program
  installation wizard and user documentation

- Extension Reference Cards

- TalkSwitch Start Guide

- Warranty Card

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

> **VoIP** **Note:** The VoIP symbol indicates sections that only apply to VoIP enabled TalkSwitch units. We recommend that VoIP users read Chapter 5: VoIP Information.

1.2 FRONT PANEL LIGHTS

The front panel consists of a power button ("Power") and five LED (Light Emitting Diode) lights indicating the usage of the lines with different states of illumination.

Light:	State	Description
Line 1	On Solid	Line 1 is currently in use.
	Pulsing Slowly	Line 1 caller is on hold.
	Flickering	Line 1 is ringing.
	Quick Pulse	Line 1 is engaged by a device that is sharing the line with TalkSwitch.
Line 2	On Solid	Line 2 is currently in use.
	Pulsing Slowly	Line 2 caller is on hold.
	Flickering	Line 2 is ringing.
	Quick Pulse	Line 2 is engaged by a device that is sharing the line with TalkSwitch.
Power/Data	On Solid	TalkSwitch is powered on.
	Flickering	The PC connected (via Serial or USB) to TalkSwitch is either sending or retrieving information from TalkSwitch.
	Pulsing Slowly	Global Message Waiting Indicator (optional).

Note: Lights for line 3 and line 4 apply to Talkswitch 48 models. For more details on line LED light error codes, see Appendix A: Help and Troubleshooting.

Line 3	On Solid	Line 3 is currently in use.
	Pulsing Slowly	Line 3 caller is on hold.
	Flickering	Line 3 is ringing.
	Quick Pulse	Line 3 is engaged by a device that is sharing the line with TalkSwitch.
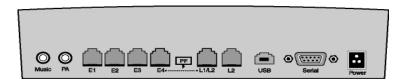
Light:	State	Description
Line 4	On Solid	Line 4 is currently in use.
	Pulsing Slowly	Line 4 caller is on hold.
	Flickering	Line 4 is ringing.
	Quick Pulse	Line 4 is engaged by a device that is sharing the line with TalkSwitch.

1.3 BACK PANEL
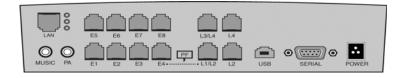
Before you connect all your phones and lines to TalkSwitch, you may want to configure the unit. This minimizes the disruption time of your telephone lines during the system setup. See Chapter 2: Configuration.

TalkSwitch configurations are subject to market availability.
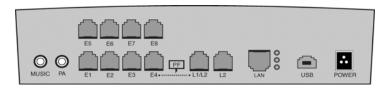
1.3.1 TalkSwitch 24 Models

1.3.2 TalkSwitch 48 Models

1.3.3 TalkSwitch 28 Models

Ports	Purpose
L1-L2 and L3-L4 (subject to model)	Telephone cables for connection to telephone company analogue (PSTN) telephone line ports or demarcation point; or telephone line lightening protection if required (see notes 1 & 2)
E1-E4 and E5-E8 (subject to model)	Local extension telephone sets, fax machines or data modems
USB	USB cable to PC for system configuration
SERIAL (subject to model)	Serial cable to PC for system configuration
Music	1/8" socket for external Music on Hold input
PA	1/8" socket for output to Public Address system
LAN (subject to model)	RJ-45 10BaseT Ethernet for connection to LAN; used for connection to other co-located TalkSwitch units, VoIP networking and configuraion from local PC or remotely over Internet
POWER	Plug in supplied AC Power Adapter; (Rating 16VAC 1.5 A) ONLY use the power adaptor supplied with the system
Memory Slot (on side of system)	For Voicemail (incl. Auto Attendant, MoH) Expansion. Works only with TalkSwitch Voicemail Expansion cards

Note: The 'PF' box in between E4 and L1/L2 represents power failure support. In the event of a power failure or loss of power to TalkSwitch, Extension 114 is able to receive and make calls on Line 1.

In cases where a telephone company's ISDN BRA service is installed using an NTE with two analogue line "ab" ports, the TalkSwitch telephone lines can be connected to the NTE's ab ports (Please check with your local telephone company).

The use of 4-wire telephone cables to combine L1/L2 or L3/L4 wiring requires a special installation procedure and may not be compatible with some standard telephone sockets. It is not recommended for general use.

If your TalkSwitch was supplied with telephone lightning protectors to comply with local regulatory requirements, they must be installed on each telephone line as detailed in the instructions provided with the lightning protectors.

1.4 PLUGGING INTO THE BACK PANEL

1.4.1 Connecting telephone lines

You can connect the TalkSwitch to the telephone company's line sockets using the 2-wire telephone cables provided to ports L1-L2 or L1-L4 (subject to model). Take note of which telephone line is connected to each Line port — this information will be used in the configuration section.

In cases where the telephone company lines are terminated on a Connection Box with IDC or screw terminals, it is recommended that "RJ-11 Wiring-Tails" are used. These are single or multi-pair cables with 0.5mm solid core wire one end terminated in an RJ-11 connector(s). These RJ-11 wiring tails may be available where you purchased your system or provided by your installer.

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

Warning: The TalkSwitch line ports are sensitive to high-voltage spikes from lightning. If you live in an area where electrical storms occur regularly, we recommend that you protect TalkSwitch by plugging the telephone cords coming from the TalkSwitch line ports to a surge protection device connected to the incoming telephone lines.

1.4.2 Connecting extension telephones and other devices

You can connect any regular analogue extension telephone set, cordless (e.g. DECT) telephone, or fax machine to the extension ports – E1-E4 and E1-E8 (subject to model).

In cases where extensions will be wired through in-building wiring and a Connection Box with IDC or screw terminals, it is recommended that "RJ-11 Wiring-Tails" are used. These are single or multi-pair cables with 0.5mm solid core wire one end terminated in an RJ-11 connector(s). These may be available from your TalkSwitch Reseller or provided by your installer.

To connect telephones with BT style plugs it is recommended that extensions are wired through a Master Socket or an in-line adaptors with a capacitor. This also ensures compatibility with some telephones that require 3-wire connections for the telephone to ring.

Attach a single-line corded or cordless telephone, fax machine or answering machine
Connect your single-line analog telephone or fax machine to one of the TalkSwitch extension ports E1-E4 (all TalkSwitch models);
E5-E8 (TalkSwitch 48 models only)) just as you would be plugging them into a standard telephone wall port.

Attach an Internal or External Modem

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

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. See section 2.3.3.9 (Local Extensions -> Direct Line Access) for more details on configuring Direct Line Access.

1.4.3 Connecting devices to the Music and PA ports

The Music port is designed to support any audio source (CD player, radio, tape player, sound card etc.) for playing music or messages to callers while on hold. Connect the audio source via its headphone output to the Music port. The Music port requires a 1/8" (3.5mm) mono phono connector. If you have a 48 model with more than one TalkSwitch connected to a LAN, you need to provide audio to the Music ports on each TalkSwitch.

The PA port 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 port requires a 1/8" (3.5mm) mono phono connector. If you have a 48 model with more than one TalkSwitch connected to a LAN, you need to provide a connection from each TalkSwitch to the PA system.

1.4.4 Connecting TalkSwitch to a LAN and/or PC

There are currently four ways to connect to TalkSwitch for PC configuration — over the LAN (Ethernet port), USB, Serial, or Internet.

LAN:	Use the provided Ethernet cable to connect TalkSwitch to the LAN via your switch (TalkSwitch models with LAN ports only).
USB:	Use a USB cable to connect TalkSwitch to an available USB port on your PC or on the USB hub.
Serial:	Use a serial cable (RS232) to connect TalkSwitch to an available Serial COM port on your PC.
Internet:	The TalkSwitch models with LAN ports support remote configuration from a PC via IP from a local or remote location.

1.4.4.1   Connecting TalkSwitch to a LAN for local configuration

If you want to connect multiple TalkSwitch units to a LAN, see section 1.5.

To connect TalkSwitch to a LAN for configuration purposes, use the supplied Ethernet cable (Category 5 cable with RJ45 connectors on either end, provided with 48 models). Connect TalkSwitch to the switch or hub. The top LED lights up to indicate a connection is established.

1.4.4.2   Connecting TalkSwitch to a PC using a USB cable

If you have an available USB port and a USB cable, connect TalkSwitch to the PC using the USB cable (provided with 24 models).

Make sure no other communications programs* are running the same time you are using the TalkSwitch configuration software.* They may include Palm Pilot, Hot Sync, TalkWorks or digital camera software. These programs tend to occupy COM ports thus making them unavailable for other programs.

1.4.4.3   Connecting TalkSwitch to a PC using a Serial cable

If you wish to use an available Serial port, connect TalkSwitch to your PC with a RS-232 Serial cable (not included). By default, TalkSwitch is shipped with the Serial and LAN ports enabled and the USB port disabled.

When you run the TalkSwitch software, select 'Serial' as the connection type then select the Serial Port associated to this physical Serial port on your PC.

Make sure you do not have any other communications programs running at the same time you want to use the TalkSwitch configuration software. These may include Palm Pilot, Hot Sync, TalkWorks, digital camera software. These

programs tend to 'hold' onto COM ports, making them unavailable for any other programs.

If you are having problems communicating with TalkSwitch, please check the Chapter 6: Help and Troubleshooting.

> **Note:** Users of a TalkSwitch without a LAN port may skip the rest of this chapter and continue with Chapter 2: TalkSwitch Configuration.

1.4.4.4   Connecting to TalkSwitch over IP

Ensure TalkSwitch is connected to a LAN with the supplied Category 5 cable. The top LED should light up to indicate a connection has been established with a switch or hub. To support remote configuration over IP, map port 9393 from your firewall to TalkSwitch. Please refer to the manual for your router/firewall to activate port forwarding.

When you open the TalkSwitch software, you are prompted to select the connection type. Select 'Internet' and enter the public IP address or the FQDN of the TalkSwitch location you wish to configure. You can click 'Address Book' to maintain a list of internet addresses.

> **Note:** Any time TalkSwitch 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 1 hour, TalkSwitch unlocks itself to allow configuration changes.

1.5 USING 2 OR MORE TALKSWITCH UNITS ON A LAN

If you have one TalkSwitch unit, ignore this section and proceed to Chapter 2: TalkSwitch Configuration.

1.5.1 Connecting 2 or more TalkSwitch units to a LAN

TalkSwitch units with a LAN port can be networked together on the same LAN. Ensure you have the appropriate firmware on all units before adding them to the LAN. Check **http://global.talkswitch.com/** for the latest updates.

We recommend integrating your phone system and your existing LAN with an Ethernet switch. A switch provides direct communication between TalkSwitch

units, thus keeping the TalkSwitch voice-over-LAN data isolated from other data on the network.

Connect up to four TalkSwitch units (2 are shown in this illustration) to the LAN switch or switched hub.

> **Note:** TalkSwitch unit enclosures are not designed for stacking. We recommend wall-mounting units in a horizontal row to maximize airflow and keep the units from overheating.

1.5.2 Setting the Unit ID for the first time

When TalkSwitch units are shipped from the manufacturer, they are programmed 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. TalkSwitch indicates this by flashing all the line lights on the front panel of the units that have the conflict.

To resolve the conflict, assign a different Unit ID each of the units. Pick up a telephone handset connected to one of the extension ports on the TalkSwitch unit you want to assign a different Unit ID to. You immediately hear a system prompt indicating that there is a conflict and a new Unit ID needs to be chosen. Select an available Unit ID between 2 and 4.

The system indicates that the update was successful and the front panel lights stop flashing after several seconds. When none of the front panel 'Line' lights are flashing, all the units are ready for network use.

> **Note:** 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 or VoIP once the system has been configured.

1.5.3 Changing the Unit ID

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

Press **#** from a Local Extension to enter command mode.

Enter a password if necessary and dial 0 0 **#** .

The system responds with the Unit ID of that particular TalkSwitch. In command mode, use any of the following commands:

Unit ID	Touch Tone Command
1	01 **#**
2	02 **#**
3	03 **#**
4	04 **#**

1.5.4 How unit IDs affect system extension numbers

When more than one TalkSwitch is connected to a LAN in networked mode, the extensions and voicemail have different numbers based on the Unit ID assigned to the TalkSwitch unit they belong to.

The extensions and voice mailboxes affected by the new numbering system are listed in the table below.

	Unit ID 1	Unit ID 2	Unit ID 3	Unit ID 4
Local Extensions	111-118	121-128	131-138	141-148
Remote Extensions	211-218	221-228	231-238	241-248
Local Mailboxes	111-118	121-128	131-138	141-148
Remote Mailboxes	211-218	221-228	231-238	241-248
General Mailboxes	410-419	420-429	430-439	440-449

None of the Extension Ring Groups are affected They are global to the entire system. It doesn't matter if there are one or four TalkSwitch units on a LAN,

there will always be 10 Extension Ring Groups with the extension numbers 300-309.

1.5.5 Keep track of the lines and extensions

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

If the TalkSwitch units are located in a room away from the extensions and you need to identify the units, there is a utility in the configuration software that allows you to identify each unit by flashing the lights on the front panel. Open the TalkSwitch Configuration software. To check details of the TalkSwitch units on the LAN, click the 'View System Information' link on the initial window below the TalkSwitch image. Select the Unit ID to verify. 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.

1.5.6 Optimizing the system for networked use

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

Configuration Settings

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

Outgoing Line Hunt Groups

By default, when two or more units are on a LAN, the system tries placing calls out on the same TalkSwitch 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 9 Auto Attendants that are shared by all units on the LAN. When an Auto Attendant is recorded on any unit, it is then automatically copied to all other units on the LAN. This design minimizes LAN traffic and also provides functionality back-up in case a unit or units have 'disappeared' off the network (adapter unplugged, LAN connection

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, 221-228 and mailboxes 420-429 are stored on the TalkSwitch with Unit ID 2. If a unit is completely filled with voicemail messages, new messages will not be stored on other units. The system will simply not accept anymore messages for mailboxes belonging to that unit. If you need more memory, TalkSwitch Memory Cards can be purchased from your reseller. You can also use the Voicemail Memory Manager to view the memory usage of each TalkSwitch unit and each mailbox.

C H A P T E R 2

TALKSWITCH CONFIGURATION

2.1 INSTALL THE TALKSWITCH CONFIGURATION SOFTWARE

Insert the TalkSwitch CD into your CD drive. The Install program starts automatically.

If you prefer to use Windows Explorer:

Double-click the My Computer icon.

Double-click the CD-ROM/DVD drive.

Double-click the Startscreen.exe file.

Follow the instructions on the screen.

> **Important:** The latest version of the TalkSwitch software can also be downloaded from the TalkSwitch website **http://global.talkswitch.com**. Select the country or region.

Configuration software system requirements:

- PC running Windows XP/2000
- 120 MB free hard disk space
- 128 MB RAM
- An available USB or Serial port (for 24 models)
- An available Ethernet port (for 48 models)
- 800 x 600 minimum video resolution

If you are having problems retrieving the settings from TalkSwitch, please check Appendix A: Help and Troubleshooting.

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

2.2 RUNNING THE TALKSWITCH SOFTWARE

After the installation is complete, double-click the TalkSwitch icon. If TalkSwitch is currently connected to the same network as your PC, the configuration software detects and retrieves the settings from TalkSwitch automatically. If TalkSwitch is not yet connected, or is connected via the USB or Serial port, the following screen displays with different configuration options.

TalkSwitch without a LAN port   TalkSwitch with a LAN port

Connection Type:

Serial   Select when a serial cable is used to connect the system to a PC. Serial is only supported with models equipped with Serial port (Serial cable must be purchased separately)

Ethernet   Select when system is connected to PC over a LAN. Ethernet is only supported with models equipped with a LAN port.

USB   Select when a USB cable is used to connect the system to a PC. (USB cable is supplied with units not equipped with a LAN port)

Internet   Select when system is connected to over the Internet and the required router ports has been opened to allow remote access to the system Internet (supported by models equipped with a LAN port)

File   Select to edit a configuration file saved on the PC.

Button — Restore Firmware

This button is only available/active if a firmware update has failed. Click it to start the TalkSwitch Firmware Update procedure and follow the instructions. For information on Firmware updates, refer to section 3.8 — Upgrading the TalkSwitch Software and Firmware.

2.3 SYSTEM CONFIGURATION

2.3.1 The configuration screen
The configuration screen consists of the following parts:

1. Menu Items

2. Configuration Navigation (expandable): Controls the display in the configuration window

3. Configuration Window: Displays configuration information and TalkSwitch image.

4. View System Information:
    Click this link to view each TalkSwitch unit's MAC address, IP address, hardware version and firmware version.

5. Region: Indicates the region where TalkSwitch is intended to operate

Note: Context sensitive help information is available for each option in the configuration software. Rest your mouse cursor over any control for a second or two and the information related to that control pops up.

2.3.1.1   File Menu

Open...	Opens existing configuration files from the PC. Default configuration files are included with the software. TalkSwitch backs up to a file called "LastSavedConfig.dat" every time you save settings to TalkSwitch.
Save to TalkSwitch	Saves the current configuration to TalkSwitch.
Save to File...	Saves the current configuration to a file.
Exit	Closes the Configuration.
Retrieve Settings	Retrieves settings from TalkSwitch connected to the PC.
Update Firmware....	This action updates the firmware. For more details on upgrading the TalkSwitch Firmware, see section 3.8.

2.3.1.2   View Menu

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

2.3.1.3   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 status of all Voicemail, 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 ▶	Real-time to Serial Port: TalkSwitch can output call detail records to a PC connected to a Serial port.
	Store to file on TalkSwitch: TalkSwitch can store call detail records to a file stored on TalkSwitch. This file can be viewed/retrieved from a web browser.
Defaults	Select this option to reset the current page or the entire configuration back to defaults.
Reboot TalkSwitch	Prompts you to save the configuration. The reboot takes 25 seconds.

2.3.1.4   Help Menu

To access TalkSwitch documents in PDF format (user guides, quick guides, references etc.), go to:
Start Menu > Programs > TalkSwitch > Documentation.

About TalkSwitch Configuration Utility...	Displays the TalkSwitch software version number, TalkSwitch firmware version number and Copyright information.
Support on the Web...	Launches your browser and links you to our support site.

2.3.2 Configuration Navigation

The Configuration Navigation organizes all the configuration topics within folders. The following is a brief description of each configuration folder:

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.
System Information	This folder allows you to configure the system setup. (i.e. activates lines, extensions, VoIP, music-on-hold).
Voicemail	This folder allows you to configure each individual voice mailbox and some global settings for the voicemail system.
Call Handling	This folder allows you to configure how to handle incoming calls. It allows you to setup your Auto Attendants and individual call handling for each line and distinctive ring number for both operating modes.

Call Back/ DISA	Configure TalkSwitch's Call Back and DISA features in this folder. Both of these features reduce your company's long distance expenses.
Options	Configure advanced settings In this folder (audio controls, troubleshooting and various operating options).

Note: Specific details describing the configuration for the use of features related to the TalkSwitch in a networked configuration, are preceded by the icon below. If you have a single Talkswitch, you can skip these sections in the guide.

2.3.2.1   Configuration considerations connecting multiple units to a LAN

When two or more units are connected to a LAN and are set up for network use, you can configure all the connected units via the Serial or the USB port to one of the TalkSwitch units, or to a PC connected to the same LAN as the TalkSwitch units.

When two or more TalkSwitch units are connected to the same LAN, they are designed to act like a single phone system. For example, if you have two TalkSwitch units with LAN ports connected on the same LAN, the system will function as a single system supporting 8 PSTN lines, 16 local extensions and 16 remote extensions.

Notice that on some configuration pages, there are up to 4 tabs across the top. They allow you to select the TalkSwitch in the group.

Note: Specific details describing the configuration or the use of features related to the TalkSwitch models with VoIP lines, are preceded by the VoIP icon. If you have a TalkSwitch without VoIP, you can skip these sections in the guide.
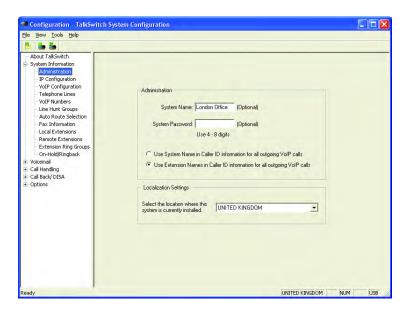
2.3.3 System Information

Click the ⊞ next to System Information to expand it. Each item is described in the 2.3.3 sub-paragraphs.

Note: The configuration screens may slightly vary by model.

2.3.3.1   Administration

This screen allows you to assign a System name and Administrator password for TalkSwitch. The Administrator password gives access to all configuration options. The password that is entered is used at configuration start-up and throughout Touch Tone configuration.

System Name: For tracking purposes, a system name can be assigned.

System   To activate the use of a system password, enter a 4- to
Password: 8-character numeric password. The password is also used
                    to gain access to the system using a touch-tone phone.

                    If you do not want a password leave the field blank. To
                    delete a password, enter the existing password. Click on
                    the password field again but do not enter anything and
                    click 'Ok'.

                    For TalkSwitch models with VoIP lines, the system name
                    or extension names can be used to appear as Caller ID
                    information for all outgoing VoIP calls.

Note: Sections 2.3.3.2 and 2.3.3.4 only apply to models with a LAN port. Owners of models without LAN ports can go to section 2.3.3.5 — Telephone Lines.

Note: An administrator password must be set to avoid unauthorized access to system configuration, including external access to system commands.

Location Settings

Subject to the system model, a Location field may display in the System Information Administrator screen. If the field is shown, set it to the country where the system is being used. If different service providers are identified for the country, select the service provider to which the system's telephone lines will be connected. Note that setting or resetting this parameter will change the system setting to the default values required for the country or service provider. The Location setting is shown in the bottom right hand corner of all configuration folder windows. If the Location field is not available on the Administrator page, the system is preconfigured for the location shown in the bottom right hand corner of the window.

2.3.3.2   IP Configuration

This section applies to models with a LAN port when VoIP or remote configuration will be used. There are two methods of configuring the IP settings: automatic and manual.

Configure TalkSwitch IP settings:

Click on System information (to expand)

Click IP Configuration.

The Radio button **"**Obtain IP and DNS information automatically" is selected.

System IP Settings section

If you have a DHCP (Dynamic Host Configuration Protocol) server and your TalkSwitch unit(s) is (are) connected to the LAN, all fields are filled with the correct information.

VoIP

Public WAN IP Address

Type of public WAN IP address for Internet Connection:

Select dynamic if your public IP address is a dynamic IP address from your Internet Service Provider (ISP). If you are unsure if your connection is dynamic or static, leave this option set to 'dynamic'. When set to dynamic, TalkSwitch checks your public IP address every 5 minutes. If it changes, TalkSwitch automatically updates to the correct information to manage VoIP calls properly.

Current public WAN IP address

When you select the public IP address type 'dynamic', this entry shows the current public IP address and is not editable. If the public IP address is 'static', you enter it here.

Public WAN IP address-checker server name

When you select 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. There is an IP checker application running on the server that responds to requests from TalkSwitch units for the public IP address at its location.
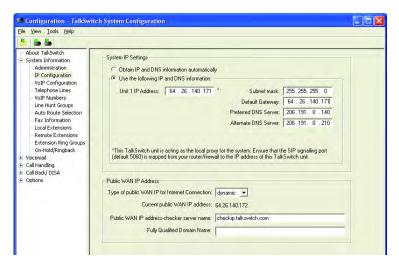
Fully Qualified Domain Name

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**.

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.

The radio button 'Use the following IP and DNS information' is selected

System IP Settings section

If there is no DHCP server present, the following fields need to be completed if you plan on using the VoIP capabilities of the TalkSwitch.

Unit n IP Address:

Assign an available static IP address to each TalkSwitch unit, where 'n' refers to the unit ID (1-4). If you use a DHCP server, you need to reserve an IP address for each TalkSwitch unit. To reserve an IP address, you need the MAC address of each unit. See section 2.3.2.1 for details on checking the MAC address.

Subnet Mask:

Enter the subnet mask for the LAN. If you have a DHCP server running, this information is obtained automatically.

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 preferred DNS server's IP address. This is also known as the primary DNS server. DNS is a Service that maintains information about a portion of the Domain Name System (DNS) database and responds to DNS queries for determining an IP address resolved from a Domain Name. For more information about DNS, see Chapter 5: VoIP Information.

Alternate DNS Server:

Enter the alternate DNS server's IP address if applicable. This is also known as the secondary DNS server.
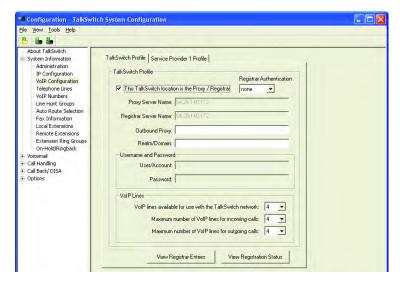
Public WAN IP Address (see previous section)

VoIP

2.3.3.3   VoIP Configuration
This section deals with configuring your TalkSwitch for VoIP.

Expand the System Information folder and click VoIP Configuration.

In the "TalkSwitch Profile" tab

TalkSwitch has a built-in SIP Server Proxy/Registrar/Redirect server. This facilitates the configuration and maintenance for multi-branch and teleworker applications. One location needs to be assigned as the SIP Server to manage call requests between VoIP locations and the other locations need to register with the SIP Server location.

To facilitate calls between TalkSwitch locations, we recommend that you use TalkSwitch location codes 250-299 as phone numbers for each of your VoIP numbers. Do not assign duplicate numbers between any two locations.

This TalkSwitch location is the Proxy/Registrar:

If this location is designated to be the Proxy/Registrar Server, check this box. When it is enabled and other SIP 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 location 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.

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, then 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 VoIP enabled TalkSwitch unit is acting as the Server, both, the Proxy Server Location and 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 to this unit. (for example, 'domain.com'). Note: This field is only 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 that 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 filled in 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 Numbers

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 requirements.

VoIP numbers 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.

Maximum number of VoIP numbers 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.

Maximum number of VoIP numbers for outgoing calls:
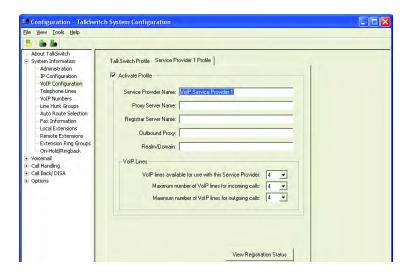
The maximum number of VoIP numbers available is determined by how many lines are available (shared) with the TalkSwitch VoIP network. For business reasons, you might want make all lines available for incoming calls and restrict outgoing calls so that the likelihood of incoming calls being missed.

View Registrar Entries (button)

If this location is the Registrar Server, click this button to view which systems are registered with the Registrar Server.

View Registration Status (button)

Click this button to view whether or not this system is registered with the designated Registrar Server.

In the "Service Provider Profile" tab

A VoIP enabled TalkSwitch can register with VoIP service providers to support calls using their service. Check with TalkSwitch for the list of Service Providers that support TalkSwitch inter operability.

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. 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. 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 numbers

The descriptions for the VoIP numbers under the TalkSwitch Profile tab are the same for the items below.

VoIP numbers available for use with this Service Provider:

Maximum number of VoIP numbers for incoming calls:

Maximum number of VoIP numbers for outgoing calls:

View Registration Status (button)
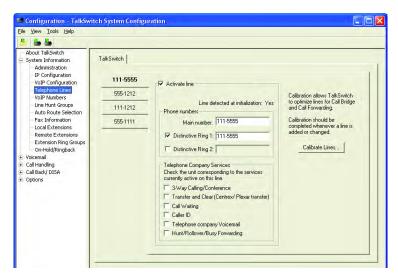
Click this button to view if this system is registered with the designated Registrar Server.

2.3.3.4   PSTN analogue telephone Lines

To configure telephone lines go to the System Information and Telephone Line folder. Select the tab for the telephone line to be configured. "Line Detected" shows if the 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 system's current settings.

> **Note:** If the units are networked, you need to configure the options for all the units. At the top of the window, select the tab for each TalkSwitch unit (labeled 'TalkSwitch 1', 'TalkSwitch 2', etc….) and follow the instructions below.

In the "Phone numbers" section

Click on Activate Line to enable the telephone line port. Enter the normal telephone number for the telephone line as the Main Number.

If your are using a Distinctive Ringing service for alternative telephone numbers from your telephone service provider check either Distinctive Ring 1 or Distinctive Ring 2 as required and enter the matching telephone numbers. When incoming calls are detected with these distinctive ringing patterns they will be handled as separate telephone lines and can be setup with different call handling (e.g. routed to a FAX machine at specific extension).

The distinctive ringing patterns are as follows:

Distinctive Ring 1 Two ring bursts per cycle of ringing (where normal ringing is a single ring burst per cycle); OR a single ring burst per cycle of ringing (where normal ringing is two ring bursts per cycle of ringing)

Distinctive Ring 2 Three ring bursts per cycle of ringing

In the "Telephone company services" section

Select any of the telephone company services that have been activated on the telephone line. See Appendix B for details of compatibility with telephone company services.

3-Way Calling/Conference

TalkSwitch can work with a service to forward calls to Remote Extensions or other external phone numbers while using the same line (Same Line Connect). You can use this service as an option for bridging calls to external numbers when calling into TalkSwitch from the outside. If you find that you are having difficulties with line volume levels when forwarding calls to Remote Extensions or using DISA, use 3-way calling to forward calls.

Transfer and Clear (Centrex/Plexar transfer)

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 TalkSwitch attempts to forward their call to the Remote Extension.

Call Waiting

Check this box if you have a Call Waiting service on the current line. It is strongly recommended that you remove this service from your line(s), since there is no way for TalkSwitch to answer a second call on the same line while it is handling the first call. You may want to consider adding the Hunt/Rollover service to your lines instead.

Caller ID

Check this box if the current line provides Caller ID information. TalkSwitch will pass along the Caller ID information to the selected extensions. Also, TalkSwitch can use the time information provided by Caller ID to update TalkSwitch's internal clock.

Telephone Company Voicemail

Check this box if you are subscribed to a Telephone Company Voicemail service.

Hunt/Rollover/Busy Forwarding
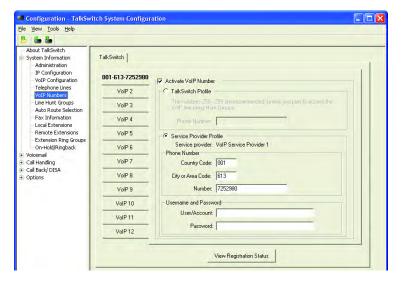
Check this box if the current line is associated with a Line Hunt or Line Rollover service from the Telephone Company. In general, any lines that belong to a Hunt/Rollover group should be configured to handle calls the same way.

Note: Section 2.3.3.5 only applies to models with VoIP trunks.

2.3.3.5　VoIP Numbers

The VoIP enabled TalkSwitch unit supports 4 VoIP lines per unit. Up to 12 VoIP phone numbers can be assigned for each unit so that unique call handling scenarios can be configured for up to 12 numbers. For 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 ring straight through to specific extensions in the office.

Activate phone numbers as required. If you had previously assigned numbers in the first slot for each of the VoIP numbers and performed a firmware update, the new layout will place these numbers in VoIP slots 1,4,7 and 10. Phone numbers need to be associated with the TalkSwitch network or a Service Provider network.

In the "TalkSwitch Profile" section:

All VoIP numbers are active when the system is first configured. Phone numbers need to be assigned so that inbound VoIP calls can be handled according to the configuration parameters set for VoIP numbers under the Call Handling section 2.3.5. Phone numbers can be one or more digits in length.

Phone numbers 250-299 have special functionality with TalkSwitch. These numbers 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 these numbers as it will facilitate calling from extensions and the Auto Attendants.

In the "Service Provider Profile" section

All VoIP numbers are active when the system is first configured. Phone numbers need to be entered exactly as provided by the Service Provider. For the United States and Canada, the Country code is 1. Enter the area code and phone number.

Enter a Username and Password. Every phone number has a unique Username and Password.

2.3.3.6   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 the Local and Remote Extensions and the DISA feature. Each Hunt Group can support several line choices and attempt to use an available line in the order that they are listed (Lines to Hunt) for that Hunt Group. These 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

Hunt Group 84:   selects telephone Line 4
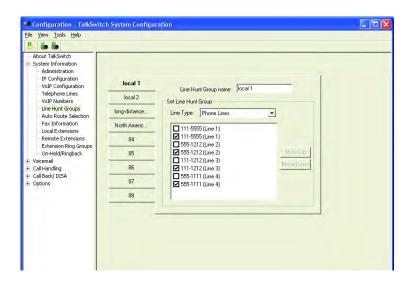
Hunt Group 88:   selects any available VoIP number (only applies
                        to units with VoIP trunks)

Select a Line Hunt Group and enter a unique name to identify this Hunt Group elsewhere in the configuration software.

Under 'Set Line Hunt Group', select 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 'VoIP numbers'.

Under 'Set Line Hunt Group', select the lines you wish to add to the Hunt Group by checking the box next to the line.

Note: If multiple TalkSwitch units are connected to the LAN, a check box is enabled, so that outbound calls always hunt the lines on the unit where the extension is connected. This minimizes network traffic between units.

2.3.3.7   Automatic Route Selection and Toll Restriction

Automatic Route Selection (ARS) allows outgoing calls to be redirected to different telephone lines or VoIP trunks and hence network operators based on the leading digits dialed to access geographical areas, mobile operators or services. For example, use this feature to block all long distance calls or redirect long distance calls to VoIP.

Toll Restriction allows calls to be blocked based on the geographical area code, mobile operator, service number or access code dialed.

The Automatic Route Selection (ARS) table is programmed by the system administrator with the leading digits of calls to be re-routed or blocked. TalkSwitch routes outgoing calls through Line Hunt Groups to telephone lines or VoIP trunks, dialing the outgoing Line Hunt Group access codes "9" or "81" to "88" or directly when Direct Line access is enabled on an extension. When ARS is activated for any Line Hunt Group and the caller has dialed, their dialed digits are compared to the leading digit entries in the Automatic Route Selection (ARS) table.

The comparison is based on finding the maximum number of leading digits that match. If a match is found the call follows the action specified in the ARS table. The action can be to block the call with an announcement or re-route it to an alternative Line Hunt Group. The Line Hunt Groups are setup to direct calls to specific telephone lines or VoIP trunks connected to different network service operators. If there are no matching entries in the ARS table, the call proceeds without any redirection or restriction to the Line Hunt Group whose access code was originally dialed by the caller or specified for direct line access.

Planning Automatic Route Selection and Toll Restriction

- Identify the leading digits of the numbers for the calls to be routed or blocked.

- Identify which telephone lines and VoIP trunks are connected to different networks and services.

- Identify which extensions' calls you wish to route or block.

Configure Line Hunt Groups (also, see section 2.3.3.6)

Ensure that the Line Hunt Groups are configured.

In the TalkSwitch **Configuration** software, click **System Information -> Line Hunt Groups.** We recommend that each network service provider's telephone lines or VoIP trunks are assigned to a separate line hunt group. You can define a name for each line hunt group and adjust the order in which the telephone lines will be used for outgoing calls.

- Type a name into the *Line Hunt Group name:* field (up to 20 characters).

- Click the check box next to the lines you wish to include in the line hunt group.

- Click/highlight the telephone line you wish to move.

- Click the *Move Up* or *Move Down* button to reflect the desired selection order.

Overflow calls to another network when preferred lines are busy

Calls can overflow to an alternative network operator's lines when the preferred lines are busy. For example, if the first two lines are reserved for local calls and both those lines are busy, you can allow outgoing calls on the next line, even thought that line might be your preferred long-distance connection.

- Click the check box next to the line(s) where you want the calls to overflow if the preferred lines are busy.

- If necessary, click *Move Up* or *Move Down* to change the order of the lines.

Entering Leading Digits and Assigning Actions
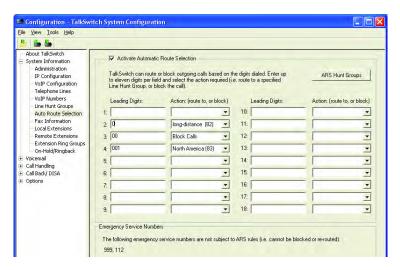
What are leading digits?

- Leading digits are the first digit or first several digits of an international access code, country code, area code or phone number.
- Leading digits can be anywhere from 1–11 characters.
- Leading digits can include numeric characters 0–9 and *.

The leading digit for long-distance is '1'. As soon as you dial '1xxx-xxx-xxxx', the system recognizes that you are making a long-distance call.

To place an overseas call, you would dial '011', followed by the country code, the area code and the phone number (i.e. 011 xx xx xxx-xxxx). Once you dial '011', the system recognizes that the caller is placing an overseas call. '011' are the leading digits.
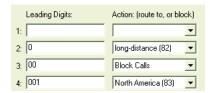
ARS will route or block calls according to the leading digits you define and the actions you chose in the TalkSwitch **Configuration** under **System Information -> Auto Route Selection**.
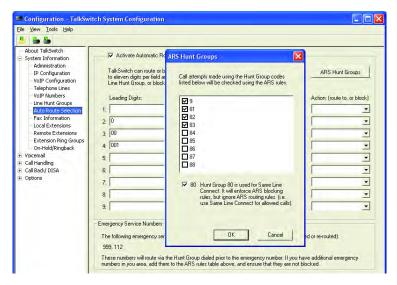
- Enter the leading digits you wish to route or block in the *Leading Digits* field.

- Select the desired *Action* (route to a specified line hunt group or block calls) from the corresponding pull-down list.

> **Note:** The entries that you specify for ARS match the dialed digits with the longest *leading digits* entry. If the numbers dialed match the digits that you specified in table shown above, the call is routed to a line hunt group or blocked according to the *Action* you selected.

Example:

- If you dial '9' followed by a phone number that does not start with '1', '01' or '01161', you access an outside line. The number is automatically routed according to the configuration of line hunt group 9.

- If you dial '9' followed by '1', a 3-digit area code and a 7-digit number, you access a line for long-distance calls in North America, as defined in line hunt group 82.

- If you dial '9' followed by '01', the call is blocked. At the extension, you hear a prompt stating "*You are not permitted to dial this number*" and then a normal dialtone.

- If you dial '9' followed by '01161', an area code and a phone number, you access a line for calls to Australia, as defined in line hunt group 83.

Set Local Extension Hunt Group Access
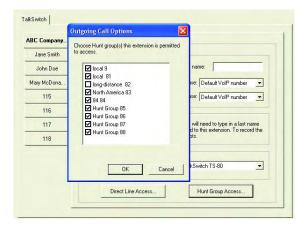
By default, all line hunt groups for each local extension are enabled. If you wish to restrict access to some line hunt groups, you need to disable the line hunt groups at the local extension.

- Open the TalkSwitch Configuration software.
- Select System Information
- Click on Local Extensions.
- Click the *Hunt Group Access* button.
- Remove the check mark (by clicking it) next to the line hunt groups you do not want to allow for that local extension.

Repeat these steps for all your local extensions. When you allow access to line hunt groups that have ARS activated, the calls are routed through ARS

Direct Line Access

If you enable *Direct Line Access* on an extension, calls placed from that local extension are routed to the line hunt group you designated. If ARS is enabled on that line hunt group, the calls are routed through ARS.

Additional Notes

Emergency Access Numbers

The emergency access numbers, shown in the Emergency Service Section of the ARS configuration window, are always allowed through ARS, even if the line is blocked. These numbers are dependant on the region and change based on the country/location settings.

Password Protection

We recommend that the administrator's password protection is enabled when you use automatic route selection and toll restriction. Enable the password protection in the TalkSwitch **Configuration** software under **System Information -> Administration.**
Enter a 4–8 character numeric password.

Telephone Line Three-Way Calling Services

The use of the telephone company's three-way calling service is not recommended when ARS is used. The TalkSwitch ARS has no control over routing of calls through this feature.

Call Detail Record Logging (CDR)

You can retrieve the records generated by the calls that are made through the TalkSwitch system. Please refer to *Call Detail Record Logging* TalkSwitch Quick Guide.

Automatic Route Selection Examples

Blocking international calls, with exceptions

Leading Digits	Action	Objective
00	Block Calls	International calls are restricted.
00	Hunt Group 81	North American calls are allowed.
08	Hunt Group 81	Redirect calls to mobile operators with access codes 08 to different line, that may be equipped fixed GSM phone services.
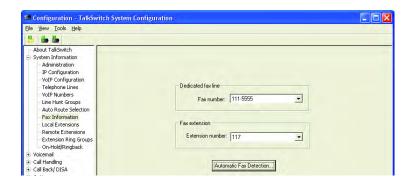
> **Call Redirection Advisory**: Use of the Automatic Route Selection and Toll Restriction features does not imply any guarantee whatsoever by regulatory authorities, your network operator(s), TalkSwitch or its distributors and resellers, with regard to the accuracy of these features and that the use of such a features may not be accepted by the network operator in any disputes which may arise regarding the accuracy of any subscriber's telephone account.

2.3.3.8 Fax Information

If you have a dedicated fax line or a Distinctive Ring number for a fax ma shine associated with a line, choose the fax number in the list. If you don't have a line dedicated for inbound faxing, select 'None'.

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

Select the extension number where you have your fax device connected. For example, if you have your fax machine connected to E8 on the rear of TalkSwitch unit 1, select extension number 118.
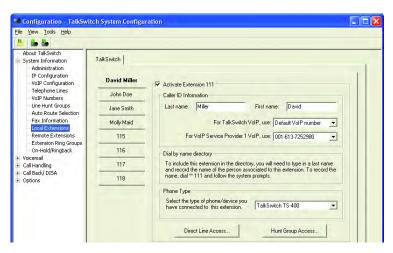
Note: If your fax machine is connected to E8 on a TalkSwitch with Unit ID 2, the fax extension number is 128. If you have a dedicated fax line or number, your fax machine should be connected to an extension port on the same TalkSwitch unit where the fax line/number is connected.
By default, when both, a dedicated line and 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.

2.3.3.9 Local Extensions

Local Extensions are phones or devices connected to any of the extension ports at the back of TalkSwitch. See sections 1.3 and 1.4 for more details on connectivity.

Note: If you have multiple units configured together on the same LAN, each unit has a separate tab at the top of the folder. Local extensions are numbered 1nx, where n is the number of the unit on the LAN and x is the extension.

Select the extension(s) you wish to configure.

Local Extensions are activated with the associated Voice mailbox by default. Voicemail boxes for each Local Extension can be de-activated in the Voicemail section in the System Information folder.

Caller ID Information section

Last Name/First Name:

Enter the first and last name of the person associated with the extension. The last name is required if the person is included in the 'Dial by Name' directory listing used by the Auto Attendant. The person using that extension must also record their name for the directory by following the instructions below the name fields. The name entered displays on the Caller ID of other Local Extensions during intercom calls.

VoIP

For TalkSwitch VoIP calls, use: [this number]

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 in the same way.

For [Service Provider] VoIP calls, use: [this number]

Select the number you want to appear at the remote location when VoIP calls are placed from this extension. If 'Default VoIP Number' is selected, then the phone number associated to the outgoing VoIP port will be used. This selection makes the most sense if your VoIP numbers have been configured to handle all inbound calls in the same manner.

Phone Type

If you have a TalkSwitch phone set connected to this extension, then select the model of phone. TalkSwitch sets have some integrated features that will not work unless the proper phone is specified. Features include intercom paging and group paging over the speaker on the TS-200, TS-400 and TS-600 phones and message waiting counter.

The models of telephone sets available will vary by market.

Direct Line Access (button)

Direct line access allows you to configure the extension to access a specific Hunt Group automatically for outbound calls when the extension goes off hook. For example, you may have a fax machine connected to this extension and don't want to reprogram the speed dial phone numbers. In this case, you select 'Direct line access' and choose the appropriate Hunt Group. As soon as

the fax goes off hook, TalkSwitch hunts the specified lines for an available line. This feature allows users to avoid dialing extra digits to access a line.

> **Warning:** When using Direct Line Access, the telephone company line dial tone is first available when you pick up the handset. The following features are only available from the TalkSwitch internal dial tone:
>
> - Intercom Calling,
> - Hold Retrieve,
> - Line Hunt Groups,
> - Voicemail Retrieval/Access,
> - Call Pick Up,
> - P.A. Access,
> - Parked Call Retrieval,
> - Do Not Disturb functions and
> - entering Command Mode.
>
> To obtain an internal dial tone at an extension which has Direct line Access enabled, dial Recall after going off hook.

Hunt Group Access (button)

This is used if you want to permit or restrict specific Hunt Group access to the extension. Click the button and organize the appropriate line Hunt Groups that are permitted for this extension (by default, extensions have no restrictions).

2.3.3.10 Remote Extensions
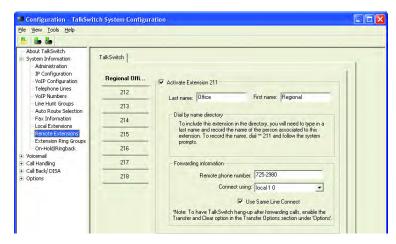
With TalkSwitch's Remote Extensions, you can have calls forwarded directly from your office to any phone number you choose. Your cell phone, pager or any out-of-office number becomes a 'Remote Extension' of your office system. You can choose to screen callers. If you are unavailable or choose not to accept calls, they can be transferred back to your office (e.g. to voicemail or another extension), or, forwarded to another out-of-office telephone number (e.g. to a business partner's cell phone) depending upon the Call Cascade configuration.

For details on how to use Remote Extensions, see sections 3.3.1 and 3.3.2.

Once your remote extensions are configured, calls can be forwarded by the Auto Attendant or transferred by someone in your office by dialing the Remote Extension number. There are 8 Remote Extensions available per TalkSwitch unit.

> **Note:** If a TalkSwitch has Unit ID 2, the Remote Extension numbers are 221–228. Unit ID 3 has remote extensions 231-238 and Unit ID 4 has 241-249.

Choose a Remote Extension to configure. You must enter a phone number in order for the extension to be considered active by the system.

Last Name/First Name:

Enter the first and last name of the person associated to the extension. The last name is required if the person is to be included in the 'Dial by Name' directory listing used by the Auto Attendant. The person using that extension must also record their name for the directory following the instructions listed below the name fields.

Forwarding Information

Remote phone number:

Enter the phone number of the Remote Extension (no specific prefixes are required other than area codes for long-distance calls). Enter the number the same way you are dialing it from a regular phone. You may use digits 0–9, space, comma, dash, **#** and *****.

Connect using:

Select the line Hunt Group that TalkSwitch will use to connect the Remote Extension. This is an advantage if you have an inexpensive long-distance plan set up for the lines in Hunt Group 84. You may want to use this Hunt Group for your long-distance Remote Extensions.

Use Same Line Connect

Check this box to allow incoming calls to use the same line of the original incoming call 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:" section.

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' in the Transfer Options page under 'Options'. If your service requires the remote end to answer the call before the transfer can be made, use the 'play accept/reject prompt' or 'play caller's name first' screening option set on the 'Answered at ext.' in the Remote Extension section under Call Handling. Check with your telephone company if you are uncertain which setting works best for you.

Note: Remote Extensions do not have the ability to transfer calls manually when using 'Same line Connect'.

Note: 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, do not use Same line Connect. It is best to experiment with different settings in order to decide what works best for your configuration.

Note: Remote extensions are designed to operate when the extension is on the network of local major telephone service providers. The feature may not function correctly with some telephone and mobile operators' networks, especially for international telephone numbers and mobile phones when roaming internationally. If the remote extension's service provider includes "prompted ring back" (an announcement offering a call back when the line busy), this service must be disabled from the remote extension's line. The remote extension feature does not operate on lines with "colour-me" ring tones. See Appendix B for compatibility with telephone company calling features.
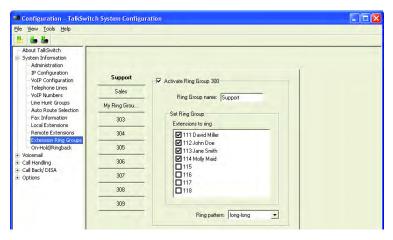
2.3.3.11 Extension Ring Groups

There are a total of 10 Extension Ring Groups available for your use. Ring Groups are extensions 300 to 309. By default, extension 300 is configured to ring all extensions.

Different uses for Extension Ring Groups

Send calls to specific groups of employees. For 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 300 to ring your extension with a different ring pattern. You tell your colleagues to choose extension 300 at the Auto Attendant or ask to be transferred to extension 300 when speaking to someone in the office. Without being subscribed to Caller ID, you can tell by how your phone rings if you consider the caller to be important.

Select the Extension Ring Group (300-309) you would like to configure. Activate it.

In the "Set Ring Group" section

Select each extension you wish to add to that Ring Group. Repeat this process for each of the Extension Ring Groups you want to activate.

Ring pattern:

Choose how you want the phone(s) to ring for the Ring Group.

2.3.3.12 On-hold/Ringback

If you wish to have callers listen to music or advertising while they are on hold, enable Music on Hold in this tab.

On-Hold Settings

There are 3 options for when callers are placed on hold.

1. The default setting is to play 'double beep' tones to callers when placed on hold. No additional configuration is required.

2. If you have a music source connected to the MUSIC port, select 'Play music whose source is connected to the MUSIC port'.

3. TalkSwitch can play a music-on-hold file once loaded on the system. This file must be in 8KHz, 8bit, mono, u-law format. You can convert MP3 and other file formats using a utility like Audio Converter or Ripper. Download a converter at: **www.audioi.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 TalkSwitch and make another test call.

Transfer Settings

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

> **Note:** If you connect a music source to the Music port but do not enable Music on hold, the caller hears both, music and a beep sound on hold. When the caller makes a selection at the Auto Attendant to go to an extension with call screening enabled, s/he will hear ringing and music.
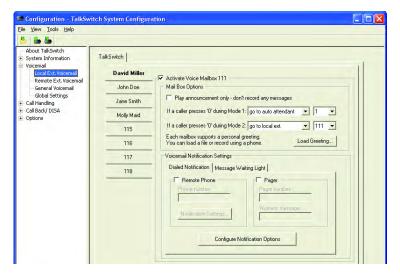
Note: If you have more than one unit connected to the LAN and you are using an external music source, you need to connect the source to all units. Obtain a 1/8" audio splitter available at most electronics stores. If you are using a .wav file loaded to TalkSwitch, no additional steps are necessary. The music file is copied to all units on the LAN.

2.3.4 Voicemail

2.3.4.1   Local Extension Voicemail

This section deals with the internal voicemail system included with TalkSwitch. It allows you to activate Local, Remote and General voice mailboxes and to change global settings to the voicemail system. We recommend to expand the memory for voicemail and auto attendants with a memory card, available at **www.talkswitch.com** or at your local reseller. For details on how to use voicemail, see Chapter 3: Using TalkSwitch.

All Local Extensions are activated with a voicemail box by default.

Mailbox Option section

Play announcement only:

If you want to use a local extension mailbox for informational purposes and prevent callers from recording messages, check the 'Play announcement only' box.

If a caller presses '0':

Select what action to take when a caller presses '0' during Mode 1 or Mode 2 while in a voice mailbox.

go to voice mailbox	Callers are transferred to the specified mailbox.
go to local extension	Callers are transferred to the specified local extension.
go to remote extension	Callers are transferred to the specified remote extension.
go to ring groups	Callers are transferred to the specified ring group.
play announcements	Callers are transferred to the specified announcement.
go to auto attendant	Callers are transferred to the specified Auto Attendant.
perform no action	No action is performed when '0' is pressed.

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.

In the Voicemail Notification Settings section

Under the **Dialed Notification** tab, set the manner in which you want to be notified when new messages arrive.

Remote Phone:

If you want to be notified at another location when new messages arrive, use Remote Phone Notification. Enter the phone number of the cell phone or location where you want to be notified.

Notification Settings (button):

Select Notification Settings to specify how you want TalkSwitch to handle notification calls.

Pager:

Click the Pager check box if you want to be notified by pager when new messages arrive. Activate the feature and enter the pager number and the numeric message to be displayed on the pager.

Configuration Notification Options (button)

Define how TalkSwitch dials out of your office when notifying you of a voicemail. If the first connection is not successful, define the number of times and the intervals 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.

Dial notification phone number using:

Select the Hunt Group you want to use for the Remote Phone and Pager Notification calls. It allows you to adjust the number of rings before it aborts the attempt and follows the parameters listed in the next section.
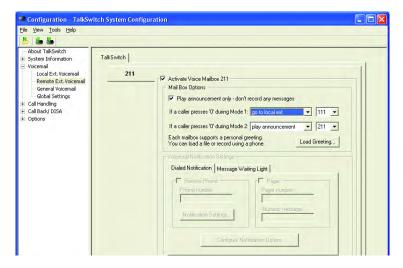
If attempt unsuccessful, try again:

These parameters control how many times TalkSwitch tries to notify the Remote Phone or Pager before it aborts the entire notification attempt. You can specify the time interval between attempts.

Under the **Message Waiting Light** tab, set a message waiting light to indicate when a message has been received for an extension. Select the local extensions in your system where the light is activated when a message arrives. The message waiting light turns on at the extension associated with a local mailbox by default.

This feature works with FSK MWI message compatible telephone sets. Stuttered dial tone is also provided.

2.3.4.2   Remote Extension Voicemail

Remote Extension voicemail is similar to the Local Extension voicemail. Ensure that the remote extension has been enabled so that you can activate its voicemail.

Please refer to section 2.3.4.1 for information on the following.

If a caller presses '0':

Go to voicemail box; Go to local ext.; Go to remote ext.; Go to ring groups; Play announcement only or perform no action.

Load Greeting button;

Voicemail Notification Settings tab;

Remote Phone Notification;

Notification Settings; Pager Notification;

Message Waiting Light;

Configuration Notification Options;
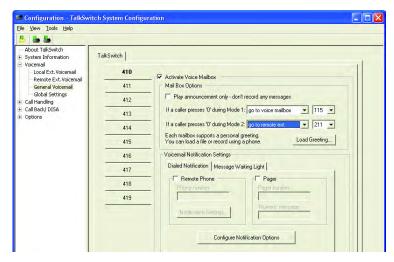
Dial notification phone number and If attempt unsuccessful.

2.3.4.3   General Voicemail

General Voicemail is similar to both Local and Remote Extension Voicemail. These are generic mailboxes used for general purposes like messages for a sales team or a technical support group.

Please refer to section 2.3.4.1 for information on the following.

If a caller presses '0':

Go to voicemail box; Go to local ext.; Go to remote ext.; Go to ring groups; Play announcement only or perform no action.

Load Greeting button;

Voicemail Notification Settings tab;

Remote Phone Notification;

Notification Settings; Pager Notification;

Message Waiting Light;

Configuration Notification Options;
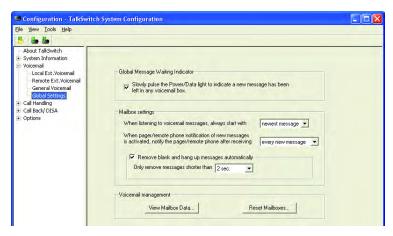
Dial notification phone number and If attempt unsuccessful.

2.3.4.4   Global Settings
Any changes made to Global Settings affect all system voice mailboxes.

Global Message Waiting Indicator

This option is helpful if you use a limited number of voice mailboxes. When a new message is left in any active voice mailbox, the Power/Data light pulses very slowly to indicate that there is at least one new message in one of the mailboxes. Once all new messages have been heard, this light returns to the solid state (default state).

Mailbox Settings

'When listening to voicemail...'
This option controls the order in which messages are played back when you listen to voicemail. The default setting plays back messages starting with the newest message and working back to the oldest. This setting is the most convenient. People store several saved messages and don't want to skip through all these messages every time they want to listen to newer ones.

'When pager/remote mail notification...'
This controls whether the system pages or calls your remote phone for every new message or only the first new message received in the voice mailbox.
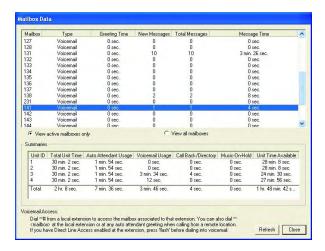
'Remove blank and hang up messages automatically
Only remove messages shorter than' (specify 1–5 seconds).
Check this box if you tend to get several messages where callers don't leave a message and hang up. Not clearing the short messages gives you the advantage of listening to the Caller ID information at the end of the message to identify callers (Caller ID service required from your telephone company).

View Mailbox Data (button)

View the details for each voice mailbox and announcement followed by a summary. This tool is useful for managing the voicemail resources of the system.

Reset Mailboxes (button)

Example: An employee has left the company and all greetings, messages and settings for a particular mailbox have to be reset. If you simply need to delete a password, use the Voicemail Manager located under the Tools menu.
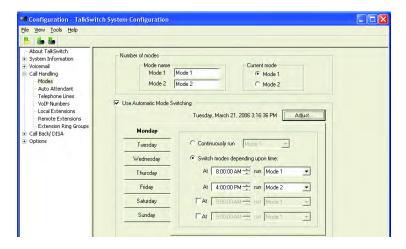
2.3.5 Call Handling

2.3.5.1   Modes

The use of Modes allows you to run different call handling configurations at different times. For example, you may want forward calls to your cell phone during the day and to your voicemail after business hours. Modes are

switched manually or automatically (with Automatic Mode Switching). Name the modes since names are displayed elsewhere in the configuration software whenever Modes are used.

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

To use the Automatic Mode Switching feature, activate it by clicking 'Use Automatic Mode Switching'. Set the start times for each mode. If you want TalkSwitch to change modes more than twice in one day, you can activate up to two extra mode changes.
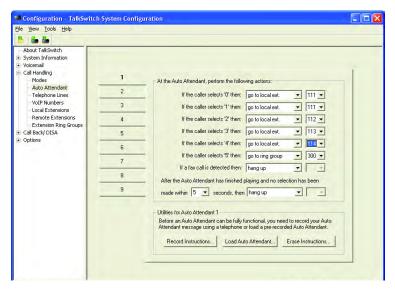
Note: If you do not want to use Automatic Mode Switching, you can switch modes manually in the software or through a telephone using touch tone commands. Enter command mode by dialing 🔘**#** and your password if required. Dial any of the following commands and press 🔘**#** to confirm.
31   Change TalkSwitch to mode 1.
32   Change TalkSwitch to mode 2.
30   TalkSwitch tells you what mode it is currently using.

2.3.5.2   Auto Attendant
The Auto Attendant greets callers and tells them which number(s) they should press on their telephone keypad to reach a particular extension, person, message, or department. Callers can dial Local Extensions (1xx), Remote Extensions (2xx), Extension Ring Groups (300-309), Voicemail (to both leave messages and access messages), access Call Back, access DISA and enter Command mode directly to make changes to the system configuration. 400 series mailboxes can be dialed at the Auto Attendant without

pressing **#** first. This is useful for providing 'virtual' Extensions associated with a mailbox for telecommuters or other part-time staff.

You can record or load up to 9 different Auto Attendants with a total combined time of 30 minutes (share with voicemail) or expand memory up to 9 hours using a TalkSwitch Memory card.

> **Note:** When two or more units are networked (models with LAN ports only), 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 TalkSwitch units on the same LAN. This feature helps reduce network traffic and allows any system to continue functioning even if another unit loses power or is disconnected from the LAN.

You can record the Auto Attendants from any Local Extension or remote phone using touch-tone commands. To load a pre-recorded message, see *Load Auto Attendant* later in this section.

Record Auto Attendant messages

Pick up any Local Extension (dial into TalkSwitch from an external source at the Auto Attendant) and enter command mode by pressing **#**. If you have an Administrator password, enter it. Begin entering the touch tone commands listed below. TalkSwitch 'walks' you through the recording. After entering a command, press **#** to confirm the entry before continuing with other commands or exiting command mode. To exit, press ***** or hang up.

Auto Attendant	Record message	Play message	Erase message
1	41	51	041
2	42	52	042
3	43	53	043
4	44	54	044
5	45	55	045
6	46	56	046
7	47	57	047
8	48	58	048
9	49	59	049

Example:

1. From the local extension go off hook and press **#** to enter command mode, then dial 4 1 **#**.

2. After the tone, record your Auto Attendant message. Press **#** when you are done.

3. Press 5 1 **#** to listen to your recording.

Note: If the Administrator password is enabled, you need to enter the password after you press **#** the first time to enter command mode.

Recording Auto Attendants from an out-of-office phone

1. Dial into TalkSwitch and wait for the Auto Attendant message.

2. Once the Auto Attendant is playing, use the same keypad commands as listed above. If Password Protection is enabled, you are prompted for your password.

At the Auto Attendant, perform the following actions

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, as opposed to 3 digits.
go to remote ext.	You can provide a single-digit option to ring a particular remote extension, as opposed to 3 digits (3 for Ring Group 300).
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 "press 1 for English, 2 for Spanish." Then 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, similar to Uniform Call Distribution.
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.
go to directory	If you want to enable callers to locate employee extension numbers by last name, select this option.

Note: TalkSwitch waits 1.5 seconds after your caller inputs a digit before following with another digit. If another digit is not 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.

If a fax call is detected then:

If TalkSwitch detects a fax call (identified by the CNG tones emitted from the fax), it can route it to a Local Extension. If you do not want to use fax detection, have TalkSwitch perform no action.

After the Auto Attendant has finished playing and no selection was made within x seconds then

If no selection is detected at the specified Auto Attendant, TalkSwitch can route the call to any Local or Remote Extension, Ring Group, Voice mailbox, Auto Attendant, Announcement, dial by name directory, queue the caller or hang up. This feature is included for callers who do not have a touch-tone phone or for a fax call where the calling fax machine does not emit a CNG tone. It can also be used to forward calls immediately after the Auto Attendant answers the call. You can specify the length of time after the Auto Attendant has finished playing before it performs the 'no selection' action.

Note: Fax calls are detected by a series of beeps called a CNG tone (not all fax machines emit a CNG tone). If a fax machine sends this tone while trying to establish a connection to a fax machine connected to TalkSwitch, TalkSwitch will handle the call accordingly. As a result, TalkSwitch is able to detect those fax calls and route them to the fax machine. To guarantee the reception of every fax, we recommend the use of a 'Distinctive Ring Number' added to one of your lines for fax calls. TalkSwitch can then handle all fax calls consistently. See section 2.3.3.8.

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 and Prompted Call Back. See section 2.3.6.1 for more details about configuring Call Back and section 3.7.2 for details on using Call Back.

Note: If you dial '6' at the Auto Attendant and no Call Back is 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 TalkSwitch is connected to another PBX system, see section 2.3.5.4 for more details.

If 80 – 88 or 9 is dialed

If you press '80 – 88 or 9' at the Auto Attendant, you access DISA if it has been permitted. See section 2.3.6.3 for more details on configuring DISA and section 3.7.1 for details on using DISA.

Utilities for an auto attendant

Record Instructions (button)

Click this button to display a dialog box showing the instructions on how to record and playback an Auto Attendant using a telephone. No adjustments are required for the recording volume. TalkSwitch adjusts the recording volume for optimal recordings.

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.

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.

Note: With the use of a memory card, you can extend the time for voicemail and Auto Attendant recordings to up to 9 hours.

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. Delete them to free up memory.
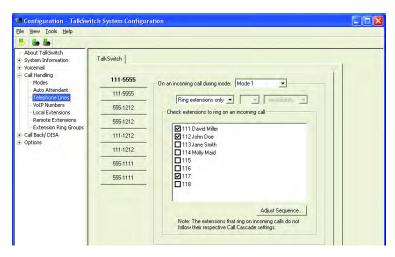
This section describes how TalkSwitch handles incoming calls based on the incoming line and ring cadence (distinctive ring numbers).

On an incoming call during mode 1 or 2

Select a line to configure and select Mode 1 or Mode 2 where you want TalkSwitch to use these settings. If you entered the phone number in the Telephone Lines section under System Information, the number displays on the corresponding button.

Ring Extensions only (no Auto Attendant is played)

If you don't want the Auto Attendant to answer calls on this line, select this option. All the Local Extensions are set to ring immediately during an inbound call by default. No Call Cascade actions (such as voicemail) can be performed if you select this option.

Select the Local Extensions you would like to ring. All activated local extensions ring by default. If you require certain Extensions to ring in a sequence (i.e. one extension rings first, then another one on the 2nd ring), click the 'Adjust Sequence' button to configure.

Note: If you call on any 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.

Play Auto Attendant

This option plays an Auto Attendant for incoming calls. Select the appropriate Auto Attendant (1–9) and the number of rings before the Auto Attendant answers.

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.
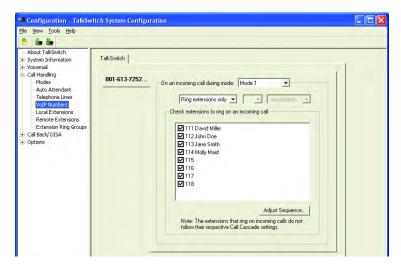
- If you have chosen the Auto Attendant to answer calls immediately, this section does not require configuration and is grayed out.
- If the Auto Attendant is set to answer after 1 or more rings, configure when specific Extensions ring before the Auto Attendant answers.
- If no Auto Attendant is selected to answer on the specified 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.

VoIP

2.3.5.4   VoIP Numbers

This section deals with how TalkSwitch with VoIP trunks will handle incoming VoIP calls.

First, select a VoIP number to configure. Then select the appropriate mode for which you want TalkSwitch to use these settings. If you entered the phone number in the VoIP Numbers section under System Information, the number will appear on the corresponding button.

Descriptions for VoIP numbers and Telephone Lines (section 2.3.5.3/4) on the four features below are the same.

Ring Extensions only (no Auto Attendant is played),

Play Auto Attendant or,
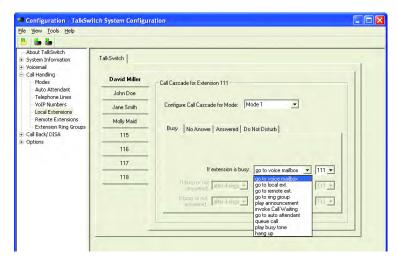
Go to Voice Mailbox.

and

Adjust Sequence... (button)

2.3.5.5   Local Extensions — Call Cascade

This section configures 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 "owns" 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 both scheduling modes (the scheduling for these modes are set up on the Modes tab).

If a call is manually transferred to a Local Extension that is busy, TalkSwitch follows the Call Cascade options for that extension.

There are 4 different 'Call Cascade' situations for Local Extensions.

1. Busy (at extension);
2. No answer (at extension);
3. Answer (at extension, routing when a call is rejected) and
4. Do not Disturb (when turned on at an extension).

For each of these situations, three 'cascade' destinations are available to control how and where you want 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,
    • invoke Call Waiting,
    • go to auto attendant,
    • queue caller,
    • play busy tone (example, if a fax machine is at the extension),
    • hang up.

    For more information on these features, see Chapter 3: Using TalkSwitch.

2. No Answer tab — routing options if this extension is not answered
    after x 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 who choose this
    extension at the Auto Attendant, to record their name. When you answer
    the call at the extension, it announces the caller (by playing their
    recording). 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.

    In the case you select 'hang up', TalkSwitch plays the following prompt:
    "I'm sorry, that extension is unavailable at this time. Please try again
    later."

4. Do not Disturb — routing options when this extension has engaged the
    'Do not Disturb' feature.

    The Do not Disturb feature is toggled on or off by dialing **＊** 62 from a
    Local Extension handset. Cascade options include forwarding calls to a:
    *   go to voice mailbox,
    *   go to local ext.,
    *   go to remote ext.,
    *   go to ring group,
    *   play announcement,
    *   go to auto attendant,
    *   hang up.
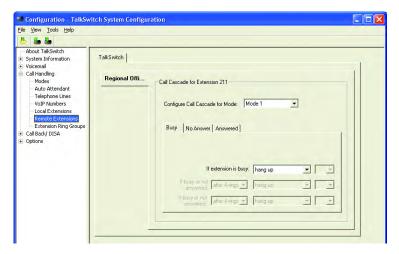
    When 'Do not Disturb' is enabled, you hear 'Do not disturb on' before the
    dial tone as a reminder that no calls ring at your extension. You may dial
    during this prompt.

Note: In the Cascade sequence, only choosing an extension (Local, Remote or Ring Group) allows you to choose an option in the next selection. Any other selection is considered an end-of-call option. Note that the last selection in the Cascade sequence only allows the choice of an 'end-of-call' option. This is done so there is no chance of a caller being "looped" within the system.

2.3.5.6   Remote Extensions — Call Cascade

Remote Extension Call Cascade works principally the same way as the local Extension Call Cascade. Please refer to section 2.3.5.5 and substitute the word Remote for Local. The Remote Extension Call Cascade does not have a 'Do not Disturb' option.

The Call Cascade routing options are only engaged if the extension is chosen from the Auto Attendant or if a call is manually transferred to the extension. Once a call enters an extension's Call Cascade sequence, the extension "owns" the call. It will not follow any other extension's options, even if chosen in the Call Cascade sequence. What you see in an extension's Call Cascade sequence is exactly what you'll get. You can set up the Call Cascade sequences for both modes (the scheduling for these modes are set up in the Modes tab).

For Remote Extensions, there are 3 different 'Call Cascade' options. For each of these options, you can control how and where you want the incoming call to the extension handled.

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,
   - play busy tone (example, if a fax machine is at the extension),
   - hang up.

   For more information on these features, see Chapter 3:
   Using TalkSwitch.

2. No Answer tab — routing options if this extension is not answered
   after x 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 prompts callers who choose this
   extension at the Auto Attendant, to record their name. When you answer
   the call at the extension, it announces the caller (by playing their
   recording). 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.

   If you select 'stay connected', the Call Cascade options are grayed out.
   TalkSwitch can control forwarded calls when prompting is enabled. If you
   use one of the prompted methods for this Remote Extension, have Music
   on hold enabled and are not using 3-way calling to forward calls to this
   Remote Extension, the Call Cascade is seamless to the caller. S/he only
   hears music throughout the entire process of trying to locate you.

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Note: 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 and not follow the Remote Extension's 'Answer at ext.' Call Cascade options.

2.3.5.7   Extension Ring Groups — Call Cascade

Extension Ring Groups — Call Cascade is similar to Local and Remote Call Cascade (sections 2.3.5.5 and 2.3.5.6).

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 "owns" 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.

For Ring Groups, there are 3 different 'Call Cascade' options. For each of these options, you can control how and where you want the incoming call to the extension handled.

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.

For more information on these features, see Chapter 3:
Using TalkSwitch.

2. No Answer tab — routing options if this extension is not answered after x 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 prompts callers who choose this extension at the Auto Attendant, to record their name. When you answer the call at the extension, it announces the caller (by playing their recording). 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.

2.3.6 Call Back/DISA
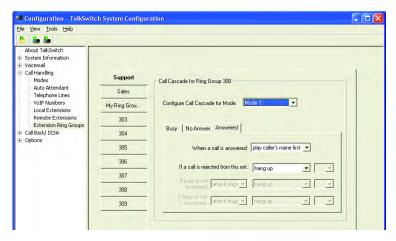Call Back and DISA act, working together, as your personal long distance operator. Whether you are across town or around the world, you can place calls from your office telephone or VoIP number(s), accessing your long distance savings plan.

> **PBX Fraud** (unauthorized use of the PBX and access to outgoing telephone lines):
> We recommend that you change Administrator, DISA and Call Back passwords frequently.

You can make a direct call to TalkSwitch, activate Call Bridge and enter the telephone number you want to dial. TalkSwitch dials the number from your office telephone or VoIP numbers and connects the two calls together (the call you made to TalkSwitch and the call TalkSwitch made from your office to the dialed party). This is how it works:

Step 1:

Call your TalkSwitch telephone number to activate Call Back, then hang up.

Step 2:

TalkSwitch calls you back within 10 seconds. You answer and accept Call Back. Your Auto Attendant plays if you have one recorded or the TalkSwitch message will play.

Step 3:

You can now access any TalkSwitch extension, voice mailbox or use DISA to make calls around the globe just as if you were sitting in your own office.

There are two ways to activate Call Back:

1. Auto Call Back — Auto Call Back is immediate and eliminates the initial long distance call charge required to activate Call Back.

    Before Call Back is activated, all the Call Back settings must be pre-configured. To use Auto Call Back, dial your office telephone number, let the call ring twice and hang up and Call Back is activated.

2. Prompted Call Back — Prompted Call Back offers the convenience of entering or changing your Call Back telephone number and message each time Call Back is activated.

    To use Prompted Call Back, dial your office telephone number. When the Auto Attendant answers your call, dial '6' to access the Call Back settings through a series of prompts which allow you to:

    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. It is
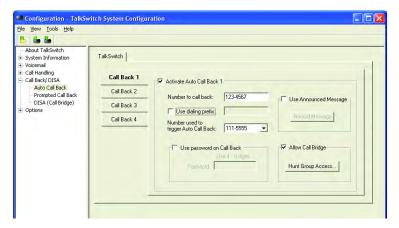
important to remember that a small initial long distance charge is incurred at your current location when you activate Prompted Call Back.

> **Note:** Prompted Call Back is not activated in the TalkSwitch configuration by default. You need to activate Prompted Call Back before you can use the feature. The Call Back out of the system to the number programmed always uses the same line that was dialed.

2.3.6.1 Auto Call Back

TalkSwitch's Call Back/DISA features gives you access to your office-based long distance savings plan from anywhere around the world.

You can configure four Auto Call Back numbers. Select an Auto Call Back 1–4. Click the check box next to 'Activate Auto Call Back x' (x represents a number from 1 to 4).

Number to call back:

Enter the Call Back number exactly as you would normally dial the number (i.e. do not enter a line Hunt Group in the dialing string).

Use dialing prefix:

If you have to dial the same prefix every time when using Auto Call Back (i.e. long distance calling card number), enable 'Use dialing prefix'. Enter the dialing prefix including any required pauses or feature keys like ✱ or #. Use a comma to insert a 2-second pause. For 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 ✱ 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.

Number used to trigger Auto Call Back:

Select the line/phone number that you dial when out of the office to trigger Auto Call Back. TalkSwitch initiates the Call Back using the same line you initially called. You can make configuration changes to Auto Call Back by dialing into this phone number only.

Use Password on Call Back:

The password option is designed to give you security and to restrict the use of this particular Call Back number. Enter a 4- to 8-digit password for the Call Back. When TalkSwitch calls you back, you are prompted for the password to accept Call Back.

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. For 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 instructions as follows:

Pick up a telephone handset connected to the TalkSwitch and use the following keypad commands to record the specific Auto Call Back Announced Message:

1. Press # to enter command mode. Enter the password if required.

2. Enter the appropriate command as listed below. After each command, press # to confirm the entry.

Auto Call Back	Recording	Playback
1	61	71
2	62	72
3	63	73
4	64	74

Note: There are 4 Auto Call Back accounts per TalkSwitch unit. If you need to record an Announced Message for a particular Auto Call Back account, it must be on the same TalkSwitch unit as the account.

Note: 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 and you can configure Call Back by pressing 6.

Allow DISA (Call Bridge):

To allow DISA (Call Bridge) after a Call Back has been accepted, check the Allow Call Bridge box. You may also restrict the Call Bridge line access by selecting the Hunt Group Access button. By default, Call Bridge is enabled when Call Back is activated. You will not be prompted for another password when you select a Hunt Group at the Auto Attendant.

Note: For a 2 second pause, dial ✱ 0 and for a #, dial ✱ #. When you have completed the entire dial string, press #.

2.3.6.2 Prompted 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 section 3.7.2.

Number to call back:

Enter this phone number exactly as you would normally dial the number from this line (do not enter a Line Hunt Group in the dialing string). TalkSwitch calls back from the same line the call originated.

Note: When you access Prompted Call Back from out of the office, you may change this number before initiating a Prompted Call Back. The benefit of Prompted Call Back is that you do not need to preset the number before you leave the office.

Use dialing prefix: (with Call Back number)

If you have to dial the same prefix when using Auto Call Back (for example a long distance calling card number), enable 'Use dialing prefix'. Enter the dialing prefix including any required pauses or feature keys like ✱ or #. Use a comma to insert a 2-second pause. For 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 ✱ 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.

This saves you from entering the same prefix every time you want to set up an Auto Call Back phone number.

Use Password on Call Back:

The password option is designed to give you security and to restrict the use of this particular Call Back number. Enter a 4- to 8-digit password for the Call Back. When TalkSwitch calls you back, you are prompted for the password to accept Call Back.

Use Announced message:

As described in the Auto Call Back (section 2.3.6.1), 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. For 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 instructions as follows:

Pick up a telephone handset connected to the TalkSwitch and use the following keypad commands to record the Prompted Call Back Announced Message:

1. Press **#** to enter command mode. Enter the password if required.

2. Enter the appropriate command as listed below. After each command, press **#** to confirm the entry.

   To record message: 65
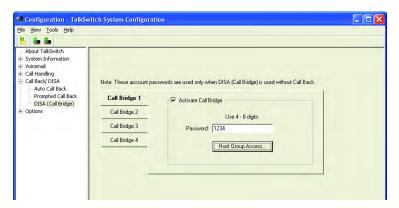
   To playback message: 75

Allow DISA (Call Bridge):

To allow DISA (Call Bridge) when a Call Back is accepted, check the Allow Call Bridge box. You can restrict the DISA line access by clicking the Hunt Group Access button. By default, Call Bridge is enabled when Call Back is activated.

> **Note:** The call back number shown is always the last number TalkSwitch dials in the last Prompted Call Back session.

2.3.6.3   DISA
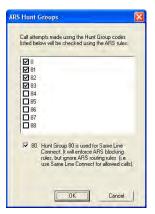
Call Bridge saves money while using Call Back or on its own. It bypasses the high long distance rates for cell phones.

How to use DISA (Call Bridge):

1. Call into TalkSwitch from any out-of-office telephone.

2. At the Auto Attendant message, dial the Hunt Group for which you have the account password (9 is default). You are prompted for the password.

3. Dial the password and press ⬤# to access a TalkSwitch telephone line.

4. At this point, you can dial a number the same way as from the office. Each TalkSwitch unit has four password activated Call Bridge accounts. Each account can be configured to allow access to certain Hunt Group(s) when using Call Bridge.

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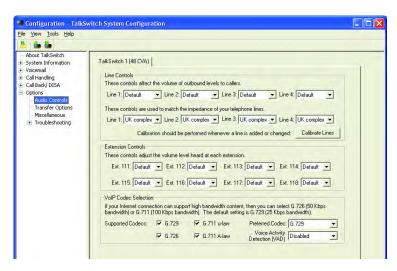
Line hunt group 80 is used for 'Same Line Connect'. To enable line hunt group 80, click the check box next to 'Allow the use of Same Line Connect [by dialing 80]'.

Note: Call Bridge account passwords are only valid when using Call Bridge by itself, not when used in conjunction with Call Back.

2.3.7 Options

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

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. To calibrate the lines once a line is added or changed, click the 'Calibrate Lines' button.

Extension Controls

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

VoIP Codec Selection

All 3 codecs are active by default. The preferred codec is G.729 (approximately 20Kbps bandwidth). For example, if you have a 'lite' version of a broadband service, you do not have enough bandwidth to support G.711 (approximately 100Kbps bandwidth) or G.726 (approximately 50Kbps bandwidth) codecs. 'Lite' connections typically have only enough bandwidth to support G.729. In this case, uncheck the G.726 and G.711 codecs so that the system does not attempt to use these codecs when setting up a call.

Using these codecs can cause one side of the call to sound 'choppy' because the broadband connection cannot handle the required bandwidth.
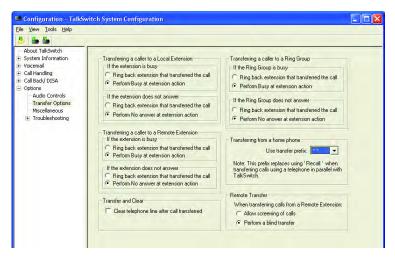
Voice Activity Detection (VAD):

A typical voice conversation may contain 35-50 percent silence. If customers need to conserve bandwidth, we recommend to enable VAD. When VAD is enabled, voice packets are suppressed during silence (i.e. an extension is on hold or there is no talking from the extension side of the conversation).

2.3.7.2   Transfer Options

This section describes TalkSwitch's actions when calls are transferred manually.

Transferring a caller to a Local Extension, Remote Extension or Ring Group

You can specify how transferred calls are handled if their target is a local extension, remote extension or ring group and if it is busy or doesn't answer. You have the choice of TalkSwitch ringing back the extension that

transferred the call or performing 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 TalkSwitch lines (for example, your home phone if you have a home-based business) to a Local or Remote Extension, Extension Ring Group, an Announcement, or a Voicemail box. 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 Recall 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. For more details, see section 2.3.7.2.

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
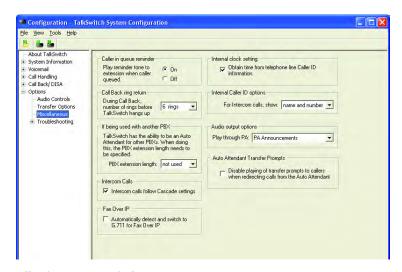
Calls that are transferred to a Remote or VoIP Extension, can be transferred back to another extension or voice mailbox. Dialing ✱ ✱ to place the caller on hold, then enter the extension or mailbox number. If this is set to allow screening calls, dial ✱ ✱ 4 to complete the transfer. If you want to cancel 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 a Remote Extension to another Remote Extension or VoIP Extension, it is always a blind transfer so that three lines are not tied-up for the transfer.

> **Note:** This feature can only be used when forwarding a call to a Remote Extension or a VoIP location over 2 lines. It cannot be used with 3-Way Calling or Centrex Transfer services.

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

Caller in queue reminder

When a caller is in queue at an extension already in use, a reminder tone plays at the extension to let them know they have a caller in their queue. This option allows you to turn off this function.

Call Back ring return

You can adjust the number of rings during a Call Back session before TalkSwitch disengages the Call Back (i.e. hangs up). 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.

> **Note:** This feature only works with PBXs that use Recall (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 TalkSwitch's Line ports. 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 TalkSwitch Lines.

3. Configure TalkSwitch's Auto Attendants to answer after x rings (answer immediately is used most commonly).

4. Record your Auto Attendants. The Auto Attendants 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 Doe's extension is 111 on the PBX), the Auto Attendant should say: "To speak with John Doe, 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 calls follows the Cascade settings of the extension dialed. If it is disabled, no fall-through actions takes place.

Fax Over IP

A VoIP enabled 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 lines, this option defaults to the active position. If it is de-activated, TalkSwitch obtains its time information from the time and date set in the 'Mode' tab under Automatic Mode Switching. TalkSwitch's 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 generate name and number Caller ID to each extension during intercom calls. If your Caller ID unit/telephone does not support the name option, choose to have TalkSwitch send the number only.

Audio Output Options

TalkSwitch can output different audio to the PA port. The default setting is to play only PA announcements through the PA port by dialing ✱ 0 from any extension.
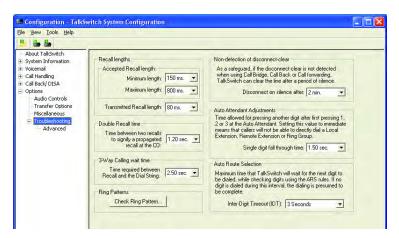
Other options include Voicemail Screening, useful in a home environment, Demo with a router, used for demonstrating the product and Static VoIP Demo, used strictly for demonstrating two TalkSwitch units connected to a hub or a switch and acting as two separate locations.

Auto Attendant Transfer Prompts

TalkSwitch can disable playing of transfer prompts to callers when redirecting calls from the Auto Attendant.

2.3.7.4   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.

Recall (Flash) Lengths

Recall or Flash is a timed disconnect of the telephone line to signal the PBX, or telephone network. The timing of the duration of the signal is dependent on normal practise of the country or telephone company. The location field on the System-Information and Administration folder will

set the normal setting for the country or telephone company. It is recommended that extension telephone sets with Recall, R, or Flash buttons are used, to ensure the correct signal duration is sent. Do to mix telephone sets on the same system with different settings for Recall or Flash.

Accepted Recall (Flash) length settings will set the minimum and maximum duration accepted from an extension. Any disconnect signal from an extension telephone set longer than the maximum will be treated as if the extension has hung-up to clear the call

Transmitted Recall (Flash) length: Must be set to the signal duration required by the telephone company. This signal is only used with the telephone company Three Way Calling/Conference feature.

Double Recall time

When two recalls are detected, TalkSwitch needs to calculate the time between the recalls to establish if they were intended to propagate a recall at the Telephone Exchange or to place a call on hold and then quickly retrieve it. The Double Recall time values signify the maximum time between two recalls that TalkSwitch recognizes as an intended recall to the Telephone Exchange. When two recalles are detected with a time gap larger than the value displayed, no recall is propagated at the Telephone Exchange. 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 recall hook and a dial string. Some telephone company 3-Way Calling/Conference services require different time allowances after the recall signal to re-establish dial tone. The default time is 2 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 displyed 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 finished.

Auto Attendant Adjustments

Since TalkSwitch allows callers to dial 100, 200, 300 series extensions and 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 after the first digit is pressed TalkSwitch waits for another digit before following the single digit option. There are 3 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 (1xx), Remote Extension (2xx), or Ring Group (3xx) 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 more than enough time to dial all 3 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.
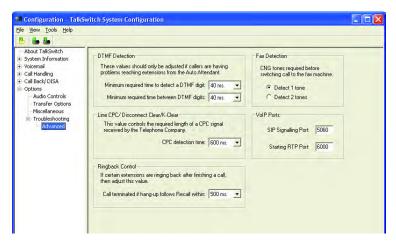
Auto Route Selection

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

2.3.7.5   Troubleshooting — Advanced

DTMF Detection

The minimum length for detecting a DTMF digit and minimum duration between DTMF digits is set by default, based on location. If problems are encountered these settings can be adjusted. Symptoms of issues that may require adjustments are the dialling of wrong numbers between extensions, or TalkSwitch not operating correctly to dialled digits at the Auto Attendant.

Line CPC/Disconnect Clear

This parameter controls when the system assumes a call on a telephone line has been cleared from the network, by detecting a disconnection of the telephone line. It is set to a default value based on Location. Some telephone companies use a specific timed disconnect signal (K-Clear) to indicate when a call has been released. In other cases it is set to a value to release the call if the line is no longer connected to the network.

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 recall 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 till it stops the ringbacks.

Fax Detection

If fax calls are not directed properly 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, then set this value to detect 2 tones.

VoIP Ports

VoIP

SIP Signalling Port:

The default SIP 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 (i.e. 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. For example, if your SIP 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

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 and helps to improve your corporate image. Once it has been configured and its messages have been recorded, the Auto Attendant takes care of the rest. It will follow the instructions it was given in the configuration. For details on configuring Auto Attendants, see section 2.3.5.2. For details on configuring TalkSwitch to answer calls using Auto Attendants, see section 2.3.5.

When calls are answered by an Auto Attendant, callers can dial any Local Extension (1xx), Remote Extension (2xx — the same applies to VoIP trunks, 250–299), Extension Ring Group (300–309), choose options 0, 1, 2, 3, 4, 5 (if configured to perform specific actions), dial voicemail directly (✱ and mailbox), send faxes, or 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, including Call Back (dial 6), DISA (dial 80–88, 9), retrieve voicemail (dial ✱ ✱ and mailbox) and enter command mode to make system changes (dial #). All of these features can be password-protected to eliminate system tampering from outside users.

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. In order for calls to go to an extension's mailbox, the Auto Attendant must first answer the call or a person in the office must transfer the call. For details on configuring TalkSwitch to ring specific extensions during inbound calls,

see section 2.3.5. Once calls have been answered at a Local Extension, they can be transferred to any extension or voice mailbox. See section 2.2.4 for more details.

3.2 IN THE OFFICE — MAKING AND RECEIVING CALLS

3.2.1 Making calls from a Local Extension

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 Extension, Remote Extension, VoIP Extension or Extension Ring Group by picking up your extension's handset and dialing the number of the extension you want to intercom to. An Intercom call is identified by a unique ring cadence (2 short rings). Exceptions: For a Remote Extension Intercom call, the phone rings as usual. If a Ring Group has a ring cadence other than 'normal'. Intercom calls follow the cascade settings of the extension dialed (if this feature is enabled in the 'Miscellaneous' section of the configuration software).

You can perform intercom paging and activate the speakerphone at other local extensions that have the TS-200, TS-400 or TS-600 phone sets. To place an intercom call to one of these sets, dial *84 + the local extension number (1xx) at the TalkSwitch dial tone.

Place 'out-of-office' calls from a Local Extension

To place a call to an out-of-office telephone number from a Local Extension, you must first access an available line by choosing a Hunt Group (9, 81-88) at the internal dial tone. Restrictions to specific Line Hunt Groups can be configured for each extension. Your extension can also be configured to Directly Access a specific line (i.e. when a handset is picked up, you hear the Telephone Company dial tone and you can dial an outside telephone number as usual).

> **Note:** If you have Direct Line Access enabled for a Local Extension, you need to dial 'Recall' before dialing any system features (i.e. extensions or voicemail).

3.2.2 Receiving calls at a Local Extension

To receive calls at a Local Extension, pick up the extension's handset when the extension rings. If the call went through the Auto Attendant to the extension and call screening is enabled (configured on the Local extension and Ring Group pages), you have the choice of accepting the call by pressing **#** or, re-routing the call to its 'Answer at ext.' Call Cascade options by pressing ***** or hanging up.

Call Pick Up

To intercept a call that is ringing on any other Local Extension before or after the Auto Attendant has engaged, pick up your handset. At the TalkSwitch internal dial tone, dial **＊**9. This routes the call to your Local Extension.

To answer a call ringing specifically at another extension before or after the Auto Attendant has answered, dial **＊**7 and the extension number (i.e. **＊**7112 to pick up a call ringing at extension 112). If you are in the middle of a call and want to connect to another incoming caller, press 'recall' to place the first caller on hold, then dial **＊**9 or **＊**7 and the extension number to connect to the second caller. Once you are finished with the second call, or, you would like to return to the first caller, press 'recall' '5'.

3.2.3 Placing calls on Hold at a Local Extension
Place calls on hold

To place callers on hold, press 'Recall' or 'Hold' on the TalkSwitch TS-200, TS-400 or TS-600 phones. Follow the same procedure to take callers off immediate hold. If you have multiple callers on hold or in queue at your extension, press 'recall' '7' to retrieve callers on a first in/first out basis.

> **Note:** If you press the 'Hold' button on a regular phone, you are not 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 a Local Extension to another Local Extension, Remote Extension or Extension Ring Group

To initiate an Unscreened Transfer, put the caller on hold (dial 'recall'), dial the appropriate extension number and hang up. The caller is taken off hold when the receiving extension answers. If the transferred call is not answered after the number of rings specified in the No Answer setting for the destination extension or if the extension is busy, one of the following happens:

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

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

> **Note:** The option can be altered in the Transfer Options section under the Options folder.

To announce calls before transferring (Screened Transfer), put the caller on hold and dial the appropriate extension number. If the person at the receiving extension answers and wishes to take the call, hang up or dial 'recall' '4' to complete the transfer and get an internal dial tone. The caller is immediately taken off hold and connected to the receiving extension. If the person at the receiving extension does not wish to take the call, press 'Recall' '5' to return to the caller.

If you dial an extension that is busy or not answered, you can complete the transfer so that the caller is directed according to that extensions' Call Cascade settings.

> **Note:** If you tend to handle many calls, you may want to complete or cancel a transfer without hanging up. Recall '4' will complete the transfer and provide an internal dial tone while recall '5' cancels the transfer and connects you back to the caller.

From a Local Extension to any outside telephone number

To initiate an Unscreened Transfer, put the caller on hold and select a Line Hunt Group (dial 80 for Same Line Connect). Dial the outside telephone number and press 'recall' '4' to complete the transfer. You cannot simply hang up to complete the transfer because this disconnects the call.

If the person at the receiving extension does not wish to take the call, press 'recall' '5' to return to the caller.

> **Note:** If you have the 3-Way Calling/Conference or 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.

3.2.5 Call Park — Parking and retrieving callers
Call Park is a feature for placing a call on hold and then retrieve it from any other Local Extension. TalkSwitch has 10 Park Orbits in all models (500–509).

3.2.5.1   Parking a caller
To place a caller in the next available Park Orbit, press 'recall' and ✱ 510. You hear a confirmation that indicates that the caller has been parked successfully and into which Park Orbit. If you use the TS-200, TS-400 or TS-600 phones, press the PARK button or softkey.

3.2.5.2   Parking a caller using Auto Park

To place a caller in the next available Park Orbit, press 'recall' and (✱)510. You hear a confirmation that indicates the caller has been parked successfully and into which Park Orbit.

3.2.5.3   Retrieving a parked call

Once the caller is parked, retrieve the caller at another Local Extension by pressing (✱) (✱) and the appropriate Park Orbit. If Direct Line Access is enabled at the extension, you must press 'recall' first. If using the TS-200, TS-400 or TS-600 phones, press the UNPARK button or softkey and enter the Park Orbit.

3.2.5.4   Using Call Park with the Paging option

If the External Paging option is enabled on TalkSwitch, you may choose to announce over the P.A. system (by pressing (✱)0) that there is a call parked in a specific Park Orbit for a co-worker. For 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 'recall' before retrieving the parked call.

If you have a combination of TS-200, TS-400 and TS-600 phone sets in the office, you can page over these handsets individually or as a group.

3.2.6   Call Queue/UCD/Call Waiting — Queuing and retrieving callers

3.2.6.1   Queuing calls to a single extension

Call Queue is a useful feature for placing multiple callers on hold at your extension while you are on an existing call. Call Queue is enabled in the TalkSwitch software's Call Cascade section in the Call Handling folder for Local Extensions and Extension Ring Groups. Callers will get queued at your extension if you set the first level in the 'Busy' Call Cascade to 'queue call'.

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 (✱)".

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 and to return to the main menu, press ⬛✱ ."

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 1."

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 'recall' and '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 up, press 'recall' and '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 receive the Caller ID for the new call. To connect to the new caller, press 'recall' 7. To let the caller fall through to the next level of the cascade, ignore the beep.

3.2.6.2   Queuing callers to an Extension Ring Group

If you want to queue callers for a Ring Group, configure this action using an Auto Attendant. This feature is also referred to as Automatic Call Distribution.

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 star." 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 picked up, you hear a prompt

that provides the option to accept the queued call (press ⬤#) or leave the caller in the queue (press ⬤*) and return to the dial tone to make a new call.

3.2.7 Using the TalkSwitch Call Waiting feature

The Call Waiting feature is activated as a 'busy' option at the Call Cascade routing for any local extension. When it is activated 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:

1. 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.

2. 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 dial 'recall' 7 to put the current caller on hold and answer the new caller. S/he has the option to dial 'recall' 5. This terminates the current call and connects the new call. If the person at the extension does not want to take the new call, s/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).

3.2.8 Conference calling with TalkSwitch

Case A: 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 TalkSwitch's conference calling capabilities.

1. Establish a call with an outside party.

2. Press 'recall' 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 'recall' and '6' to establish the 3-way call.

Case B: Two outside callers and one Local Extension
(e.g.: Line 1 and Line 2)

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. If you don't have 'Direct Line Access' enabled at your extension, dial '9' or another Hunt Group before dialing the second phone number.

3. When the call is answered, press 'recall' '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 'recall' 80 to place the outside caller on hold and dial the number of the other outside party.

3. When the call is answered, press 'recall' 80 to establish the 3-way call.

3.2.9 Making and receiving calls using VoIP

VoIP

TalkSwitch models with VoIP trunks support inter-branch connectivity via PSTN and VoIP. Once properly configured, calls are made between locations 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 any extension or auto attendant. TalkSwitch automatically hunts 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 numbers for an available line. In order to reach a remote location, a VoIP phone number has to be assigned to the VoIP number you intend to call.

Example: You have 3 branch locations (New York, Miami, San Jose), all with VoIP enabled TalkSwitch units.

Location 1: New York
Phone numbers assigned to 4 VoIP numbers: 250-261 (12 phone numbers)

Location 2: Miami
Phone numbers assigned to 4 VoIP numbers: 270-273 (4 phone numbers)

Location 3: San Jose
Phone numbers assigned to 4 VoIP numbers: 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.

3.2.10 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 'barge in' on any telephone call on any of TalkSwitch's Lines.

Making and receiving calls on phones connected in parallel with TalkSwitch

Making Calls: Make calls on parallel-connected phones as if TalkSwitch were not sharing the telephone line with them.

Answering Calls: If a call is answered on a parallel-connected phone before the Auto Attendant engages, continue with the call as usual. TalkSwitch takes no further action. If a call is answered on a parallel-connected phone after the Auto Attendant engages, turn off the recorded Auto Attendant message using the phone's touch-tone keypad and dial '✱ ✱ 0'. Continue the call on the parallel-connected phone.

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, key in the 'Transfer Prefix ✱ ✱'. After you hear 2 'beeps', enter the extension or voice mailbox number 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 the extension is not answered within the 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.

Note: 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.

3.2.11 Modems and telephone line access

If your modem is attached to TalkSwitch as an extension, you can configure it to access your telephone lines in one of two ways:

1. Direct Line Access: Give the extension where your modem is attached 'Direct Access' to one of your telephone lines (i.e. you don't have to dial 9, or any other Hunt Group to access an outside line). This is useful if you don't want to change your dial-up options for the modem. See section 2.3.1.4 for more details on configuring Direct Line Access.

2. Hunt Groups: If you use a telephone and modem on the same extension, Direct Line Access is not the best solution. TalkSwitch extensions with Direct Line Access immediately connect to one of your lines when you pick up the telephone's handset, disabling you to select a specific telephone line or to access voicemail, intercom or other system features without first pressing 'recall'.

To have your modem access a specific line using a Hunt Group:

Change your modem's 'Dial Properties' to include the intended Hunt Group (81-88 or 9) followed by the letter 'w', before the 'Dial up Networking' telephone number (e.g. 9w555-5555). Including the letter 'w' after your 'Dial out Access' number instructs your modem to

a) dial the access number 81-88 or 9,

b) wait until it detects a dial tone, and then

c) dial your 'Dial up Networking' number.

If your modem has problems dialing out, use a comma (,) (2 second pause) instead of the 'w' for (wait for dial tone).

Note: Because TalkSwitch with or without LAN port are digital systems, modem calls will only operate at a maximum speed of approximately 32 Kbps.

3.3 OUT OF THE OFFICE — RECEIVING CALLS THROUGH CALL FORWARDING

3.3.1 The three ways to forward calls

1. Automatic call forwarding
   The Auto Attendant can automatically forward office calls to your call forwarding telephone number(s) (Remote Extensions). Use the Auto Attendant message to prompt callers to select your Remote Extension number(s). (e.g. "To speak to John Doe, 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. Put the call on hold at the Local Extension, dial the Remote Extension number (e.g. 211) to which you want the call forwarded and hang-up. 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 at ext.' settings configurable for each Local Extension, Remote Extension and Ring Group.

3.3.2 Transferring calls from a Remote Extension

Calls forwarded to a Remote Extension (not using Centrex or 3-Way Calling services), can be transferred to any local or remote extension, ring group or voice mailbox. This feature also applies to VoIP numbers 250-299.

Transferring calls

Calls are placed on hold at a Remote Extension by dialing ✱✱ anytime during a conversation. You hear the dial tone that you hear at a Local Extension. Perform any of the following actions at any time:

1. Press ✱✱ to retrieve the call placed on hold.

2. Dial any Local Extension, Remote Extension or Ring Group.

3. Dial ✱ and mailbox number to transfer a caller to a voice mailbox.

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 calls screening, you dial ⊛ ⊛ 4 to complete a transfer or ⊛ ⊛ 5 to cancel the transfer and return to the caller.

3.3.3 Screening options for forwarded calls

The screening options for Call Forwarding are configured in the TalkSwitch software using the Call Cascade's 'Answer at ext.' options available in the 'Remote Extension' window. See section 2.3.3.10.

Forwarding calls with screening

When 'play accept/reject prompt' is configured in the 'Answer at ext.' Call Cascade option, TalkSwitch plays a pre-recorded prompt to callers asking them to hold and dials your call forwarding telephone number. When you answer the forwarded call, TalkSwitch plays a pre-recorded prompt that identifies the call as forwarded ("This is Call Forward"). To accept the forwarded call, press # on your telephone keypad or, to re-route the call to its call screening Call Cascade options, press ⊛ or hang up.

When 'play caller's name first' is configured in the 'Answer at ext.' 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 that identifies the call as forwarded ("This is Call Forward, you have a call from...") and plays the recording of the callers name. To accept the forwarded call press # on your telephone keypad or, to re-route the call to its call screening Call Cascade options press ⊛ or hang up.

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, the call follows the Remote Extension's 'No answer at ext.' or 'Busy at extension' Call Cascade action.

- If the call was manually forwarded, the call follows the Remote Extension's 'No answer at ext.' or 'Busy at extension' Call Cascade action. This action can be changed to ring the extension back to the person who performed the transfer (Options>Manual Transfer).

- If the call was conditionally forwarded, the call follows the Call Cascade action in the 'No answer at ext.' or 'Busy at extension' for the original extension selected.

3.3.4 Calls over VoIP with SIP phones and Gateways

VoIP

If teleworkers are connected to the system via a certified compatible SIP device (i.e. gateway or SIP phone), there are several ways for that teleworker to interact with TalkSwitch.

Making calls

Once the phone or gateway is registered with the TalkSwitch 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 any extension, VoIP location or voice mailbox. To place calls through the regular phone lines, you must activate and configure a DISA account. To access regular phone lines dial '9' and enter the password for DISA.

Note: Currently, TalkSwitch does not support the ability to place calls on hold or transfer calls when making calls to other locations, since the call does not actually go through TalkSwitch.

Receiving calls

Once the phone or gateway is registered with the TalkSwitch 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 another SIP gateway or phone, you can transfer the call if your service provider supports this feature.

3.4 USING THE TALKSWITCH VOICEMAIL SYSTEM

Each TalkSwitch unit comes with approximately 30 minutes of voicemail recording capacity (shared with Auto Attendant greetings). This capacity can be increased by adding TalkSwitch memory cards. You can add 1, 2, 4.5, or 9 hours of storage to each unit. Contact your TalkSwitch dealer to purchase memory cards. When you insert a TalkSwitch memory card in the memory slot, the system configures the card automatically within 20 second.

3.4.1 Activating voicemail boxes

Activate Voice mailboxes in the TalkSwitch configuration software. There are three series of Voice mailboxes that you can activate in the Voicemail section of the configuration.

1. Local Extension mailboxes

   These mailboxes are generally associated with the Local Extensions. All Local Extensions are activated with a respective mailbox by default.

2. Remote Extension mailboxes

   The Remote mailboxes are associated with the Remote Extensions. If you activate a Remote Extension and want a voice mailbox associated with that extension, activate that mailbox in the Voicemail tab.

3. General mailboxes

   Activated these mailboxes in the General Voicemail window. They are not associated with particular extensions.

> **Note:** If you want inbound calls to go straight to a voice mailbox after x number of rings, program an Auto Attendant to answer and provide a fall through to voicemail. See section 3.3.3 for an example of this configuration.

Callers can reach voicemail or announcements via the Auto Attendant, Call Cascade, or be transferred from a Local Extension. To leave a message from a local extension, they pick up a Local Extension, dial **✱** and the mailbox number. To transfer a caller to voicemail, press 'recall' **✱** and the mailbox number.

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.

3.4.2 Retrieving messages/Accessing a voice mailbox

When a new message is left in a Local Extension mailbox, TalkSwitch plays a stutter dial tone to that extension when the handset is picked up. It can light up the message waiting light on the phone if the phone supports FSK Message Waiting Indicator. The 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.

To access a mailbox and retrieve messages, or, to change mailbox options, dial **✱** **✱** **#** from a Local Extension or **✱** **✱** and the mailbox number at the Auto Attendant if dialing from an outside location (or a local extension).

Local extensions can transfer employees to access voicemail by pressing 'recall', then pressing ✱ ✱ and entering the mailbox number. If Direct Line Access is enabled, be sure to press 'recall' before trying to access your mailbox.

> **Note:** If you have a line configured to go straight to voicemail, dial '8' during the greeting to access messages from outside phones.

The first time you access your voicemail, you are prompted to set up your mailbox by assigning a password (optional), recording a greeting and recording your name for the dial-by-name directory. There are no directory listings for the general mailboxes 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 The directory is related to the extension and its mailbox.

Once your mailbox has been setup and you access it through prompt instructions, you may do the following:

Listen to messages (press 1)

If there are messages in your mailbox, press 1 to access them. TalkSwitch always starts with new messages before it plays back any older messages. You can perform the following actions during or after the message.

Dial 1:	Rewind 10 seconds
Dial 11:	Rewind to beginning of message
Dial 3:	Skip ahead 10 seconds
Dial 33:	Skip to end of message
Dial 5:	Listen to time and date stamp
Dial 6:	Forward the message (with or without a note)
Dial 7:	Delete message
Dial 9:	Save message

Dial **✱**:　　　　　Back up one level

Dial **#**:　　　　　Skip to next message – This leaves new messages as new.

Change greeting options (press 2)

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 previous Auto Attendant (if the call came from the Auto Attendant) while listening to your greeting. When they are done recording, they can press **#** for more options. You may want to give your callers these options for their convenience.

Change personal options (press 3)

This allows you to change your current password, turn auto date and time stamp on and phone number on/off and set up or change Pager and Remote Phone Notification for new messages. By default, auto date and time stamp is on and this information is played at the end of each message. If you have Caller ID service, the phone number is also included with the time and date.

Record name for Dial by Name Directory (press 4)

If you not yet recorded your name for the directory or wish to change the recording, do so here. 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 and the extension number. It then gives callers the option to connect to that extension.

> **Note:** You must record your name in order for the extension to be included in the dial-by-name directory.

3.4.3 Pager and Cell Phone notification

When a new message is left in a voice mailbox, it can notify a pager, cell phone or any remote number if the mailbox is configured to do so.

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.

If you use Remote Phone Notification, you have the option to accept or reject the notification call when you answer. To listen to messages, press ⬛#. To postpone listening to messages, press ⬛*.

For details on configuring Pager and Remote Phone Notification, please see section 3.4.

3.5 MUSIC ON HOLD

TalkSwitch can play music to callers when they are on hold, parked or queued. The music played to the callers is provided by the source you have connected to the MUSIC port or from a .wav file (8 KHz, 8 bit, mono, u-law, wav format) stored internally on the unit(s).

When a .wav file is loaded on the TalkSwitch units, the file is duplicated to all units on the LAN for redundancy in case the LAN goes down or a particular unit is unavailable.

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 TalkSwitch's Call Forwarding, 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 tries to locate their party. This prompt explains to callers why all of a sudden they hear 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.

3.6 MODE SWITCHING OPTIONS

Manual Mode Switching

Use telephone keypad commands at a Local Extension or at the Auto Attendant to switch TalkSwitch Modes.

At the Local Extension dial tone or at the Auto Attendant, dial # to enter command mode and enter a password (if required). Dial any of the following Quick Commands and press the # sign.

Activate Mode 1:	3 1
Activate Mode 2:	3 2
Check Current Mode:	3 0

Automatic Mode Switching using the Time Scheduler

Use the configuration software to activate TalkSwitch's Time Scheduler feature. You can configure TalkSwitch to switch between Modes automatically by configuring the times for Mode changes in the 'Modes' window under Call Handling in the configuration software.

If you switch modes manually, TalkSwitch automatically switches to the proper mode during the next scheduled mode change.

3.7 OUT OF THE OFFICE — MAKING CALLS WITH CALL BACK AND DISA (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 telephone line(s).

3.7.1 Using (Call Bridge)

Call Bridge allows you to place calls using your office telephone lines when you are not in the office. You can make a call to TalkSwitch, access a 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 dial a long-distance number. To avoid long-distance cell rates, all you need to do is make the local call to TalkSwitch to access Call Bridge and make the long-distance call, accessing your office savings plan.

How to use DISA (Call Bridge)

Dial into one of TalkSwitch's Lines. When the Auto Attendant answers, select a Hunt Group (81 – 88 or 9) or Same Line Connect (80). You are then asked to supply your Account password to receive dial tone. If you are using Same Line Connect (80) and your line supports the 3-Way Calling/Conference service, you are prompted to enter the phone number.

When you have completed your call (or if the telephone number dialed was busy), do one of the following:

a) Make another call
   Press **#** **#**. TalkSwitch disconnects you from the call in progress. You can dial another number or redial the same number.

b) Activate the Auto Attendant
   Press **#** *****. TalkSwitch ends the DISA session and activates the Auto Attendant. Once the Auto Attendant message begins playing, you have the following options:

   - Dial **#** (and enter a password if applicable) to enter Command Mode.

   - Dial '6' to change/enter Call Back settings.

   - Dial one of your extension numbers (Local, Remote or Ring Group). This option allows you to contact someone in your office, check your voicemail, etc.

c) End the Call Bridge Session
   Hang up. This ends your DISA session and disconnects you from TalkSwitch. If you want to guarantee that TalkSwitch disconnects at the end of a DISA session, press **#** **#** and then hang up.

The DISA telephone number dialed can be any telephone number: local, long distance, toll free (800 and 888) or international. Enter the telephone number as you would in your office. For long distance calls, do not forget to include the '1', country code, and area code.

3.7.2 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 any of the following:

- Call Bridge
- Local Extensions, Remote Extensions and Extension Ring Groups
- Voicemail
- Configuration settings

Using Call Back involves 3 easy steps:

1. You call your TalkSwitch telephone number to activate Call Back.

2. You answer and accept the Call Back. (TalkSwitch calls you back within 30 seconds.)

3. At the Auto Attendant you can choose to perform DISA, dial any 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, dial a TalkSwitch telephone number. Let the line ring until the Auto Attendant answers your call and then dial '6' on the telephone keypad. If you have enabled password protection you are asked to supply your password. TalkSwitch then offers a series of prompts. You can enter 1 and hang up to initiate the Call Back, or, if you need to enter a new Prompted Call Back number, enter 2 to access the Prompted Call Back settings, then follow the prompts.

> **Note:** For Prompted Call Back, TalkSwitch always dials the last Prompted Call Back number entered. If you do not have the Auto Attendant set to 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, dial one of the Call Back telephone numbers that has been pre-configured for Auto Call Back activation. Let the line ring at least once but no more than three times and hang up before the call is answered. Within 20 seconds, TalkSwitch calls your Auto Call Back number.

> **Note:** It is important to remember that all Call Back settings must be pre-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. Call Back is normally accepted by answering the call and pressing **#**. If required, enter your password. This option is used when you have a direct line to your Call Back telephone number.

2. 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 Jane Doe 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.

3.8 UPGRADING THE TALKSWITCH SOFTWARE AND FIRMWARE

We are continually looking for ways to enhance your communications capabilities. When new features are added, we want TalkSwitch users to have immediate access to the update.

TalkSwitch supports firmware updates for adding new features. You can easily update your TalkSwitch. Visit our website at **http://global.talkswitch.com**.

Step #1 — Check current version
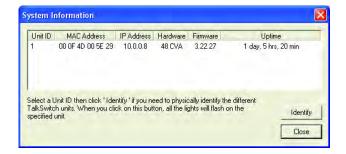
To identify the appropriate update file, you must determine your TalkSwitch firmware and software versions. To find your TalkSwitch software version number, go to the **Help -> About TalkSwitch Configuration Utility...**

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

> **Note:** The instructions on our website take precedence over any instructions in this user guide.

The TalkSwitch firmware version number of each unit is in the System information window. In the TalkSwitch Configuration start window, click the link **'View System Information'.**

Step #2 — Download new software and firmware

When you select and click on the appropriate version for download, a screen displays asking you what you would like to name the file you are about to download and in where (folder/directory) you would like to save it. We suggest leaving the file name as it is (e.g. install.exe) and saving it on your Desktop. 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 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. Follow the instructions on the website to perform the update.

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's front panel show diagnostic indicators for the update.

When the update is completed, you are prompted to reboot TalkSwitch. In the dialogue box, click the 'Proceed' button.

Note: If you have two or more units networked and are running 3.22 firmware, TalkSwitch automatically copies the files to the other units on the LAN for update. Updating units remotely over IP is also supported with release 3.21. Units do not operate properly if they are networked together with mismatched firmware versions. To confirm all units are properly loaded, select 'View Details' on the 'About TalkSwitch' page in the configuration software. See below.

If the firmware update was unsuccessful, please follow the instructions through the update process again. If problems persist, please contact your reseller or TalkSwitch.

Note: Firmware updates do not alter your system configuration. In a major system update, it may be necessary for the configuration file to be reset to default values. In this case, reconfigure the entire system. During a major system update, TalkSwitch mentions any issues involved with the firmware update.

If you replaced your computer with a new one, please use the software that matches the latest version of firmware running on TalkSwitch. If obtained the software from our website, download the install file. All install files on our website have both, the entire software program and the latest firmware.

CHAPTER 4

CALL DETAIL RECORD LOGGING (CDR)

This chapter covers the setup and retrieval of records generated by calls through the TalkSwitch phone system.

4.1 ENABLING CALL DETAIL RECORD (CDR) LOGGING

TalkSwitch can report the CDRs in "real-time" to a computer connected to the unit's serial port, or log the CDRs in a file on the unit itself.

1. Connect TalkSwitch to your computer and open the TalkSwitch **Configuration** software.

2. Click the **Tools** menu **-> Call Logging Output.**

3. Select **Real-time to Serial Port** or **Store to File on TalkSwitch.**

4. **File -> Save to TalkSwitch**

When storing the CDR to a file on TalkSwitch, approximately 6000 records are stored in a circular buffer. This means that sometime after 6000 records, the oldest records will be deleted as new ones are created. This amounts to 6000 records cover anywhere from a couple of days on a very busy system to over a month on systems that have less traffic.

When the CDR data is streaming real-time out of the Serial port, no file is generated on TalkSwitch, therefore, there is no buffer limit. As each event in a call is completed, the record will be sent out the Serial port.

4.2 RETRIEVING DATA

There are 2 different methods of retrieving the data from TalkSwitch:

- **Web Interface** — Store to File on TalkSwitch* (*only on models with LAN ports)
- **Serial Interface** — Real-time to Serial Interface.

4.2.1 Web Interface — Store to File on TalkSwitch
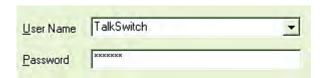
You can access the CDR information through an HTTP interface. In a web browser's Address Bar, enter the Box IP address of the TalkSwitch system.

1. In a web browser enter the TalkSwitch unit IP address followed by the forward slash and login.html

Note: You can confirm the unit's IP address from the
System Information -> IP Configuration page in the software.
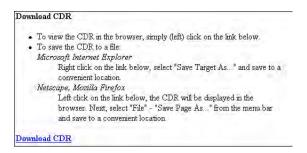
2. At the login screen, enter the System name for username, if set, and the administrator password in the password field. You can find this information in Talkswitch configuration software under
System Information -> Administration.

3. Click on the link for **Call Detail Record (CDR) Management.**

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.

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. To cancel, select **No.**

4.2.2 Serial Interface — Real Time to Serial Interface

In order to capture the data in a real time environment you must have a PC connected to TalkSwitch Unit ID 1 with a serial cable.

1. Launch Hyper Terminal by clicking **Start -> Programs -> Accessories -> Communications -> HyperTerminal**.

2. A window pops up prompting for a connection name. Enter **TalkSwitch,** then click **OK.**

Note: If you have not used Hyper Terminal before you will be prompted to set this program as a default telnet application and enter your area code.

3. In the **Connect using:** drop-down menu, select the PC COM port that the serial cable is connected to, then click **OK.**

4. Set the bits per second to 57600. All other values may be left at their defaults, then click **OK**.

5. Click on **Transfer -> Capture Text...**

6. Select the appropriate directory and filename to save the data. Click **Start**.

Note: It is recommended to stop the capturing of text periodically and restart saving it to a new file. This allows easier viewing and analyzing of information.

4.3 ANALYZING THE DATA

The data can be viewed either through the web interface or downloaded and opened into a spreadsheet (e.g. Excel) as a delimited text file. The log is broken up into various columns:

1. Call Type	5. Time	9. Name from
2. Log Number	6. Event Duration	Caller ID
3. Event Counter	7. Connection	10. Line
4. Date	8. Phone Number	11. Account Number

Example:

Type	Log#	Event	Date	Time	Duration	Conn.	Phone Nr.	Caller ID Name	Line	Account Nr.
I	12328	01	16/02/2006	8:38	0:00:37	A004	02055512233	Joe Smith	13	ABCDE
T	12328	02	16/02/2006	8:38	0:00:35	R302	02055512233	Joe Smith	13	ABCDE
T	12328	03	16/02/2006	8:39	0:00:17	M442	02055512233	Joe Smith	13	ABCDE
X	12328	04	16/02/2006	8:39	0:00:05	D103	01173345789	Mary Miller	19	
I	12329	01	16/02/2006	8:45	0:00:11	A137	02047812987	Jane Doe	21	
B	12329	02	16/02/2006	8:48	0:00:09	E195	02047812987	Jane Doe	22	0987C
O	14080	01	16/02/2006	8:49	0:00:36	C021	02099090022		10	
I	15652	01	16/02/2006	11:38	0:00:12	RC0	02099090022	Jen Jones		

Each column contains specific information related to the current state of the call. The following is a list of the columns with an explanation for each one:

1. **Type** — This column contains the following characters:
   a) **I** — Indicates an Inbound Call
   b) **T** — Indicates a Transfer State
   c) **O** — Indicates an Outbound Call
   d) **B** — Indicates a Bridged Call. For example a call in on one line that is forwarded over another. This will include calls forwarded to Remote Extensions.
   e) **X** — Indicates a Blocked Call
   f) **A** — Indicates an Account Number has been 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 will remain the same, and the Event number will increment with each change of state. The first digit of the log number indicates on which box 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. Date format is 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 caller (if available) or the outbound caller.

9. **Name** — This column displays the name of the caller (if available).

10. **Line** — This column indicates on which Box and Line the call came in or out. The format of this information is Box # then Line #. Lines 1-4 are PSTN lines and 5-8 are VoIP lines. Example:

    11 — Box 1, Line 1
    23 — Box 2, Line 3
    16 — Box 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 in length. To assign an account number after a call has been completed, dial ✱ 8 8 + <Account Number> + # .

    If Direct Line Access is enabled on an extension,
    dial ✱ 8 8 + <Account Number> + # .

The data can easily 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

C H A P T E R   5

VOIP INFORMATION

5.1 INTRODUCTION TO VOIP

The TalkSwitch with VoIP lines is a hybrid PSTN/VoIP phone system. The addition of VoIP to the TalkSwitch product line adds the ability to use a broadband IP connection for voice calls.

The VoIP enabled TalkSwitch can be used in many different applications. The most popular configurations are multi-branch, teleworker integration and service provider integration.

5.2 OPTIMIZING YOUR IP NETWORK FOR VOIP

5.2.1 The Broadband Connection
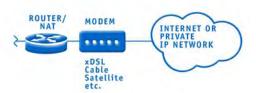
For optimum quality of VoIP calls on your system, we recommend that you subscribe to a high-speed broadband connection from your Service Provider. 'Lite' broadband connections (typically 128 Kbps downstream and 64 Kbps upstream) are not suitable for simultaneous voice and data traffic. The limiting factor for a broadband connection is the upstream bandwidth to the Internet Service Provider.

Example: If you have a 1Mbps ADSL connection from your service provider, it usually means you have an upstream bandwidth of approximately 380 Kbps. Just because the service provider says that you have 380 Kbps upstream bandwidth does not mean that the full 380 Kbps is available at all times. In fact, a conservative approach is to estimate about half of the upstream bandwidth as being available. In this case, you could safely support 2 simultaneous G.711 calls or 4 simultaneous G.726 calls or 10 simultaneous G.729 calls. These estimates do not factor 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 by the link is reduced.
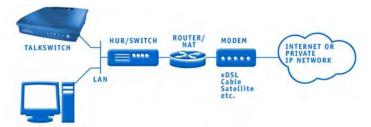
5.2.2 The Router/NAT/Firewall

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 spend all their time looking at the destination IP addresses of the packets passing through and routing them accordingly. In a network that shares both 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 such purposes. These routers are identified as devices that support QoS, VoIP or prioritization. An inexpensive router for the small office market is the Linksys BEFSR81. It can be configured to provide prioritization of the voice traffic on the broadband connection to ensure that voice traffic gets through. It also supports DHCP, UPnP and VPN Pass Through.

TalkSwitch uses UPnP (Universal Plug and Play) to automatically configure port forwarding VoIP ports, and receive updates whenever the WAN IP address changes so that it can properly update SIP messages.

5.2.3 The Local Area Network

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 to ensure that data traffic doesn't compete with voice traffic. A router is required to separate voice and data traffic (different subnets) successfully.

Example: The Linksys BEFSR81 router has a built-in 8 port switch which is perfect for a LAN supporting both TalkSwitch units and computers. If you're not certain whether your equipment is an Ethernet switch, contact the manufacturer.

5.3 FAQ

How does VoIP work?

Voice over IP, or 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 in the same way as other data, 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 can sound as good or even better than regular phone calls. Voice quality is affected by numerous issues, including how it is compressed for delivery over the Internet, decompressed and how packets are processed. A few steps can help ensure call quality.

1. **High-speed connections on both ends of the call**
   Good quality VoIP is not possible over a dial-up modem connection. A broadband connection is required to support more simultaneous VoIP calls. Please visit our site at **www.talkswitch.com/voip/voip_test.php** to determine what your Internet connection can support.

2. **Choose the right CODEC for your location**

TalkSwitch supports 3 CODECs (compression algorithms). The default CODEC is G.729 using approximately 25 Kbps bandwidth upstream and 25Kbps bandwidth downstream for each call. G.729 provides very good call quality while minimizing bandwidth usage.

The G.726 (32Kbps) 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 50Kbps bandwidth downstream for each call.

The G.711 CODEC provides the best voice quality. The trade-off is the bandwidth requirement. G.711 calls typically requires up to 100 Kbps bandwidth upstream and 100 Kbps bandwidth downstream.

If the power goes out, does the VoIP network stay up?

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 regular phone lines.

What happens to VoIP if the IP network fails?

If the connection to the IP network is lost, it will not be possible to make VoIP calls. Calls can still be placed over the regular phone lines.

Can a firewall prevent VoIP calls from passing through?

The purpose of a firewall is to control what kinds of traffic enter and leave your network. The VoIP enabled 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 default 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 one 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 two-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 SIP as its protocol of choice for signalling. SIP 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. SIP just initiates, terminates and modifies sessions.

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.

What is a VPN? Can a VPN help me to carry data securely over the Internet?

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. A VPN provides security by using encryption/decryption. Using a process called 'tunneling', private data is encrypted and then encapsulated before being sent across the network. These packets of information are sent to their destination. Only those with the correct protocol information are able to enter a company's network.

A VPN is very useful for providing employees with remote access to the company network without compromising security. Encryption can have a negative impact on the call quality, as the overhead on the network connection is increased. For example, IPSec adds approximately 10 per cent 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 LAN (Local Area Network) 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 1 IP address but you have more than 1 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).

Note that addresses of the form192.168.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 as this prevents destinations from becoming unreachable for periods of time when the IP address changes. But since this can be an expensive option, the VoIP enabled TalkSwitch supports the ability to handle Dynamic Public IP addresses for the Internet connection. 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 is a SIP Proxy and Registrar?

SIP's basic architecture is client/server in nature. The main entities in the SIP server are the Proxy, Redirect Server and the Registrar.

The VoIP enabled TalkSwitch has the SIP Server fully integrated for easy deployment and maintenance of your VoIP network. If you are deploying a multi-site configuration which can comprise of VoIP TalkSwitch units, certified 3rd party gateways and certified 3rd party SIP phones, one VoIP enabled TalkSwitch can act as the SIP Proxy and Registrar. The other SIP 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, gateway or phone), it will register itself with the VoIP TalkSwitch that is configured as the SIP Server so that calls can be directed to your location.

When you initiate a VoIP call, your device (TalkSwitch, gateway or phone) will contact the TalkSwitch that is configured as the SIP 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 SIP Server is no longer involved in the call.

If the location of the SIP Proxy and Registrar has a Dynamic IP Address, you have to use Dynamic DNS since the SIP 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 domain name — FQDN (which does not change) to locate a device with a Dynamic IP address. There are companies on the Internet which can provide you with a DNS address,

even if your IP address changes. For example, www.dyndns.org or hn.org or www.no-ip.com. You should consult with them about 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 SIP Server resides and where the public IP address is dynamic. All other locations will point to this SIP 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 take no longer than 15 minutes to propagate throughout the DNS system. It is possible that propagation could take longer under extenuating circumstances. Propagation shouldn't take much more than an hour at the very 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 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 the number after the colon, 5060 in this example.

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 friend in an apartment building at a specific address. You need to specify the apartment number as well as the building address, so that the mail can reach the proper final destination.

HELP AND TROUBLESHOOTING

If you are having problems with the configuration or operation of TalkSwitch, please check this section. If you don't find the solution to your problem here, try checking the FAQ section located in the Support area at our website: **http://global.talkswitch.com.**

If you cannot resolve your problem and require further assistance, you can do one of the following:

1. Contact your TalkSwitch dealer.

2. Visit the TalkSwitch Support website and select your location, for additional information and help on TalkSwitch usage: **http://global.talkswitch.com.**

3. Email your TalkSwitch Technical Support team at the address provided on the website.

4. Call your regional TalkSwitch Technical Support specialist at the telephone number provided on the website.

Please have the following information available when you call:

   a) Your e-mail address and telephone number.

   b) The serial number located on the bottom of TalkSwitch.

   c) Your date of purchase.

   d) The number of telephone lines that you have connected to TalkSwitch.

   e) The telephone company services that you are using with TalkSwitch.

   f) Your PC's operating system (e.g. Windows 95, 98, ME, 2000, XP).

Troubleshooting

This section of the manual is designed to help you quickly resolve problems with TalkSwitch. The problems are organized in the following manner.

1. Problems that can occur when installing the configuration software.

2. Problems that can occur during Configuration;

3. Problems that can occur while using TalkSwitch;

4. Problems that can occur with 2 or more networked units .

Before examining the problems, review these common installation checks.

1. Check that your telephone cables are connected to a wall port at one end and TalkSwitch at the other. Ensure the devices you are using as TalkSwitch extensions (phones, faxes, etc.) are plugged into the extension ports located on the back panel of TalkSwitch. (Do not plug your extensions directly into a wall port unless that port is wired to a TalkSwitch extension port.)

2. Check the Serial, USB or LAN connection between TalkSwitch and your PC.

3. Ensure that TalkSwitch's power adapter is plugged into a working power outlet and that TalkSwitch is turned on. Please make sure that you are using the power adapter that was provided with TalkSwitch. This is a unique adapter and it cannot be substituted with any other adapter. If your adapter appears to be defective, please contact your TalkSwitch dealer to order the proper replacement adapter.

4. If networked, ensure that you have the latest firmware loaded on all units. In the TalkSwitch software, select 'Configure', then on the 'About TalkSwitch' page, click on the 'View Details' button in the upper right corner of the screen. All units listed should have the same firmware version.

Diagnostic codes for the lights flashing on the front panel of TalkSwitch

All line lights flashing simultaneously: The Unit ID of this TalkSwitch is in conflict with another TalkSwitch unit on the same LAN. Make sure you assign a different Unit ID to each TalkSwitch unit. See section 2.5.3 for more information on changing a unit ID.

Line 2 and 3 lights flashing simultaneously: There is a file system error. To fix the problem, reboot TalkSwitch. It will do a file system check and fix any problems encountered. If this does not resolve the problem, contact your TalkSwitch dealer.

Line 1 and 4 lights flashing simultaneously: There is a voice mailbox error. To fix the problem, reboot TalkSwitch. It will do a file system check and fix any problems encountered.

Line 3 and 4 lights flashing simultaneously: Firmware not loaded or corrupted. To fix the problem, reload the TalkSwitch firmware. Make sure you update with the last version you updated with. Do not use an old version off the install CD if you have updated your system.

Line 1 and 4 then Line 2 and 3 lights flashing alternately: System prompts not loaded or corrupted. To fix the problem, reload the TalkSwitch firmware. Make sure you update with the last version you updated with. Do not use an old version off the install CD if you have updated your system.

Problems that may occur during Configuration

Problem: I am not able to "Retrieve settings" from my TalkSwitch.

Solution: Try the following solutions, in order, until the problem is resolved.

- Ensure TalkSwitch is turned on and plugged into a working power outlet. Then select "Retrieve settings", under the File menu in the configuration software.
- Make sure the serial or USB cable connecting TalkSwitch and your PC is securely attached to the ports. Select "Connect", under the TalkSwitch configuration window.
- From the Control Center section of the TalkSwitch software, choose 'PC Connection' and then select COM Port. Ensure that the configuration software is using the correct port to communicate with TalkSwitch. Then select"Retrieve settings", under the File menu in the configuration software.
- Try rebooting TalkSwitch (turn the power off and then back on). Then select "Retrieve settings", under the File menu in the configuration software. This will often correct any miscommunication that may have occurred between TalkSwitch and your PC.
- Make sure you do not have any communications programs running at the same time you want to use the TalkSwitch configuration software. These can include Palm Pilot, Hot Sync, TalkWorks, digital camera software or others. These programs tend to hold on to COM ports, making them unavailable for other programs.

Problem: I am unable to configure TalkSwitch using a touch-tone phone and the Keypad Commands.

Solution: Try the following solutions, in order, until the problem is resolved.

● TalkSwitch cannot be configured using a phone that is sharing a line with it.

● If you are trying to configure TalkSwitch using an Out-of-Office phone, you must first enter Command Mode. Dial **#** at the auto attendant If you have enabled password protection, you must also first enter the correct password.

● If you are trying to configure TalkSwitch using a Local Extension, make sure the extension has not been given 'Direct Access' to your telephone lines. If so, you will have to dial 'Recall' to receive internal dialtone.

● If the TalkSwitch configuration software is open, you cannot configure TalkSwitch using touch-tone keypad commands. If the software is closed and you continue to get the prompt "I'm sorry, TalkSwitch is currently being configured." Try rebooting TalkSwitch by turning the power off and then back on.

Problems that may occur while using TalkSwitch's features

The Auto Attendant:

Problem: The Auto Attendant does not play when calls come in.

● Make sure you have recorded the Auto Attendant message.

● Check your Call Handling settings for the 'Auto Attendant answers' number of rings. It may be that the number of rings is set too high.

● Check the mode TalkSwitch is currently running. It is possible that you are in Mode 2 and have not configured TalkSwitch properly for this mode. Check what you have programmed for Mode 2 in the Telephone Lines section of the TalkSwitch software under 'Call Handling'.

● The line is not connected properly to TalkSwitch. Check to see that the line light associated to the line is flickering. If not, try replacing the phone cord.

Problem: The Auto Attendant message is broken up or very faint.

● Adjust the Line Controls volume setting located in the Audio Controls tab in the Options folder of the configuration software.

● The quality of the microphone in the telephone handset that you are using to record your Auto Attendant also affects the quality of your recording. Try recording the Auto Attendant using another extension/telephone.

Problem: The Auto Attendant answers calls before any of the extensions ring.

This is a problem for those TalkSwitch users who wish to have their Local Extensions ring before the Auto Attendant picks up incoming calls.

- Check the 'Extensions to ring' column in the Telephone Lines tab in the 'Call Handling' folder. The number of rings may be set too low.

Problem: The Auto Attendant is transferring calls to the wrong extension.

- Verify the extension ports your devices (phones, faxes) are attached to.

Problem: The Auto Attendant answers calls, but does not transfer them to the extensions.

- Make sure your extensions are plugged into the extension ports on the back panel of TalkSwitch (instead of your telephone wall ports).
- Check your settings in the Auto Attendant page. Are the options pointing to the correct extensions?

Music on Hold:

Problem: Callers hear the 'One moment please' message and then total silence when the Auto Attendant transfers their call to an extension.

- You have enabled the Music on Hold feature, but have not attached an audio device to the MUSIC port on TalkSwitch.
- There is a problem with the music source. Check all connections and power.

Problem: Callers hear only silence when put on hold at an extension.

- You have enabled the Music on Hold feature, but have not attached an audio device to the MUSIC port on TalkSwitch.
- Make sure you use the 'Recall' or 'Link' button to put callers on hold. If you use the 'Hold' button on a regular phone, callers will be placed on hold at the phone itself, and not through TalkSwitch.

Problem: Sometimes callers do not hear Music on Hold.

- Make sure you have a music source connected to the MUSIC port at the back of each TalkSwitch unit connected to the LAN.

Call Routing with Local Extensions and Home Phones (Transfer and Intercom):

Problem: Callers are disconnected when transferring calls from one extension to another.

- It is possible that if you are using the hookswitch to put callers on hold rather than a Recall button, you may be holding the hookswitch down too long, and, as a result, are actually hanging up on the caller.

Problem: 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. This option is located on the Telephone Lines page in the 'System Information' folder of the configuration software.

Problem: Unable to place intercom calls from a Local Extension.

- If the Local Extension has been configured for Direct Access to your telephone lines, you need to press 'Recall' before intercom calling.

Answering and Fax Machines:

Problem: The answering machine and/or fax machine picks up calls before they can be answered by telephone extensions.

- Check your settings in the Telephone Lines tab in the 'Call Handling' section in the configuration software. The problem may be that your fax machine and/or answering machine extension is 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.

Problem: Incoming faxes are not automatically detected and routed to the fax machine.

- On the Auto Attendant tab of the configuration software, check to ensure that you have chosen the appropriate extension for the field labeled 'if a fax call is detected'.
- Not all fax machines emit a CNG tone that TalkSwitch can detect and route. Therefore, it is wise to consider incorporating an additional method of fax routing. This may include the use of a distinctive ring number or giving the caller the option to select the fax extension. See section 3.3.1.7.

- 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 are using the 'User does not make a selection' option to route incoming faxes, the fax call may 'time out' before it reaches your fax extension. If your Auto Attendant is lengthy, you may want to shorten it or use another fax routing option.
- The level of the Auto Attendant is too loud and as a result is overpowering the CNG tone, affecting detection. You may want to lower the volume level of the Auto Attendant.

Other Possible Local Extension Problems:

Problem: I am unable to access TalkSwitch Lines with a Local Extension.

- Check your 'Direct Line Access' settings on the Local Extension page under 'System Information' of the configuration software. If a Local Extension has not been given 'Direct Access' to your telephone lines, you must dial a Line Hunt Group access code.

Problem: My PC (Internal or External) Modem cannot dial out.

- If the extension to which your Modem is attached has not been given 'Direct Access' to telephone lines, you will have to include a Line Hunt Group access code before the number you are dialing.

Problem: 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 that have mechanical ringers. These phones are too much of a load for TalkSwitch and should not be used as an extension.
- Make sure the telephone cord you are using between the phone and TalkSwitch is working properly.
- Check to make sure the ringer is turned on for your phone.

Problem: 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. This is the reason the Auto Attendant may not pick up the disconnect signal right away when a caller hangs up. Because the Auto Attendant will receive no response after playing its message, the call will fall through to your settings for the 'User does not make a selection' option associated with that Auto Attendant.

Problem: When I try to access voicemail, I hear a busy tone.

• Check your 'Direct Line Access' settings on the Local Extensions screen under 'System Information' of the configuration software. If a Local Extension has been given 'Direct Access' to your telephone lines, you must first press Recall before dialing the mailbox. The same applies for dialing other extensions to initiate intercom calls.

Problems that may occur with 2 or more TalkSwitch units connected to the same LAN.

Problem: 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 there is dialtone, then check your LAN connections. On the rear panel, the top LED of the 3 LEDs next to the LAN port should be on. If there is no dialtone, then make sure the unit is powered on.

Problem: 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, then 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 assigned. Follow the instructions accordingly. You can dial **#**00**#** at any time to check the Unit ID of the unit you are connected to. See section 2.5 for more information.

Problem: 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 then 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.

Problem: Calls are not always reaching my voicemail.

• In extreme cases, when the system is in high use, all channels across the LAN might be in use so calls cannot be transferred to extensions or voicemail. If this situation appears to be happening frequently, you may want to consider optimizing the way the system uses lines. In the Line Hunt Groups section of the software, we recommend that you enable the 'Hunt lines on same unit first' option. Also, ensure that frequently used extensions are connected to units with ample lines out, to reduce LAN traffic.

Example: You have 2 TalkSwitch units, 6 lines and 12 employees. The 6 lines have the Hunt/Rollover service from the phone company.

Suggested setup:

Plug the 4 highest traffic lines into unit 1. Also connect the 8 highest phone usage employee extensions to unit 1. Next, connect the remaining lines and extensions to unit 2. Make sure you leave the 'Hunt lines on same unit first' option enabled in the Line Hunt Group section of the software. This arrangement keeps most of the traffic on unit 1 and any outbound calls made from any extension will usually go out on the same unit the extension is connected to. The same will apply for inbound calls forwarded to Remote Extensions. The call will be forwarded out on a line belonging to the same unit that the call came in on. Only if a line on that unit is unavailable will the call be forwarded out on a line on the other TalkSwitch unit.

Problem: Calls across the LAN are lower in volume

• Check the line gain levels on the Audio Controls page located in the 'Options' folder in the TalkSwitch configuration software. You may need to boost the Line gain levels on one or more lines to compensate for high loss lines. Take note that if these controls are set improperly and your lines are problematic, you may hear an echo when calls are made across the LAN.

Problems that may occur with VoIP

Problem: I hear 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, SIP gateway or SIP phone is registered with.

• Check that TalkSwitch has all proper settings in the IP Configuration and VoIP Configuration pages

• Contact TalkSwitch for help at Technical Support.

Problem: Callers complain the sound is distorted or choppy.

• Your broadband connection may not have enough upstream bandwidth to support many simultaneous VoIP calls. In the 'Options' section under '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.

Problem: 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.

For more VoIP information, see Chapter 5: VoIP Information.

USING TALKSWITCH WITH TELEPHONE COMPANY CALLING SERVICES

The TalkSwitch is compatible with some calling services offered by your local telephone company. Use of telephone company services may require subscription; and some services may not be available – please consult your local telephone service provider for details.

Caller Display:

(Also known as: "Caller Identity", "Identicall", …)

Caller Display provides the caller's telephone number and name. TalkSwitch receives the caller's identity. When the incoming call is transferred to a local extension, the caller's identity is also sent to the extension. If a call is transferred from one extension to another, the caller's identity will also be transferred. The system can also use the Caller Display time stamp to set its internal clock.

Distinctive Ringing

(Also known as: "Call Sign", "My-Ring",)

TalkSwitch can identify the distinctive ringing pattern assigned to a telephone line's second or third directory number. An incoming call with a distinctive ring can be configured to receive different call handling treatment. A common use of distinctive ringing is to share a fax number with an ordinary telephone number using the same telephone line. When the

fax distinctive ring is detected the system can be configured to route the call to an extension dedicated to a fax machine. Note if distinctive ring is present on an incoming call, but not activated in the line configuration it will be subject the same call handling as the telephone line's primary number. Note that some telephone companies may assign different uses to distinctive ringing patterns, such as signalling a Call Back.

Multi-Line Service / Hunting Facilities

Most telephone companies offer line hunting service, when more than one line is provided. This feature is recommended when more than one line is used. Using a single telephone number, incoming calls "hunt" to find a free telephone line to the PBX. Note which order the telephone company selects telephone lines for incoming calls, and arrange the Line Hunt Groups to search for available line in the opposite order, to minimize glare (collision between incoming and outgoing calls). An alternative to multi-line hunting is to use Call Forward on Busy, if available.

Voicemail

(Also known as "Call Answer", "Messaging")

This telephone company service is similar to the voicemail provided by TalkSwitch. Although the TalkSwitch provides an integrated voicemail, the telephone company's voicemail service may be useful in situations where all telephone lines are busy.

Three Way Calling/Conference

This telephone company service allows a third party to join a call. This service is useful when used with the TalkSwitch "Same Line Connect" and Hunt Group 80 features to avoid using two telephone lines when two parties are on the PSTN – such as an outgoing DISA call.

The use of the telephone company Three-Way Calling feature is not recommended where ARS and Toll Restriction are used, as the TalkSwitch can not control the telephone company's routing of the call to the third party.

The TalkSwitch may not be compatible with all telephone companies' implementation of Three-Way Call/Conference service. The feature should be compatible with the following telephone companies: British Telecom, Belgacom, Telkom SA, SwissCom, MaltaCom.

Centrex Transfer

This is a Centrex service offered by a few telephone companies, whereby on releasing a three-way call the other two parties continue to be connected. If available this feature can be used with the "Same Line Connect" feature of TalkSwitch.

Application Notes – Telephone Company Services

Remote Extensions – TalkSwitch Remote Extension feature is designed to work with major local telephone and mobile service providers. The feature may not function correctly with some telephone companies and mobile operators, especially international telephone numbers and mobile phones when roaming internationally. When configuring remote extensions to PSTN telephone numbers it is recommended that they are tested, when the remote extension is idle, busy and does not answer.

Prompted Ring Back – This is a telephone company feature which, when the called line is busy, provides the caller with an announcement offering a Call Back when the called line is free. This telephone company feature is not compatible with Remote Extensions. It is recommended that the remote extension's telephone company is requested to disable Prompted Ring Backs on the remote extensions telephone line.

Re-Answer Supervision – This is a feature offered by some telephone companies where the telephone can be placed on-hook for a period, typically 30 seconds, during which time going off-hook again will reconnect the call to the originating party. It is recommended that re-answer supervision is not used on lines connected to TalkSwitch.

Colour Me Tones – This is a feature offered by some telephone and mobile operators where Ringing Tone is replaced by the subscriber's choice of music or an announcement. It is recommended that this service is not used on Remote Extensions

Call-Back – Telephone companies may offer various services which result in automated call back. The use of such services is not recommended on telephone lines connected to the TalkSwitch.

Telephone company calling feature names may be registered trademarks of their respective owners.

QUICK COMMANDS
AND TOUCH-TONE FUNCTIONS

These Touch-Tone Quick Commands can be dialed from a Local Extension or an outside telephone at the Auto Attendant. To enter Command mode, dial **#** at a Local Extension or at the Auto Attendant. If you are requested to enter a password, enter the Administrator password. Next, enter any command followed by the **#** sign.

Command	Action
00	TalkSwitch tells you the Unit ID of the unit you are connected to.
0x	Assign Unit ID 1 - 4 to TalkSwitch (x = 1 to 4)
30	TalkSwitch tells you which mode it is currently using.
3x	Switch Modes (x = 1 or 2)
4x	Record Auto Attendant (x = 1 to 9)
04x	Erase Auto Attendant messages (x = 1 to 9)
5x	Playback Auto Attendant messages (x = 1 to 9)
61 - 64	Record Auto Call Back Announced Messages 1 - 4
65	Record Prompted Call Back Announced Message
061 - 064	Erase Call Back Announced Messages 1 - 4
065	Erase Prompted Call Back Announced Message
71 - 74	Playback Auto Call Back Announced Messages 1 - 4
75	Playback Prompted Call Back Announced Message
90	Set to Serial connection mode (default)
91	Set to USB connection mode
2xx 8<phone number>	Enter Remote ext. (2xx) phone number
2xx 980 0	Enter Remote ext. (2xx) Turn off "Same Line Connect"

2xx 980 1	Enter Remote ext. (2xx) Turn on "Same Line Connect"
2xx 9 <Hunt Group>	Enter Remote ext. (2xx) "Connect using:" Hunt Group (Hunt Group = 9, 81-88)
1xx 61	Enable 'Do not Disturb' at the Local Extension
1xx 60	Disable 'Do not Disturb' at the Local Extension

Touch-tone features dialed from a Local Extension

Command	Function performed	Notes
1x1-1x8	Intercoms a Local Extension	x represents the Unit ID
2x1-2x8	Dials a Remote Extension (associated with an external phone number).	x represents the Unit ID
250-299	Speed dial for VoIP calls. TalkSwitch will hunt the VoIP ports, go off hook then dial the digits entered.	
300-309	Dials an Extension Ring Group. 10 in total for the entire system. Mapped to local extension(s), with specific fall through options.	
<recall> if connected to a caller	Puts caller on hold.	After putting a caller on hold, you receive internal dialtone
<recall> at internal dialtone	It retrieves the most recent caller put on hold.	To retrieve the caller on hold for the most time, dial 7 at internal dialtone.
<recall> at external dialtone	Switches to internal dialtone. Used mainly when an extension has Direct Line Access enabled.	

(*)1x1-1x8 (*)2x1-2x8 (*)4x0-4x9	Dials into a local ext. voice mailbox. Dials into a remote ext. voice mailbox. Dials into a general ext. voice mailbox.	To transfer a caller to a mailbox, press 'recall' to place a call on hold, then (*)1x1-1x8, (*)2x1-2x8, or (*) 4x0-4x9 and hang up. x represents the unit ID
(*)(*)1x1-1x8 (*)(*)2x1-2x8 (*)(*)4x0-4x9 (*)(*)(#)	Internal voicemail retrieval/access. (*)(*)(#) accesses the mailbox associated to that extension.	To retrieve voicemail. x represents the Unit ID
<recall>4	Completes a transfer and returns to internal dialtone.	Action performed after putting a caller on Hold then dialing an intended destination.
(*)0	External P.A. Access	To make an announcement through the attached external P.A. system.
<recall>5	Disconnects current caller and reconnects immediately to the last caller put on hold.	
(*)500-509	Call Parking Spots — 10 in total for the entire system.	Press 'recall' to place a call on hold, then assign a parking space (*)500-509.
(*)510	Auto Park a call.	Press 'recall', then (*)510 to place a caller in the next available parking spot (park orbit).
(*)(*)500-509	Parked Call Retrieval	To pick up a parked call.

<recall>6	Conference Call for 3 parties	Press 'recall' to put the first caller on hold, dial another extension or external number, then press 'recall' 6.
＊60 **＊**61/62	Toggles 'Do Not Disturb' at the current extension. Enables/Disables 'Do Not Disturb' at the current extension.	User will hear confirmation. Reminder is played every time the extension is picked up when enabled.
＊7 and 1xx	Call Pick up — Extension Specific	To pick up a call ringing at another extension at anytime including after the Auto Attendant answers.
<recall>7	Hold Retrieve — used for retrieving any calls placed on hold at that extension.	Or press 'recall', then 7 to cycle through on a first in / first out basis.
80	Accesses Same Line Connect when transferring an outside caller to another outside number.	Put the caller on Hold, dial 80, and then dial the outside number. Hang up to complete the transfer. Can only be used when the line has 3-Way Calling/Conference or Transfer (Centrex/Plexar) service.
81-88, 9	Line Hunt Groups used to access a line.	
＊80	Play music source through PA output.	To toggle on or off music through the PA port. Useful for office background music.
＊9	Call Pick up	To pick up a call ringing at any other extension.
＃	Enters command mode.	

> **Note:** If you have Direct Line Access enabled at your Local Extension, then you must first press 'recall' before using most of the functions listed.

Touch-tone features dialed at the Auto Attendant message from outside phones

Digit dialed	Function performed
0	User Defined — An Extension, Ring Group, Auto Attendant, Voice mailbox, Announcement or perform no action.
1x1-1x4 1x1-1x8	Dials Local Extensions. x represents the Unit ID number.
2x1-2x8	Dials Remote Extensions associated with an external phone number. x represents the Unit ID number (1-4).
250-299	Speed dial for VoIP calls. TalkSwitch will hunt the VoIP ports, go off hook, then dial the digits entered.
300-309	Dials Extension Ring Groups — 10 in total for the entire system. Mapped to Local Extension(s), with specific fall through options.
1,2,3,4,5	User Defined — An extension, Ring Group, Auto Attendant, Voice mailbox, Announcement or Perform no action.
6	Enters Call Back configuration.
7(xxx...)	PBX Transfer — A recall hook transfer with disconnect after dialed. Usually used only in systems connected to another PBX.
81-88, 9	"Line Hunt Group access code for DISA for — password protected at the Auto Attendant.
80	DISA using Same Line Connect. Requires 3-Way Calling/Conference on the line.

✱1x1-1x8 ✱2x1-2x8 ✱4x0-4x9 4x0-4x9	Enter Internal voicemail to leave a message or listen to an announcement. If you press 8 during the greeting, you can access messages. x represents the Unit ID number. Callers can directly also dial 400 series mailboxes.
✱✱1x1-1x8 ✱✱2x1-2x8 ✱✱4x0-4x9	Internal voicemail retrieval/access. x represents the Unit ID number.

REGULATORY INFORMATION

Product Regulatory Information

TalkSwitch complies with:

- EN 60950-1:2001 "Information Technology Equipment - Safety".
- CISPR 22 "Information Technology Equipment - Radio Disturbance Characteristics", (Class B),
  with Amendment 1:2000 and Amendment 2:2002
- EN55022:1998 "Information Technology Equipment - Radio Disturbance Characteristics", (Class B),
  with Amendment A1:2000 and Amendment A2:2003
- EN55024:1998 "Information technology equipment – Immunity Characteristics",
  with Amendment A1:2001 and Amendment A2:2003
- EN61000-3-2:2000, "Electromagnet Compatibility (EMC) - Part 2-3: Limits for Harmonic Emissions",
- EN61000-3-3:1995, "Electromagnet Compatibility (EMC) - Part 3: Voltage Fluctuations and Flicker",
  with Amendment A1:2001

United Kingdom and Ireland

$C\mathcal{E}$

The product models listed below are intended for use in European markets indicated, comply with the essential requirements of the European "Radio and Telecom Terminal Equipment" (R&TTE) directive 1999/5/EC, and are for connection to analogue telephone lines (PSTN).

TalkSwitch models are intended for use in the following countries:

Product Suffix	Intended Markets
-UK	Great Britain and Northern Ireland [GB], and the Republic of Ireland [EI]

For Declaration of Conformity please contact local distributor
(see: **http://global.talkswitch.com**)

Disposal

At the end of the product's life, please ensure disposal is in compliance with local regulations for electrical and electronic waste.

TALKSWITCH ONE-YEAR WARRANTY

TalkSwitch Limited Warranty

TalkSwitch, a division of Centrepoint Technologies ("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) an event where the power unit has been connected to a voltage supply other than that for which it is specified; (d) damage from corrosive environments and moisture; (e) static discharges; (f) high voltage surges from external power sources such as power line, telephone line or connection to other equipment; (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.

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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.

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 Centrepoint Technologies Inc. ("TalkSwitch") will not be able to provide you with any kind of warranty service.

To be eligible to obtain warranty service during the warranty period the Customer must provide proof of the date of purchase, serial number, and obtain warranty return instructions from the Authorized Reseller, or the TalkSwitch Technical Support Centre listed on **http://global.talkswitch.com.** To obtain warranty service the Customer is responsible for returning the TalkSwitch to the Authorized Reseller or the TalkSwitch Warranty Centre in suitable packaging. Any applicable duties or taxes for the return of repaired or replacement product are the responsibility of the Customer. Replacement or repaired product is returned free of charge (excluding duties and taxes) is warranted to be free from defects in material or workmanship for the remainder of the limited warranty period of the originally purchased Product.

SPECIFICATIONS

PSTN Analogue Telephone Line Interface:

Line Type	Loop Start
Line Impedance	BT SIN 315 for Terminal Impedance [UK]
	ETSI TBR21 [Ireland]
	(options for 600 ohms, 900 ohms and TBR 21)
Ringing Detection	25 Hz
Connector Type	RJ-11 (2 wire telephone cable with BT style plug provided for use in UK)
Caller ID Support	All Lines
Caller ID Alert	LR+DT-AS and RP-AS
Caller ID data	V.23
Distinctive Ring Support	Yes
	# 1: one burst per ring cycle
	# 2: three bursts per ring cycle

Extension Interface

Loop range	600 ohms
On hook/Off hook voltage	35 V
Ringing voltage	90 V rms
Ringing frequency	25 Hz
Connector type	RJ-11, 2 pin
Extension Impedance	BT SIN 315 for Network Impedance [UK]
	ETSI TBR21 [Ireland}
Signaling type	DTMF only

Supervisory Signaling	Loop Calling
Recall	Yes Timed Break Recall
Caller ID Support	all extensions
Caller ID Alert	LR+DT-AS [UK]
	RP-AS [Ireland]
Caller ID data	V.23
Message Waiting Indicator	MWI message & stutter dial tone

System

PSTN Analogue Telephone Lines:	4 (models with LAN port), 2 (models without LAN port)
Extensions	8 (models with LAN port), 4 (models without LAN port)
Serial Port	RS 232, 9 pin
USB Port	USB 1.1
LAN Interface	10BaseT, RJ-45 (models with LAN port)
Voicemail	30 minutes on-board, 100 messages per box
Voice Mailboxes	Local Extensions: 8 (models with LAN port), 4 (models without LAN port) + Remote Extensions: 8; + General/Guest 10
Voicemail Expansion	1 - 9 hours with expansion card
Auto-Attendant Positions	9 positions each with 6 options each (+ fax detect and timeout)
Announcement File Format:	".wav" format, 8 bit, mono, u-Law.
Music on Hold input	1/8"/3.5mm mono
P.A. output	1/8"/3.5mm mono
Power Source	220-240 V, 50Hz, adaptor supplied
Power Switch	Push Button
Dial Tone generators	12
Ring Detectors	4
Power Fail Pass-Through	Line 1 to Extension 4 (1x4)
Dimensions (TalkSwitch Unit)	(H) 2.031"/51.6mm, (W) 7.992"/203.2mm, (D) 8.401"/213.4mm
Weight (TalkSwitch Unit)	22.75 oz/0.645kg (models without LAN port), 31.21 oz/0.885kg (models with LAN port), 31.74 oz/0.9kg (VoIP enabled TalkSwitch)
Power Adaptor (220-240 V model)	20.31 oz/0.576kg

Specifications subject to change

TALKSWITCH
AND POWER INTERRUPTIONS

TalkSwitch settings and configurations:

In the event of a power failure, your TalkSwitch settings and configurations will not be lost.

TalkSwitch's feature settings (configuration) are stored in an EEPROM. An EEPROM is a form of non-volatile memory. Non-volatile memory means that your feature settings (configurations) remain in TalkSwitch's memory despite interruptions in the power supply.

Calling Features:

In the event of a power failure, no TalkSwitch features or capabilities (including VoIP) will be available other than:

• Local Extension, 114, 124, 134 and 144 can make or receive calls on Line 1 of their respective units.

In the event of a power return, all TalkSwitch 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 TalkSwitch's 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, TalkSwitch will check the time and determine the mode it should be in according to the configuration and set itself to that mode.

To check the current mode of operation, dial **#** at a Local Extension to enter command mode, then enter a password if required. Next, dial 30 then **#**. TalkSwitch will report the current mode.

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GLOSSARY

* (Star/Asterisk Sign)
The character on the bottom left of your touch-tone telephone keypad.

 # (Hash Sign)
The character on the bottom right of your touch-tone telephone keypad. It is also commonly called the number sign.

3-Way Calling/Conference
A service provided by telephone companies which allows you to connect parties on three different telephone lines together for one conversation.

3-Way Conference Call
Connecting three parties together for one telephone conversation. TalkSwitch offers its own 3-Way Conference feature and also allows you to use the 3-Way Conference (Calling) service provided by local telephone companies.

Analogue
All sound actually begins in the form of sine waves and must be in this form to be heard by the human ear. Analogue devices (phones, answering machines, etc.) transmit and receive sound in the form of 'sine waves'. The telephone lines in your home are also designed to carry analogue signals (sine waves). TalkSwitch is an analogue device and will work with any other analogue device or telephone system.

Auto Attendant Message
TalkSwitch's electronic attendant. A self-recorded message that greets and directs callers to select an extension number based on information provided in the message.

Auto Call Back
Allows you to activate Call Back by dialing one of TalkSwitch's telephone numbers, letting the call ring at least once and then hanging up.

Busy Tone
    Tone used to indicate that the called party is busy.

Call Back
    A TalkSwitch feature which, once activated, instructs TalkSwitch to call you back at a preset Out-of-Office telephone number. You can then access TalkSwitch's DISA — Direct Inward System Access feature. TalkSwitch offers two ways to activate Call Back — Prompted and Auto.

Call Forward
    A TalkSwitch feature which forwards calls to one of 8 or more Remote Extensions.

Caller ID
    A service provided by telephone companies which transmits information about incoming calls (callers name and telephone number).

Call Waiting
    A service provided by telephone companies which allows 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. A type of PBX service offered by the telephone company on a 'pay per service' basis. All call switching occurs at a local telephone station instead of a company's premises; Service may not be

available from local telephone company.

CNG
    A 'chirping' tone emitted by fax modems and most fax machines to initiate contact with another fax machine/ modem.

COM Port
    Communications Port. Internal virtual pathways used to transmit data to and from a PC's exterior Serial Port and software running on the PC. The majority of PCs have four Com Ports: COM1, COM2, COM3, and COM4. Most software programs gain access to the COM Ports via system interrupts (IRQs) and only two IRQs are reserved for the four ports. If you have two devices using COM Ports that share an IRQ, you cannot use the devices simultaneously. Typically, COM1 and COM3 use IRQ4, while COM2 and COM4 use IRQ3.

Communications Software
    Software that makes it possible to send and receive data over telephone lines via modems.

Compression (Data)
    Storing information in a format that requires less space than usual. Compression saves transmission time, capacity and storage space.

Computer Telephony Integration
    Refers to systems that enable a computer to act as a call

center, accepting incoming calls and routing them to the appropriate device or person. Today's CTI systems, like TalkSwitch, are quite sophisticated and can handle all sorts of incoming and outgoing communications, including phone calls, faxes, and Internet messages.

Conference Call
Connecting three or more parties together for one conversation.

Configuration
The way a system is setup. For TalkSwitch the configuration is the collection of values or options you have chosen for each of TalkSwitch's features. Together, the settings for each of these features (The configuration) determine what TalkSwitch will do. TalkSwitch can be configured using the Configuration software or Telephone Keypad Commands.

Connector
The part of a cable that plugs into a port or interface to connect one device to another. Most connectors are either male or female.

CTI
Computer Telephony Integration.

Dial Tone
Tone used to indicate that equipment is ready to receive dialed digits. Tone is usually constant; and may be

interrupted when features are activated on the line, such as message waiting or call forwarding.

Digit
Any whole number from 0 to 9. The numbers on your telephone keypad can also be referred to as digits.

Digital
Digital devices (phones, answering machines, etc.) convert analogue sine waves into digital signals (a combination of zeroes and ones) for storage and/or transmission over digital lines. The digital signal is then converted back to analogue sine waves, that can be heard by human ears.

Digitize
To translate into a digital form. For example, optical scanners digitize images by translating them into bit maps. It is also possible to digitize sound, video, and any type of movement. In all these cases, digitization is performed by sampling at discrete intervals. To digitize sound, for example, a device measures a sound wave's amplitude many times per second. These numeric values can then be recorded digitally.

Distinctive Ring
A service provided by some telephone companies, where more than one telephone number that callers can dial to

reach the same telephone line — such as a second number for a fax machine. Each Distinctive Ring number has a different ringing cadence. The TalkSwitch can detect one, two, and three ring bursts per ringing cycle, and provide different call handling to incoming calls with each of these Distinctive Ringing cadences.

Driver
A program that controls a device. Every device, whether it is a printer, disk drive, or keyboard, must have a driver program. A driver acts like a translator between the device and programs that use the device.

DTMF
Dual Tone Multi-frequency. The system used by touch-tone telephones. DTMF assigns a specific frequency, or tone, to each key so that it can easily be identified by a microprocessor.

Extension (Local)
A Local Extension is any telephone or analogue device (answering machine, fax, etc.) that is plugged into one of the 'Extensions' ports located on the back panel of your TalkSwitch.

Firmware
Firmware is a combination of software and hardware. More precisely, it is software (programs or data) that has

been written onto hardware (micro-chips). Micro-chips (ROMs, PROMs and EPROMs) that have data or programs recorded on them are called firmware.

Icon
A small picture that represents a program, action or feature in the configuration software. When you click on an icon, that program, action or feature is activated.

Intercom
A dedicated voice service within a specified user environment. TalkSwitch offers intercom capabilities between its Local Extensions.

Interface
Something that connects two separate entities. A user interface is the part of a program that connects the computer with a human operator (user).

Internet
A global network connecting millions of computers. Unlike online service networks (.com, .ca, .net, etc.) which are centrally controlled, the Internet is decentralized by design. Each Internet computer, called a host, is independent.

Keypad Commands
Combinations of digits that are used to configure TalkSwitch from a touch-tone telephone.

Keypad (Telephone)
The push buttons on a telephone.

Key Telephone System
With Key Telephone Systems, access to telephone lines and calling features is controlled internally within each telephone unit. Users can gain direct access to telephone lines, etc. by pressing a key (or button) on their telephone.

LED
Light Emitting Diode. An electronic device that lights up when electricity is passed through it. LEDs are used in displays on electronic devices such as computers. TalkSwitch has five LEDs on its front panel. They show which telephone is in use, whether data is being transferred to or received from an attached PC, and which Mode setting is in use.

Modem
Acronym for MODulator-DEModulator. A modem is a device or program that converts digital information stored on a Computer into analogue signals for transmission over telephone lines and, in reverse, converts analogue signals into digital.

Modes
The 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 Software Configuration package. The Monitor provides a graphical representation of telephone traffic as it flows through the TalkSwitch 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. Access to telephone lines and services are controlled by the PBX. A PBX allows many users (extensions) to share a smaller number of telephone lines and have access to additional call handling benefits (call hold, intercom calling, etc.). TalkSwitch is a 'mini' or 'micro' PBX.

Phone Load
see Ringing Equivalency.

Prompt
Recorded instructions delivered by voice processing units. TalkSwitch provides prompts for you when using such features as Call Back, DISA - Direct Inward System Access and Selective Call Forward. Prompts are also provided when configuring TalkSwitch

using a telephone Keypad and for callers being put 'on hold'.

Recall Button
Timed break recall, used to activate features

Remote Extension
A telephone or mobile number on the public network, that is setup as a virtual extension of the TalkSwitch. The Remote extension is assigned an extension number and provided with a voicemail box.

Ringing Equivalency
(Ringer Equivalency Number [REN]) A number, determined in accordance with the US Code of Federal Regulations, which represents the ringer loading effect on a line. A ringer equivalency number of 1 represents the loading effect of a traditional telephone set ringing circuit.

Ringing Tone
Tone used to indicate that the called party telephone is ringing. Tone is not always synchronized to the actual ringing signal at the called telephone, and may not be the same ringing pattern.

RJ-11
Registered Jack 11 is used to connect single 2-wire telephone lines and used with such devices as telephones, cordless and DECT phones, fax machines and modems.

RJ-45
Registered Jacks 45 is a technical term for a telephone jack that supports 8 conductors and is used generally for Networking applications.

Prompted Call Back
Allows you to activate Call Back by dialing one of TalkSwitch's telephone numbers, letting the Auto Attendant answer and then entering the Call Back settings.

Routing
Determining the path that a message or call is to take over circuits. TalkSwitch offers 'routing' of incoming calls using DSP technology.

Serial Communication
A type of electronic communication that, unlike parallel communication, requires that data bits be sent one after the other. Modems rely on serial communication to send data over telephone lines.

Shortcut Key
A special key combination that causes a specific command to be executed. Typically, shortcut keys combine the Ctrl or Alt keys with some other keys. In Windows environments, for example, Ctrl and C is used as the shortcut key to copy. On PCs, the function keys are also often used as shortcut keys.

SIP

Session Initiation Protocol (SIP). Developed by the Internet Engineering Task Force (IETF), this is a signaling protocol designed to be much more simple and flexible than the H.323 protocol. SIP uses the existing Internet and HTML structures to provide VoIP transmissions. In addition, SIP is more scalable, allowing applications to grow in scope without any concern for the existing architecture. SIP gives network managers the flexibility to select any type of architecture, from a heavily centralized network controlled extensively by a server, to a less concentrated network where intelligence is found at the endpoints that communicate directly with each other.

Serial Port

Serial Ports are physical 'plug-ins'. They allow two devices to exchange information or data. A Serial Cable is used to connect the two Serial Ports together.

SOHO

Acronym for Small Office/Home Office. It's a burgeoning market comprised of solo professionals, part-time work-at-homes, telecommuters and small businesses.

Tel Lines 1, 2, 3, or 4

A port or plug-in on the back panel of TalkSwitch which can accept a RJ-11 plug from either a telephone line or PBX system. A single line telephone cable can be plugged into this port and will become a usable TalkSwitch Line.

Telephone Keypad

The collection of push buttons on a touch-tone telephone.

Telephone Wall Jack

A wall outlet for a standard telephone jack.

Touch-tone telephone

A push-button telephone which emits distinctive (DTMF) tones as different buttons on the keypad are pressed.

Two Line Telephone

A telephone which allows you to make and receive calls on two separate telephone lines.

TX/RX

An acronym for transmit and receive.

USB

Short for Universal Serial Bus, the USB connection supports more bandwidth than a serial or parallel connection.

Virtual Extension

A telephone or other communications device not physically connected to TalkSwitch as an extension, but which provides the same functionality. In general, the term 'virtual' distinguishes something that is merely conceptual from something that has physical reality.

TalkSwitch's Remote Extensions are virtual extensions.

Voicemail (Call Answer)
A service provided by TalkSwitch or telephone companies which gives callers the opportunity to leave a message when your extension or telephone line is busy or unanswered.

VoIP
Voice over IP is a term used to describe the technology of taking voice and then applying algorithms to digitize, compress and carry the data over an IP connection.

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