VRCPA **RingCentral® Page Adapter Configuration Guide**

Rev. 3.1

Valcom's VRCPA RingCentral® Page Adapter is designed to provide access to paging systems from RingCentral hosted Session Initiation Protocol (SIP) voice over IP (VoIP) connections, or through analog trunk or station inputs (FXO/FXS). When paired with RingCentral SIP connections the VRCPA supports Valcom handsfree page units for talkback paging and up to 150 Valcom one-way amplified speakers. The VRCPA can connect multiple units for multi-zone paging.

Access Methods
- RingCentral SIP (Session Initiation Protocol)
- FXS extension
- PBX station extension
- Electronic key system line key
- PBX loop start trunk port
- Standard telephone

Features

- Use with SIP (Session Initiation Protocol) VoIP PBX

- Use with Valcom multi-zone page control unit with FXS or analog station ports (DTMF signaling only)

- Use with Valcom handsfree page units for talkback paging

- Multiple units may be connected for multi-zone paging with all call behind a PBX

- Background music, stereo or mono input; music mutes during page

- Volume controls for background music and tones

- Security code option to restrict page access

- Store & Play feedback eliminator option

- Multiple disconnect options

- Page verification tone

- Customizable Page alert tone

- Drive up to 150 one-way amplified speakers on 8 Ohm output (one-way output)

- Multiple options for programming

- Power Supply included

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The VRCPA RingCentral® Page Adapter is primarily designed to accept input from RingCentral via the SIP interface, but can also accept input from an FXO (loop start trunk) port or FXS (CO station) port. The output from the VRCPA can be Line Level audio, FXO (loop start signaling) or audio transmitted over the IP network using Valcom IP protocol.

Hardware Settings

The selection of which analog input and analog output will be active is set using DIP switches on the rear panel. These selections are not configurable through software. The VRCPA must be power cycled for DIP switch changes to take effect.

Page Input (SW 6)

Two RJ11 jacks are provided for connecting the VRCPA to an analog phone port on a PBX, ATA, etc. One of these ports will be selected as the active port using DIP switch 6. The ports are labeled according to the type of signaling provided. The port labeled FXO is intended to connect to an FXS port (analog station port) and the port labeled FXS is intended to connect to an FXO (loop start trunk) port. The FXS port detects off-hook condition and provides dial tone to the circuit. The FXS port can be used with a standard phone or loop-start trunk port (SW 6 ON). The FXO port is a loop-start trunk port and can be used with a Station Port or FXS on a PBX (SW 6 OFF).

Page Output (SW 1)

The analog output of the VRCPA is accessed by screw terminals labeled One Way Page Out or Talkback Page Out. The One Way Page Out (SW 1 OFF) provides a standard line-level audio tip/ring output. The Talkback Page Out (SW 1 ON) provides FXO signaling (loop start trunk) for use with most multizone page controllers. One of these outputs will be selected as the active output using DIP switch 1.
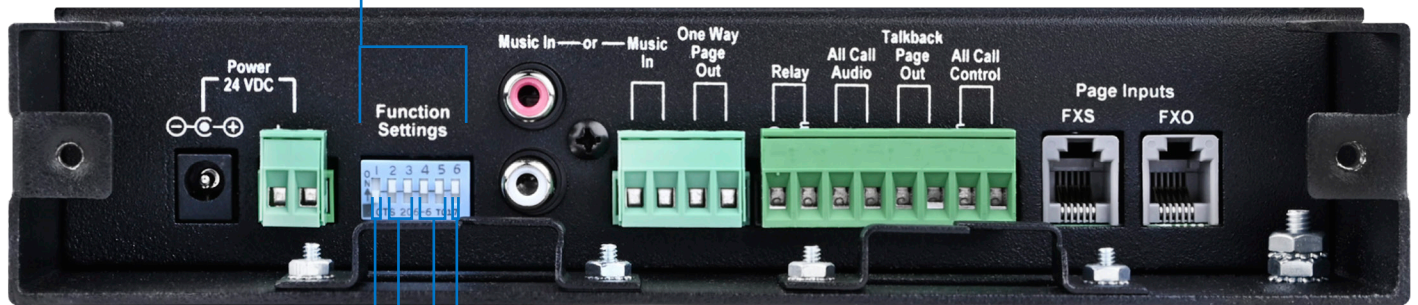
Timeout Settings (SW 2 & 3, SW 4 & 5)

Two options are available for disconnecting calls automatically. The **Silence Timeout** indicates how many seconds the VRCPA will stay active after the audio input has stopped. This can be used to disconnect a call after the caller has stopped speaking. Three values are available with the switch settings; additional options are available in software programming. If set to 0 (SW 2 & 3 both ON), the timeout is disabled in hardware but may be set to a value between 0-600 seconds in software programming.

The **Absolute Timeout** indicates how many seconds the VRCPA will stay active for any caller. When the timeout setting expires, the call is disconnected. Three values are available with the switch settings; additional options are available in software programming. If set to 0 (SW 4 & 5 both ON), the timeout is disabled in hardware but may be set to a value between 0-3600 seconds in software programming.

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Device needs to be power cycled for any DIP switch changes to take effect!
6-position dip switches are used to configure the basic functions:

| Power 24 VDC | Function Settings | Music In —— or —— Music In | One Way Page Out | Relay | All Call Audio | Talkback Page Out | All Call Control | Page Inputs FXS | FXO |

Page Output Selection

SW1
OFF One-Way Page Out Active Talkback Page Out Disabled
ON Talkback Page Out Active One-Way Page Out Disabled

Silence Time Out

SW2	SW3	Timeout
OFF	OFF	8 Seconds
OFF	ON	4 Seconds
ON	OFF	2 Seconds
ON	ON	Software Control Default 0 Seconds

Absolute Time Out

SW4	SW5	Timeout
OFF	OFF	128 Seconds
OFF	ON	16 Seconds
ON	OFF	8 Seconds
ON	ON	Software Control Default 0 Seconds

Page Input Selection

SW6
OFF FXO Port Active FXS Port Disabled
ON FXS Active FXO Port Disabled

For simple installations, these hardware switch settings may be the only configuration necessary. If so, proceed to the wiring connections section for diagrams showing typical connections to amplifiers or page controllers.

For access to other features available on the VRCPA, additional programming may be done using any of the following methods:

- DTMF tones from a telephone system connected to the FXS or FXO Input port. A single-line phone may be used by itself on the FXS port.

- Using the Valcom VIP-102B IP Solutions Setup Tool. The tool and reference manual are available for download from www.valcom.com/vipsetuptool.

- Through the built-in web interface

Access to the web interface or with the VIP-102B Setup tool for programming requires the VRCPA to have a valid IP address. By default, the VRCPA will try to acquire an IP address using DHCP (Dynamic Host Configuration Protocol) when connected to the network. Check with the local network administrator to determine what IP address was assigned from DHCP. If DHCP is not available, the VRCPA will default to an IP address of 192.168.6.207 after 3 minutes. The VIP-102B Setup tool may be used to discover the IP address, either by using the Scan function or the Network Diagnostics function in the tool. The tool can also be used to set the IP address using multicast communication. Refer to the VIP-102B Setup tool reference manual for more details. Using a telephone connected to the analog port on the VRCPA to set the IP address is discussed in the next section.

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The programming interface using DTMF from a telephone on the Input port provides a limited set of options.
It is very useful for setting the initial IP address so that further programming can be done over the IP network.

Entering DTMF Program Mode

The VRCPA monitors the telephony audio port for the DTMF digits "**" + Pass-code + "*" at any time during a paging event. Using the default values, this would be **9970*. If the VRCPA detects that sequence, the VRCPA enters into DTMF Program Mode. The outbound page interface will be disconnected and returned to idle condition.

DTMF Program Mode Acknowledgement

When entering DTMF Program Mode, the VRCPA responds to the user with an acknowledgement tone on the telephony input. The VRCPA shall then monitor the telephony input for DTMF parameter index combinations per DTMF Programming Options in the table shown below.

DTMF Programming Timeout

When DTMF programming mode is entered, the absolute timeout time shall be set for the duration of the call to 10 minutes, and the silence timeout disabled. If this absolute timeout is reached before the user terminates programming by hanging up, the VRCPA shall issue a reorder tone to the user and then hang up the telephony connection.

DTMF Program Menu

The DTMF Menu consists of the user keying in two DTMF digits to select the item to be programmed (the parameter index) and then a variable number of digits to perform the actual setting. To abort the current selection or to terminate a variable length field, the user will key in a '#'. After entry of a successful DTMF parameter index, the acknowledgement tone is transmitted and the VRCPA enters a state where the parameter value can be entered.
If the DTMF parameter index is invalid, the VRCPA returns a tone to indicate command rejection, and then waits for the next menu item selection.

Parameter	DTMF Parameter Index	Default Value	Valid Entry	Notes
Silence Timeout	01	00	00-99. 00 disables timeout	Takes effect on next call if DIP switches 3 and 4 are both "ON".
Absolute Timeout	02	000	000-999. 000 disables timeout	Takes effect on next call if DIP switches 5 and 6 are both "ON".
DHCP	20	1	0 disables, 1 enables	Takes effect upon system reboot.
IP Address	21	DHCP, with fallback to 192.168.6.207	xxx*xxx*xxx*xxx, where xxx are numbers between 0-255 and the * key denotes the dot between the octets. The variable length entry must be terminated via the # key.	Takes effect upon system reboot, if and only if the IP Address, netmask, and gateway comprise a valid subnet.
Netmask	22	24	01-31. This specifies the netmask using CIDR notation. (i.e. IP/Netmask)	Takes effect upon system reboot, if and only if the IP Address, netmask, and gateway comprise a valid subnet.
Gateway	23	DHCP, with fallback to 192.168.6.1	xxx*xxx*xxx*xxx, where xxx are numbers between 0-255 and the * key denotes the dot between the octets. The variable length entry must be terminated via the # key.	Takes effect upon system reboot, if and only if the IP Address, netmask, and gateway comprise a valid subnet.
Pass-code	90	9970	Variable length entry from 0-10 characters long consisting of the DTMF digits 0-9. The variable length entry must be terminated via the # key.	Takes effect on next call.
Reset Defaults	95	N/A	N/A	Resets all settings to system defaults and reboots unit.
Reboot	99	N/A	N/A	Reboots unit.

CIDR Values for Dotted Decimal Netmasks

Netmask	Value	Netmask	Value	Netmask	Value	Netmask	Value
128.0.0.0	01	255.128.0.0	09	255.255.128.0	17	255.255.255.128	25
192.0.0.0	02	255.192.0.0	10	255.255.192.0	18	255.255.255.192	26
224.0.0.0	03	255.224.0.0	11	255.255.224.0	19	255.255.255.224	27
240.0.0.0	04	255.240.0.0	12	255.255.240.0	20	255.255.255.240	28
248.0.0.0	05	255.248.0.0	13	255.255.248.0	21	255.255.255.248	29
252.0.0.0	06	255.252.0.0	14	255.255.252.0	22	255.255.255.252	30
254.0.0.0	07	255.254.0.0	15	255.255.254.0	23	255.255.255.254	31
255.0.0.0	08	255.255.0.0	16	255.255.255.0	24		

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The VIP-102B IP Solutions Setup Tool provides access for programming all of the available software-controlled functions of the VRCPA. The tool is available as a free download from the Valcom web site at https://www.valcom.com/vipsetuptool. A Reference Manual for the program is also available at the same location.

When first starting the VIP-102B program, the initial screen will prompt to scan the network for available devices. If the VRCPA has a valid IP address (from DHCP or from DTMF programming, for example), the tool should find the VRCPA (and any other Valcom IP equipment). Follow the prompts in the VIP-102B tool to scan the found devices into the tool for further programming. The scanning process is beyond the scope of this document, but full details are in the VIP-102B tool Reference Manual noted previously.

The VIP-102B Programming tool downloads the programming information from the VRCPA. Changes to the programming are made off-line in the tool, then uploaded back to the VRCPA. Until the programming is uploaded, the VRCPA is not changed.

In the following discussion, some of the Properties and Time programming tabs/screens are not shown, as they are typical for all Valcom devices and are fully described in the VIP-102B Reference Manual.

After scanning and selecting the discovered devices the VRCPA displays as a **Universal Page Interface** device type. The **Summary** tab shows a read-only view of the basic properties of the device.

| Summary | Properties | Network | Time | System | Channels | Group Membership | SIP |

Host Name:

Domain Name:

Use DHCP: ☐

IP Address: 192.168.98.154

Subnet Mask: 255.255.255.0 - (/24)

Gateway IP Address: 192.168.98.1

Preferred DNS Server:

Alternate DNS Server:

SIP SDP NAT:

Continuous Beacon: ☑   LLMNR: ☑

Use Syslog Daemon: ☐   Transport: UDP

Syslog Daemon: Port: 514

The **Network** tab contains fields to configure the **IP address** statically or check the "**Use DHCP**" box to acquire an IP address from a DHCP server on the network.

Summary	Properties	Network	Time	System	Channels	Group Membership	SIP

Page Input:	FXS Loop Start
Page Output:	Talkback Page Out
Music Volume:	1.99
Pre-Announce Tone Volume:	0.99
Paging Mode:	Analog
Absolute Timeout (secs):	0
Silence Timeout (secs):	0
Phone Admin Passcode:	9970
Page Security Code:	
Page Volume:	Low
Pre-Announce Tone:	☑
Use Custom Pre-Announce Tone:	☐ Upload File
Paging Method:	Store and Play
Page Replay Control:	0
Inter-Page Timing (secs):	1
FXS / FXO Channel Priority:	Medium
Collect DTMF:	☑
Dial Code Length:	3
Delay After DTMF (secs):	2

Defaults   ✔ Apply   ✖ Cancel

The **System** tab is where most of the device-specific capabilities are configured.

The first items shown are read-only and indicate the hardware switch positions and volume control knob positions. These must be set directly on the VRCPA and cannot be controlled through software.

Page Input – Indicates which Analog input port (either FXS or FXO) is active. This is controlled by DIP switch SW6 with OFF making FXO active and ON making FXS active.

Page Output – Indicates the active Analog output connection. This can be One-Way Page Out or Talkback Page Out and is controlled by DIP switch SW1 with OFF being One-Way Page Out and ON being Talkback Page Out.

Music Volume – Indicates the volume level for Background Music set by the knob labeled Music on the front of the VRCPA. The range is approximately 0 to 2.

Pre-Announce Tone Volume – Indicates the volume level for Page Alert Tone played just before a page announcement. It is set by the knob labeled Tone on the front of the VRCPA and has a range of approximately 0 to 2.

The following items are configurable.

Paging Mode – Select between Analog and Valcom.
- **Analog** – Basic input from FXS, FXO, or SIP to the Analog One-Way Page output or Talkback output. Paging is output only from this device to the selected analog output.
- **Valcom** – Acts as a gateway between FXS, FXO or SIP to Valcom IP devices for paging or intercom. Calls may be inbound or outbound from this device

Absolute Timeout – The maximum number of seconds before the device will hang up the page. This is only configurable in software if DIP switch SW4 and SW5 are ON. The allowable range is 0-3600 seconds. Set to 0 to disable this timeout.

Silence Timeout – The number of seconds during which no audio is present on the input before the device will hang up the page. This is only configurable in software if DIP switch SW2 and SW3 are ON. Allowable range is 0-600 seconds. Set to 0 to disable this timeout.

Phone Admin Passcode – The digits that must be keyed into the phone to gain access to DTMF programming. This can be from 0-10 digits (leave blank to disable).

Page Security Code – The digits that must be keyed into the phone to make a page. This can be 0-10 digits (leave blank to disable).

Page Volume – Selectable volume output of Low (-10dBm) or High (2.2dBm) for analog One Way Page Out signal.

Pre-Announce Tone – Check to play alert tone over speakers at the start of a page. Uncheck to disable.

Use Custom Pre-Announce Tone – If checked, allows loading a custom page tone. An Upload Device Files dialog box will display when selecting the Upload File button. (See Upload Device Files under the Communications menu option.)

Paging Method – Live or Store and Play. If Live is chosen, no further options are available and paging audio will pass through the VRCPA as it is spoken. If Store and Play is chosen, the paging audio will be recorded by the VRCPA and played out when the call is disconnected. Other configuration options become available with Store and Play.
- **Page Replay Control** – This is the number of times (0 to 5) to repeat the page when using Store and Play.
- **Inter-Page Timing** – The number of seconds (1 to 10) between repeats when using Store and Play.
- **FXS / FXO Channel Priority** – The priority of the page that comes from either the FXS or FXO input. Selectable options are Low, Medium, and High. Page Group priority 20, 35, and 50 respectively. (SIP tab Channel Priority determines IP priority with same options)
- **Collect DTMF** – This is only available when using the Talkback Page Out (DIP switch SW1 is ON). Typically, this is used when connecting to a multi-zone controller which requires DTMF digits to select a paging zone.
  — Dial Code Length – Sets the number of digits to collect. The digits keyed in will be output before the recorded audio page is played.
  — Delay After DTMF – The number of seconds to wait between the output of the last recorded DTMF digit and the beginning of the recorded audio.

When used in combination with other Valcom IP devices, the information in the **Channels** tab (such as Dial Code) would need to be entered with values appropriate for the overall installation.

The **Group Membership** tab is used to assigned this VRCPA to appropriate Valcom IP Paging groups. Refer to the VIP-102B Reference Manual for more information.

The **SIP** tab is used to enter the necessary information for the VRCPA to be a RingCentral SIP endpoint. When Store and Play is selected (Paging Method on the System tab), up to four phone extensions may be configured on each VRCPA. The main fields are described below. For a full description of all fields, refer to the VIP-102B Reference Manual.

Transport – Sets the transport protocol to be used for SIP communications. Selections indicate what transport protocol the VIP device will accept inbound and which protocol to originate outbound. "Accept: TCP+UDP, Originate: UDP" is the default. Other selections allow define UDP to be accepted and originate with UDP, or allow TCP or UDP to be accepted and originate TCP.

Phone Number – Defines the unique number that is assigned to this device. The SIP Phone Number, or directory number, is the number this device will respond to for inbound SIP connections. Phone Numbers can contain letters, numbers, period (.) or hyphen (-).

Description – The text description of this SIP identity (30 characters maximum). This is for information only.

Authentication Name – The identification used to register with a SIP server or Proxy. This can also be referred to as a username for the SIP server.

Secret – The password used with the Authentication Name when registering with a SIP server.

Realm – The security domain in which the Authentication Name and Secret are defined. Generally, this field can be left blank unless required by the SIP server or provider.

SIP Servers – The SIP registrar server(s) to which this endpoint device will attempt to register. At least one primary (required) and up to 3 backup servers may be entered. These entries may be an IP address or server hostnames (if DNS is properly configured on the Network Tab). The Register checkbox must be selected (checked) to enable the registration to occur. The DNS SRV can be checked if the SIP servers are defined using DNS SRV records to provide a list of SIP servers.

Channel Priority – Sets the priority of a SIP call for paging relative to calls originating on the FXS or FXO inputs when Store and Play is used.

With **Store and Play** enabled on the **System** tab, the SIP tab will now display four **SIP** subtabs that can be programmed for up to four separate SIP extensions. There will also be a new selection named **Max Calls** to allow a single SIP phone number to accept simultaneous calls. Each additional call assigned to a phone number will reduce the available channels to maintain a maximum of four total simultaneous calls for the device.

A **Night Ring** function is also available. A SIP number with Night Ring checked will not answer the inbound call, but will play a ring tone to its analog output while that SIP call goes unanswered.

Night Ring:	☐
CID Number:	5405632000
CID Name:	VRCPA
Channel Priority:	Medium ⌄
Max Calls:	1 ⌄

If the Valcom option for **Paging Mode** was selected on the **System** tab, additional features are available.

Paging Mode:	Valcom ⌄

The **Night Ring** function now allows selecting a **Valcom audio group of IP endpoints** to which the Night Ring audio will be sent via multicast. The VRCPA can also be a member of that group and will play the audio to its One-way Page Out port. The **Auto Destination** field on the **SIP** tab can be used to have intercom calls from an IP Speaker to be forwarded to an IP phone from the VRCPA.

Night Ring:	☐	Night Ring Group:	⌄
CID Number:	5405632000		
CID Name:	VRCPA		
Auto Destination:			

The VRCPA includes a web interface for programming most of the available options.
To access the web programming pages, open a web browser and enter the IP address of the VRCPA.

Valcom RingCentral® Paging Adapter

Password

•••••

LOGIN

The default **password** is *admin*. Enter this and click on **LOGIN**. The Summary page will be displayed When programming through the web interface, the changes made on each page are not stored until the Apply button is clicked. Moving to another page without clicking Apply will discard the changes. Settings are also not effective until the VRCPA has been rebooted. If multiple changes are being made, click Apply before moving to a different page, but the Reboot can be delayed until finished with other changes.

Valcom RingCentral® Paging Adapter

| RingCentral SIP | Summary | General | Network | Store and Play | Firmware Update | | Reboot |

| 1 | 2 | 3 | 4 |

SIP Transport

Accept: TCP+UDP, Originate: UDP

✓ Register

SIP Server

sip.ringcentral.com

Remote SIP Port

5060

Local SIP Port

5060

Outbound Proxy

sip10.ringcentral.com

Outbound Proxy Port

5090

SIP User ID

15405632000

Authenticate ID

850543587654

Authenticate Password

Ku5v99r4

The **SIP tab** provides fields to configure the basic settings to allow this device to register with RingCentral. The entries are labeled similar to the names used in the RingCentral provisioning web pages.

SIP Transport – Allows configuring for accepting/originating either UDP or TCP protocol
Accept: TCP+UDP, Originate: UDP (default)   **Accept:** UDP, Originate: UDP   **Accept:** TCP+UDP, Originate: TCP

The **Register checkbox** must be selected (checked) to enable the registration to occur.

SIP Server is also known as the **SIP Domain** in RingCentral provisioning.

Remote SIP Port is the destination port on the SIP server. In RingCentral Phone provisioning, this is usually listed as a number after the colon in the SIP Domain.

Local SIP Port is the port on which the VRCPA will listen for inbound SIP communications.

Outbound Proxy and **Outbound Proxy Port** is the address and port to which SIP communications should be directed to be "proxied", or forwarded, to the next segment of the network. This entry is usually a different address than the SIP Server entry. This entry may be an IP address, or, if DNS is properly configured on the Network Tab, a server name. In RingCentral Phone provisioning, this is usually listed as a number after the colon in the SIP Domain.

SIP User ID is the **User Name** in RingCentral provisioning.

Authenticate ID and **Authenticate Password** correspond to the Authorization ID and Password in RingCentral provisioning.

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The Advanced Configuration section includes additional settings that may be used, but are typically not necessary for use with RingCentral.

Advanced Configuration

Max Simultaneous Calls

> 1

SIP Channel Priority

> Normal

Description

VRCPA

Realm

CID Name

VRCPA

CID Number

5405632000

Backup 1 SIP Server	Backup 1 SIP Server Port
	5060

Backup 2 SIP Server	Backup 2 SIP Server Port
	5060

Backup 3 SIP Server	Backup 3 SIP Server Port
	5060

Max Simultaneous Calls indicates the number of calls allowed on this SIP channel before returning a busy signal. Leave on the default of 1 for RingCentral.

SIP Channel Priority sets the priority of a SIP call for paging relative to calls originating on the FXS or FXO inputs when Store and Play is used.

Description is a text field for a description of this SIP channel.

Realm is the security domain in which the Authentication Name and Secret are defined. Generally, this field can be left blank unless required by the SIP server or provider; not used with RingCentral.

CID Name is the text that will be provided by this endpoint device as the Caller ID information for an outbound connection to other endpoints. Endpoints with Caller ID display capability will be able to display this information.

CID Number is the telephone number that will appear on endpoint devices with Caller ID capability.

Backup SIP Servers and **Backup SIP Server Ports** are the registrar server address(es) and port(s) to which this endpoint device will attempt to register. Additional servers are typically not necessary with RingCentral, but can be entered. These entries may be an IP address, or, if DNS is properly configured on the Network Tab, server names.

DNS SRV

RTP Port

20000

Keep Alive Timeout (seconds)

3600

Options Timer (seconds)

0

Idle Timeout (seconds)

0

Max Call Timer (seconds)

0

Night Ring

Auto Destination

APPLY   CANCEL

The **DNS SRV box** can be checked if using DNS to provide a list of SIP servers to which the VRCPA will attempt to register. This is not used with RingCentral.

RTP Port is the IP port number used for Real Time Protocol communication. RTP is the protocol used for sending/receiving audio over the network. This entry should be an even number to conform to industry standards.

Keep Alive Timeout (secs) is the interval in seconds (20 – 3600) that SIP registration is sent to a SIP server (default 3600 seconds).

Options Timer (secs) is the interval in seconds (0 – 300) that a SIP Options message is sent to the SIP server. (default 0 disables the option)

Idle Timeout (secs) is configurable from 0 – 60. If audio packets are not received after the set number of seconds the device will terminate the call. (default 0 disables the timeout.)

Max Call Timer (secs) is configurable from 0 – 600. The device will terminate the call when the timer reaches the number of seconds configured. (default 0 disables the timeout) This option is typically used when calls are not hung up properly.

Night Ring, when checked, enables this SIP phone number to be used as a Night Ring station. The VRCPA will not answer the inbound call, but will play a ringback tone to signal that a call is being received.

Auto Destination is a phone number to which calls originating from the VRCPA will be placed. This is used with talkback IP speakers.

Valcom RingCentral® Paging Adapter

RingCentral SIP | Summary | General | Network | Store and Play | Firmware Update   Reboot

Software Revision
2.0.12

IP Address
192.168.98.154

MAC Address
00:D0:5F:04:49:42

Enhanced SIP Option
Installed

Valcom Gateway Option
Installed

Device Status
Idle

All of the fields on the Summary page are read-only, and reflect the current status of the VRCPA. The first section of the screen displays property information about the device.

Hardware Controlled Settings

Page Input
FXS - Loop Start

Page Output
Talkback Page Out (FXO)

Silence Timeout
Software Control -- See General Page

Absolute Timeout
Software Control -- See General Page

Music Volume
1.99

Pre-Announce Tone Volume
0.99

The second half of this screen shows configuration settings of the device.   These fields indicate the settings of the hardware switches and volume control knobs on the VRCPA.

Note: if DIP switch 4 and 5 are ON, Silence Timeout and Call Timeout values are configured on the General page.

Valcom RingCentral® Paging Adapter

RingCentral SIP	Summary	General	Network	Store and Play	Firmware Update		Reboot

Paging Mode

Analog (SIP/FXO/FXS <-> Analog output)

Silence Timeout (in seconds, 0 disables)

0

Page Volume

Low

✓ Pre-Announce Tone

☐ Use Custom Pre-Announce Tone

Page Security Code (Leave Blank to Disable)

Phone Administration Passcode

••••

Web Administration Password   Confirm Web Administration Password

APPLY   CANCEL

The **General** page includes settings for the Absolute Timeout (0-3600) and the **Silence Timeout** (0-600) in number of seconds (If DIP switches 2-5 are ON). Entering 0 for either disables that timeout. If **Store and Play** is active then Absolute Timeout is not configurable or displayed.

Page Volume can be selected as Low or High.

Pre-Announce Tone, if checked, plays a tone prior to the page.
The Pre-Announce Tone can be customized using an audio file in 16kHz, single channel WAV format. To do this, check the "**Use Custom Pre-Announce Tone**" and an additional line will be displayed to select and upload the new audio file.

✓ Use Custom Pre-Announce Tone

Pre-Announce Tone File (16kHz single channel WAV format file) [Choose File] No file chosen

Click "**Choose file**" to select a file in 16kHz single channel WAV format, preferably no longer than a few seconds in duration.

The **Page Security Code** is a numeric value, up to 10 digits, that can be set to restrict paging. When a page is attempted, the caller must enter this code to complete the connection.

Phone Administration Passcode defines the digits that must be keyed into the phone to gain access to DTMF programming. This can be from 0-10 digits (default is 9970; leave blank to disable).

The **Web Administration Password** is the password used to access this web interface for programming. Default password is admin.

The **Network** page contains fields for setting the **Host** and **Domain Names** for the VRCPA and other IP network values. The VRCPA is initially set to use **DHCP** to acquire an IP address from the network. When set for DHCP, other fields will allow setting a **DHCP Timeout** and **Fallback** addresses if DHCP is not successful. The Timeout value sets how long the VRCPA will attempt to get an address from DHCP, and the Fallback values are the IP address settings the VRCPA will use if DHCP does not provide an address.

The **SIP SDP NAT External IP** is used to enter an external (public) IP address that should be substituted in the SIP SDP data for the IP address of the VRCPA. This is typically used when connecting to hosted SIP phone systems when the local Internet firewall does not handle the SIP NAT translations.

Link Local Name Resolution Service checkbox is an optional selection that enables the VRCPA to send LLMNR packets onto the network. This can aid in network troubleshooting if necessary.

☐ Use DHCP

IP Address
192.168.98.154

Subnet Mask
255.255.255.0

Gateway IP Address
192.168.98.1

Preferred DNS Server

Alternate DNS Server

If the Use DHCP checkbox is <u>not</u> checked, the IP address information can be entered as static values. The web page will display fields for the **IP Address, Subnet Mask** and **Gateway IP Address**. The fields for **DNS Server** entries are optional, although other programming options for SIP may require the DNS entries to be entered.

Valcom RingCentral® Paging Adapter

RingCentral SIP | Summary | General | Network | Store and Play | Firmware Update | Reboot

Paging Mode
Live ▾

APPLY   CANCEL

On the **Store and Play** tab, the **Paging Mode** can be configured as either **Live** (default) or **Store and Play**.
In **Live** mode, the audio from the caller is sent to the output immediately.

Valcom RingCentral® Paging Adapter

RingCentral SIP | Summary | General | Network | Store and Play | Firmware Update | Reboot

Paging Mode
Store and Play

Page Replay Control
0

Inter-Page Timing
1

FXS/FXO Channel Priority
Normal

✓ Collect DTMF

Dial Code Length
3

Delay After DTMF Signaling
2

APPLY | CANCEL

When **Store and Play** is selected, several items are available to control the playback of the recorded audio.

Page Replay Control sets the number of times the recorded page is repeated. The page can be replayed up to 5 times. The default (0) indicates no repeats.

Inter-Page Timing controls the timing between pages, in seconds. The range is 1 to 10.

FXS/FXO Channel Priority sets the priority (Low, Normal or High) of the messages received on the analog interface.

If the analog output is set for Talkback Page Out (FXO), the option to **Collect DTMF** appears. If selected, the VRCPA will record DTMF digits keyed in from the phone (up to the **Dial Code Length**) before recording the voice audio. The **Delay After DTMF Signaling** (0-20 seconds) is the amount of time the VRCPA will wait before playing the DTMF codes from the output connection. After the DTMF is played out, the recorded voice audio will be played.

Valcom RingCentral® Paging Adapter

RingCentral SIP   Summary   General   Network   Store and Play   Firmware Update   Reboot

Select Update File   Choose File   No file chosen

UPDATE

Firmware Update tab allows you to upload a firmware update that you received from Valcom.
Click **Choose file** button, and you will be prompted with a **File Open** dialog box to select the firmware file.
Click **UPDATE** and the device will load the new firmware.

Wiring **Connections**

Connections to Self-Amplified Speakers or External Amplifier

FXS PORT OR
PBX STATION PORT
(DIP SWITCH SW6 OFF)

ANALOG TELEPHONE OR
PBX TRUNK PORT
(DIP SWITCH SW6 ON)

SELF-AMPLIFIED
HORN/SPEAKER

VP-624
Power
Supply

Other
-24VDC
Power
Supply*

Low
Level
Music
Source

-24VDC
POWER
SUPPLY

EXTERNAL
AMPLIFIER

* May be provided
by Page Control
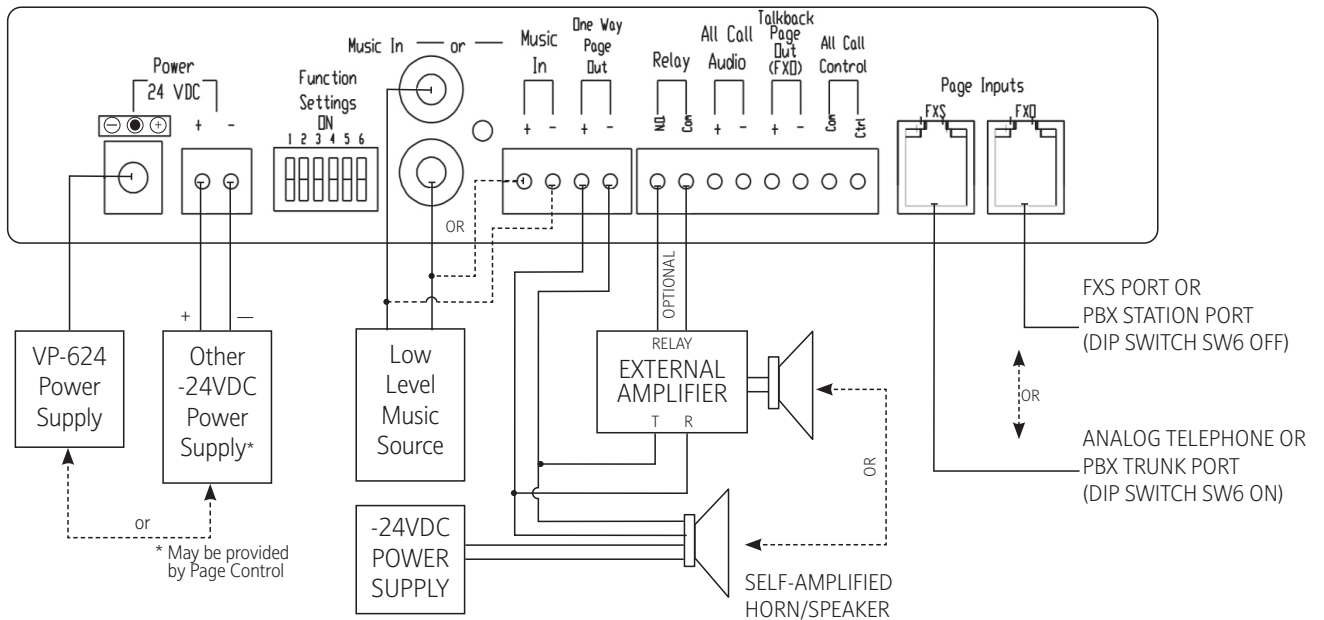
Connections to Multi-Zone Controllers

FXS PORT OR
PBX STATION PORT
(DIP SWITCH SW6 OFF)

ANALOG TELEPHONE OR
PBX TRUNK PORT
(DIP SWITCH SW6 ON)

VP-624
Power
Supply

Other
-24VDC
Power
Supply*

Low
Level
Music
Source

One-Way or Talkback
Valcom Zone Controller
V-2003A, V-2006A, V-ACS, Etc.
(Battery Feed Switch On)

T/R Input

* May be provided
by Page Control

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