3Com® VCX™ V7111 VoIP SIP Gateways User Manual

Version 4.2

`http://www.3com.com`

Part Number 900-0182-01 Rev AA

Published June 2004

3Com Corporation, 350 Campus Drive, Marlborough MA 01752-3064

CONTENTS

ABOUT THIS GUIDE

This User's Guide describes the following 3Com® VCX™ V7111 Gateways:

- The 24-port VCX V7111 24FXS

- The eight-port VCX V7111 8FXS and 8FXO

- The four-port VCX V7111 4FXS and 4FXO

- The two-port VCX V7111 2FXS

These products are supported by software version 4.2, and enable you to send voice fax and data over the same IP network.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, because of on-going product improvements and revisions, 3Com cannot guarantee the accuracy of printed material after the Date Published nor can it accept responsibility for errors or omissions. Updates to this document and other documents can be viewed by registered Technical Support customers. See "Appendix E: Obtaining Support for Your 3Com Products" in the *User's Guide* for details on how to register your product and get support from 3Com.

FCC Notice to Users

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.

- Increase the separation between the equipment and receiver.

- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

- Consult the dealer or an experienced radio/TV technician for help.

> ⚠ **CAUTION:** *Installation and service of 3Com VCX V7111 products must be performed only by authorized, qualified service personnel.*

Telecommunication Safety

The safety status of each port on the device is declared and detailed in the following table:

Table 1   Safety Status Indicators

Ports	Safety Status
Ethernet (100 Base-T)	SELV
FXS	TNV-3

- **TNV-3** — Circuit whose normal operating voltages exceeds the limits for an SELV circuit under normal operating conditions and on which over voltages from Telecommunication Networks are possible.

- **SELV** — Safety extra low voltage circuit.

How to Use This Guide

This book covers these topics:

- Chapter1: VCX V7111/SIP Overview

- Chapter 2: 2, 4, or 8 Channel VCX V7111 Hardware Installation

- Chapter 3: VCX V7111 24FXS Hardware Installation

- Chapter 4: Software Installation

- Chapter 5: Profiling and Operation

- Chapter 6: Provisioning

- Chapter 7: Device Management

- Chapter 8: Diagnostics

- Chapter 9: Specifications

- Appendix A: BootP/TFTP Configuration Utility

- Appendix B: RTP/RTCP Payload Types

- Appendix C: DTMF, Fax, and Modem Modes

- Appendix D: DHCP Server Configuration

- Appendix E: Obtaining Support for Your 3Com Products

Conventions

Table 2 lists conventions that are used throughout this guide.

Table 2   Notice Icons

Icon	Notice Type	Description
	Information note	Information that describes important features or instructions.
	Caution	Information that alerts you to potential loss of data or potential damage to an application, device, system, or network.
	Warning	Information that alerts you to potential personal injury or death.

Each abbreviation, unless widely used, is spelled out in full when first used, and only Industry standard terms are used throughout this manual. The $ symbol indicates hexadecimal notation.

When referring to VCX V7111 products, this document uses the following abbreviations:

Table 3   Documentation Abbreviations

Abbreviation	Product Name
2FXS	VCX V7111 2FXS
4FXS	VCX V7111 4FXS
4FXO	VCX V7111 4FXO
8FXS	VCX V7111 8FXS
8FXO	VCX V7111 8FXO
24FXS	VCX V7111 24FXS
FXS	Any FXS Gateway:2FXS, 4FXS, 8FXS, or the 24FXS.
FXO	Any FXO Gateway: the 4FXO or the 8FXO

When referring to all these products generally, this document uses the term *VCX V7111*.

Related Documentation

The following documents are available on the 3Com Partner Access website for the 3Com VCX V7111 Gateway:

- *3Com VCX V7111 SIP Release Notes*

- *3Com VCX V7111 Gateway Fast Track Installation Guide*

Documentation Comments

Your suggestions are important to us because we want to make our documentation more useful to you.

Please send e-mail comments about this guide or any of the VCX 7111 documentation and Help systems to:

VOICE_TECHCOMM_COMMENTS@3com.com

Please include the following information with your comments:

- Document title

- Document part number (usually found on the front page)

- Page number

- Your name and organization (optional)

Example:

3Com VCX V7111 VoIP SIP Gateways User Manual
Page 25

CHAPTER 1: VCX V7111 SIP OVERVIEW

This document provides you with the information about installation, configuration, and operation of the VCX 7111 24FXS, 8FXS, 8FXO, 4FXS, 4FXO, and 2FXS VoIP Gateways. As these units have similar functionality, except for the number of channels and some minor features. It is expected that the readers are familiar with regular telephony and data networking concepts.

Gateway Description

The VCX V7111 telephony Gateways provide excellent voice quality and optimized packet voice streaming over IP networks. The product enables voice, fax, and data traffic to be sent over the same IP network. It is based on the award-winning, field-proven TrunkPack design and uses well-established DSP voice compression technology.

The VCX V7111 incorporates up to 24 analog ports for connection, either directly to an enterprise PBX (VCX V7111 FXO), to phones, or to fax (VCX V7111 FXS), supporting up to 24 simultaneous VoIP calls.

Additionally, the VCX V7111 Gateways are equipped with a 10/100 Base-T Ethernet port for connection to the LAN.

The VCX V7111 Gateways are best suited for small to medium size enterprises, branch offices or for residential Gateway solutions.

The VCX V7111 Gateways enable you to make free local or international telephone and fax calls between the distributed company offices, using their existing telephones/fax. These calls are routed over the existing IP Internet or Intranet corporate data networks ensuring that voice traffic takes the minimum of space on the data network.

The VCX V7111 Gateways are very compact devices, designed to be installed either as a desktop unit (see Figure 2) or installed in a 19-inch rack (see Figure 6 on page 20).

The VCX V7111 supports the SIP protocol, enabling the deployment of voice over packet solutions in environments where each enterprise or residential location is provided with a simple Gateway.

This provides the enterprise with a telephone connection (for example, RJ-11), and can transmit the voice and telephony signals over a packet network.

Additionally, for emergency use, the VCX V7111 FXS Gateway provides a Life Line, connected to the unused pins on port 4 (or port 2 for the 2FXS), with a relay to an analog line, even if the Gateway is powered off.

The layout diagram, <u>Figure 5</u>, illustrates a typical VCX V7111 8FXS or 8FXO and a VCX V7111 4FXS or 2FXS VoIP application.

Figure 1   VCX V7111 24FXS VoIP Gateway

Figure 2   VCX V7111 8FXS Front View

Figure 3   VCX V7111 4FXS Front View

Figure 4   VCX V7111 2FXS Front View

Figure 5   Typical VCX V7111 VoIP Application

VCX V7111 Key Features

▪ High quality voice, data, and fax over IP networks.

▪ The VCX V7111 24FXS supports up to 24 analog telephone loop start FXS ports as shown in <u>Figure 1</u> on page 14.

▪ The VCX V7111 8FXS and 8FXO supports up to eight analog telephone loop start FXS or FXO ports as shown in <u>Figure 2</u> on page 14.

▪ The VCX V7111 4FXS and 4FXO supports up to four analog telephone loop start FXS or FXO ports as shown in <u>Figure 3</u> on page 14.

▪ The 2FXS supports up to two analog telephone loop start FXS ports, shown in <u>Figure 4</u> on page 14.

▪ Connected to the IP network using a 10/100 Base-T Ethernet interface.

▪ Coders include: G.711, G.723.1, G.726, G.727, G.729A, and NetCoder from 6.4 through 8.8 Kbps, selectable per channel.

▪ T.38 Fax with superior performance (round trip delay up to 9 seconds).

▪ Compliant with SIP (RFC 3261).

▪ Life Line, connected to the unused pins on port 4 (or port 2 for the 2FXS), with a relay to an analog line, even if the VCX V7111 FXS is powered off.

▪ LEDs on the front and rear panels provide information on the operating Gateway status and of the network interface.

▪ Restart button on the front panel restarts the VCX V7111 Gateway.

▪ The VCX V7111 8FXS or 8FXO compact, rugged enclosure providing up to eight analog RJ-11 ports within a compact housing of only one-half of a 19-inch rack unit, 1 U high (1.75" or 44.5 mm).

▪ The VCX V7111 24FXS 19-inch, 1 U rugged enclosure provides up to 24 analog FXS ports, using a single 50 pin Telco connector.

▪ Mounting option of installing two VCX V7111 4FXS, 4FXO, 8FXS, or 8FXO Gateways in a single 19-inch rack shelf, one U high (1.75" or 44.5 mm).

Reader's Notes

CHAPTER 2: 2, 4, OR 8 CHANNEL VCX V7111 HARDWARE INSTALLATION

> **i** *All VCX V7111 Gateways have similar functionality except for the number of channels (the VCX V7111 24 FXS and 2FXS support only FXS), and all versions are referred to collectively in this document as the VCX V7111.*
>
> *VCX V7111 FXS refers only to the 24FXS, 8FXS, 4FXS, and 2FXS Gateways.*
>
> *VCX V7111 FXO refers only to 8FXO and 4FXO Gateways.*

Hardware Installation Procedure

Unpacking

To unpack the hardware, follow these steps:

1 Open the carton and remove packing materials.

2 Remove the VCX V7111 Gateway from the carton.

3 Check that there is no equipment damage.

4 Check, retain, and process any documents.

5 Notify 3Com of any damage or discrepancies.

> **⚠** **CAUTION:** *Installation and service of 3Com VCX V7111 products must be performed only by authorized, qualified service personnel.*

Rack Mounting Installation

Figure 6   Rack Mounting

The VCX V7111 is installed into a standard 19-inch rack by the addition of the two brackets supplied.

To install the VCX V7111, follow these steps:

1   Fasten the short bracket to the right-hand side of the Gateway using the two screws provided, as shown in Figure 6, and carefully positioning the peg into a convenient ventilation hole in the side of the VCX V7111 box.

2   Fasten the long bracket to the left-hand side of the Gateway using the two screws provided as shown in Figure 6, and carefully positioning the peg into a convenient ventilation hole in the side of the VCX V7111 box.

3   Insert the Gateway into the 19-inch rack and fasten the left-hand and right-hand brackets to the vertical tracks of the 19-inch rack, using standard 19-inch rack bolts (not provided).

To connect the cables, see "Cable Connections" on page 21.

Desktop Mounting Installation

Figure 7   VCX V7111 Desktop or Shelf

The VCX V7111 is installed on a desk or shelf without additional brackets.

To connect the cables, see "Cable Connections" on page 21.

Cable Connections

The RJ-45 (Ethernet) and RJ-11 (Ports) pinouts and connectors are shown in Figure 8, and pins are numbered from the left with the latching finger position at the bottom.

Figure 8   RJ-45 LAN and RJ-11 Port Connectors and Pinouts

RJ-45 Connector and Pinout

```
1 2 3 4 5 6 7 8
```

```
1 - Tx+        4, 5, 7, 8
2 - Tx-          not
3 - Rx+       connected
6 - Rx-
```

RJ-11 Connector and Pinout

```
1 2 3 4
```

```
1 -  Not connected
2 -  Tip
3 -  Ring
4 -  Not connected
```

To connect the cables for desktop or rack-mount, follow these steps:

1 Perform one of the following:

  a) When using a 4FXS or 8FXS Gateway, insert each of the RJ-11 connectors on the 2-wire line cords of the POTS phones into the RJ-11 sockets on the rear of the Gateway.

  b) When using a VCX V7111 FXO Gateway, insert each of the RJ-11 connectors on the 2-wire line cords coming from PSTN/PBX into the RJ-11 sockets on the rear of the Gateway.

   Telephone lines and extensions of up to 7,300 m (24,000 ft) can be achieved using regular 24 AWG line cord.

2 Insert the RJ-45 connector on the 10/100 Base-T cable from your LAN to the ETH RJ-45 socket (on the rear of the VCX V7111) to provide the link to your LAN.

3 Connect the VCX V7111 Gateway to the correct AC power supply, and the installation is now complete.

Installation of the VCX V711 FXS Life Line

8FXS and 4FXS Gateways provide a Life Line connection on port 4.

The VCX V7111 2FXS Gateway provides a Life Line connection on port 2.

The 24 FXS and VCX V7111 FXO Gateways do not support the Life Line.

This feature provides a wired phone connection to any PSTN or PBX FXS port, upon power-down conditions. When the power outage occurs, the phone that is connected to the Life Line port, on pins 2 and 3, is wired to the PSTN or PBX FXS wires on pins 1 and 4 on the same connector. Therefore, you can use the phone even when the Gateway is not powered-on. To use this function, you must use a splitter that connects pins 1 and 4 to another source of an FXS port, and pins 2 and 3 to the POTS phone.

The pinout of the Life Line RJ-11 phone connector is as follows:

- 1 = Life Line TIP

- 2 = TIP

- 3 = RING

- 4 = Life Line RING

See Figure 9 for the RJ-11 connector pinout.

Figure 9   RJ-11 Connector and Life Line Pinout for VCX V7111 FXS Gateways

RJ-11 Connector and Life Line Pinout


```
1 2 3 4          1 -   Life Line Tip
                 2 -   Tip
                 3 -   Ring
                 4 -   Life Line Ring
```

Front Panel LED Indicators

The VCX V7111 front panel LEDs indicate the Ethernet LAN status, Data (RTP) activity, and state of the Gateway's ports.

Figure 10   Front Panel LED Indicators

Table 4   Figure 10 Legend

1	Restart button

Functionality of the Front Panel LEDs is explained in Table 5.

Table 5   Front Panel Network LED Indicators

Label	Type	Color	State	Meaning
LAN	Ethernet Link Status	Green	ON	Valid Connection to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Sending and receiving SIP messages.
		Red		Not supported in current release
Ready	Device Status	Green	ON	Device Powered, Self test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction

Table 6   VCX V7111 Channel LEDs

Label	Type	Color	State	Meaning
Activity	FXS Tel Port	Green	ON	Off-Hook/Ringing for Phone Port
	FXO Line Port	Green	ON	Line-Seize/Ringing State for Line Port

Rear Panel LED Indicators and Connectors

Figure 11   Rear Panel LED Indicators and Connectors

Table 7   Figure 11 Legend

1	Power Supply Inlet
2	Chassis Ground Screw
3	10/100 Base-T RJ-45 Port
4	9-pin RS-232 Port
5	Eight RJ-11 Ports FXS or FXO

Table 8   Meaning of Rear Panel LED Indicators

Label	Type	Color	State	Meaning
ETH-1	Ethernet Status	Yellow	ON	Ethernet Port Receiving Data
		Red	ON	Collision

Table 9   Explanation of Rear Panel Connectors/Switches

Label	Type	Function	Comment
100-240V ~ 1A	3-pin power inlet	AC input	Connection to external power supply
1 to 8	RJ-11	Eight FXS or FXO Ports	VCX V7111 8FXS or 8FXO 2-wire Loop Start interface
1 to 4	RJ-11	Four FXS or FXO Ports	VCX V7111 4FXS or 4FXO 2-wire Loop Start interface
1 to 2	RJ-11	Two FXS Ports	VCX V7111 2FXS 2-wire Loop Start interface
ETH 1	RJ-45	10/100 Base-T Port	Shielded
RS-232	DB-9, DCE	Status Messages	Gateway connects to PC's RS-232 port with a straight cable (Figure 58 on page 125).
⏚	Grounding screw	Chassis Ground	*Must* be securely connected.

Reader's Notes

CHAPTER 3: VCX V7111 24FXS HARDWARE INSTALLATION

⚠ **CAUTION:** *Installation and service of 3Com VCX V7111 products must be performed only by authorized, qualified service personnel.*

Hardware Installation Procedure

Unpacking
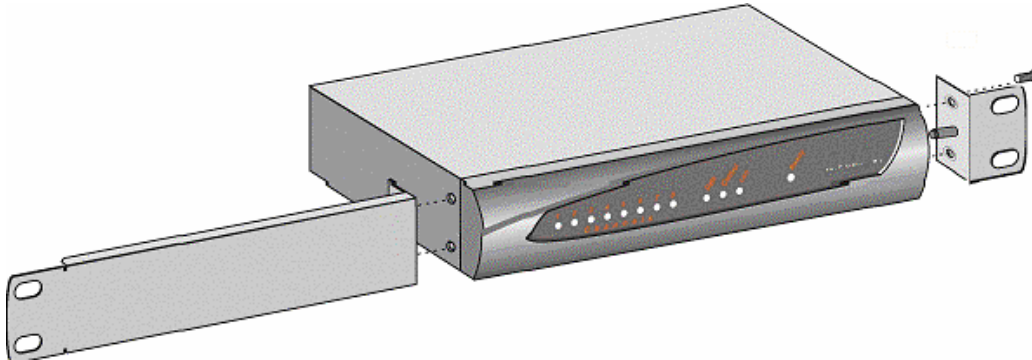
To unpack the VCX V7111 24FXS, follow these steps:

1 Open the carton and remove packing materials.

2 Remove the VCX V7111 from the carton.

3 Check that there is no equipment damage.

4 Check, retain and process any documents.

5 Notify 3Com of any damage or discrepancies.

MDF Adapter

To connect twenty-four 2-wire lines into the VCX V7111, a Main Distribution Frame (MDF) Adapter Block should be used as shown in . This converts the standard RJ-11 connector into a plain pair of wires that are terminated within a 50-pin Telco connector.

Figure 12   VCX V7111 24FXS in a 19-inch Rack with MDF Adapter

> *The only equipment shown in Figure 12 and supplied by 3Com is the VCX V7111 24FXS Gateway and, as an option, the MDF Adapter.*

As input (on the front of the 19-inch rack), the Adapter Block takes in twenty-four 2-wire lines with standard RJ-11 connectors.

As output (on the rear of the 19-inch rack), the Adapter Block provides 24 wire pairs, which need to be terminated into a single 50-pin male Telco connector.

The 50-pin connector must be wired according to the pinout in Table 10 and shown in Figure 13.

Cable Connections

The 50-pin Telco connector mounted on the rear of the VCX V7111 24FXS is wired according to the pinout in Table 10 and shown in Figure 13. The user's cable-mounted 50-pin Telco connector, supporting the twenty-four 2-wire phone lines, must be wired identically.

Table 10   Pin Allocation in 50-pin Telco Connector

Phone Channel	Connector Pins	Phone Channel	Connector Pins
1	1/26	13	13/38
2	2/27	14	14/39
3	3/28	15	15/40
4	4/29	16	16/41
5	5/30	17	17/42
6	6/31	18	18/43
7	7/32	19	19/44
8	8/33	20	20/45
9	9/34	21	21/46
10	10/35	22	22/47
11	11/36	23	23/48
12	12/37	24	24/49

Figure 13   50-pin Telco Connector

The RJ-45 (Ethernet) and RJ-11 (POTS) pinouts and connectors are shown in Figure 14. Pins are numbered from the left with the latching finger position at the bottom.

Figure 14   RJ-45 and RJ-11 Connectors and Pinouts

RJ-45 Connector and Pinout

1 - Tx+
2 - Tx-
3 - Rx+
6 - Rx-

4, 5, 7, 8
not
connected

RJ-11 Connector and Pinout

1 - Not connected
2 - Tip
3 - Ring
4 - Not connected

Figure 15   19-inch Rack Mounting

The VCX V7111 24FXS Gateway is supplied with brackets (ears) fitted to each side of the enclosure so that the VCX V7111 24FXS can be immediately installed in the 19-inch rack.

To install the rack mount VCX V7111 24FXS, follow these steps:

1 Insert the VCX V7111 24FXS into the 19-inch rack, adjust it to the correct position and use two standard rack-screws (not supplied) to secure each of the two brackets to the rack frame.

2 Insert each of the RJ-11 connectors on the 2-wire line cords of the POTS phones into the RJ-11 sockets on the front of the MDF Adapter Block.

   Up to 3,000 m (10,000 feet) of 24 AWG line cord can be used to connect telephones.

3 Attach to the each of the sockets on the rear of the MDF Adapter Block one pair of wires from a 25-pair Octopus cable.

4 Connect the wire-pairs at the other end of the Octopus cable to a male 50-pin Telco connector. The pinout must be that shown in Table 10 and Figure 13 on page 29.

5 Insert and fasten this 50-pin connector into the female 50-pin Telco connector mounted at the rear of the VCX V7111 24FXS and labeled Analog Lines 1-24.

6 Insert the RJ-45 connector of the 10/100 Base-T cable into the RJ-45 connector mounted at the rear of the VCX V7111 24FXS and labeled Eth 1 for connection to your LAN.

7 Connect an electrically grounded strap to the chassis ground screw on the rear of the VCX V711 24FXS and fasten it securely according to the current standards.

8 Connect an electric power cord of the correct rating, from a grounded supply of the correct voltage, into the power socket mounted at the rear of the VCX V7111 24FXS and labeled 100 – 250 V ~ 50 – 60 Hz 2A.

9 Observe the front panel LEDs to determine the functioning of the VCX V7111 24FXS.

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The Channel LEDs indicate that the telephones connected to the rear 50-pin connector are each in one of the following states:

- Ringing or in the Off Hook position (LED lights green)

- Normal operation (LED is not lit)

- Not functioning (LED lights red)

The functions of all the LEDs of the VCX V7111 24FXS are shown in Table 12.

Front Panel LED Indicators

Figure 16   Front Panel LED Indicators

Table 11   Figure 16 Legend

1	Channels 1-24
2	Data
3	Control
4	LAN
5	Ready

Table 12 Function of Front Panel LED Indicators

Label	Type	Color	State	Function
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Currently not implemented
		Red	ON	Currently not implemented
		Orange		Currently not implemented
LAN	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Ready	Device Status	Green	ON	Device Powered and Self-test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction
Channels 1- 24	Tel Port	Green	ON	Off-Hook/Ringing for FXS Phone Port
	Tel Status	Red	ON	Malfunction
		Blank	-	Normal

Rear Panel LED Indicators/Connectors

Figure 17 Rear Panel LED Indicators and Connectors

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Table 13   Figure 17 Legend

1	Grounding Screw
2	AC Power Inlet
3	50-pin Telco Connector
4	Data LED
5	Cntrl LED
6	Ready LED
7	Monitor Port
8	LAN LEDs
9	Eth 1 and Eth 2

Table 14   Function of Rear Panel LED Indicators

Label	Type	Color	State	Function
Data	Packet Status	Green	ON	Transmitting RTP Packets
		Red	ON	Receiving RTP Packets
		Blank	-	No traffic
Cntrl	Control Link	Green	ON	Currently not implemented
		Red	ON	Currently not implemented
		Orange	ON	Currently not implemented
Ready	Device Status	Green	ON	Device Powered and Self-test OK
		Orange	ON	Software Loading/Initialization
		Red	ON	Malfunction
Eth 1	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Eth 2	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction

Table 15   Function of Rear Panel Connectors/Switches

Label	Type	Function	Comment
100-250 V~ 50 60 Hz 2A	3-pin AC	AC input	Connection to AC power cord
⏚	Grounding Screw	Chassis ground	
Analog Lines 1 to 24	50-pin Telco connector	FXS Ports	2-wire Loop Start interface
Eth 1	RJ-45	10/100 Base-T	Shielded port to Ethernet LAN. This is the default port.
Eth 2	RJ-45	10/100 Base-T	Shielded port to Ethernet LAN. The port is not in use for current SW release.
RS-232	DB-9, DCE	Status Messages	Gateway connects to PC's RS-232 port with a straight cable (see Figure 58 on page 125).

The DIP switch located on the VCX V7111 24FXS rear panel is not functional and should not be used.

Reader's Notes

CHAPTER 4: SOFTWARE INSTALLATION

All VCX V7111 Gateways have similar functionality except for the number of channels (the VCX V7111 24 FXS and 2FXS support only FXS), and all versions are referred to collectively in this document as the VCX V7111.

VCX V7111 FXS refers only to the 24FXS, 8FXS, 4FXS, and 2FXS Gateways.

VCX V7111 FXO refers only to 8FXO and 4FXO Gateways.

Installation Package

The Installation Package includes the following software and utilities:

Software:

- **MP108_ram.cmp** – Image Software for download to 8FXS, 4FXS, and VCX V7111 FXO Gateways.

- **MP124_ram.cmp** – Image Software for download to the 24FXS Gateway.

- **usa_tones.dat** – Call progress tones DAT file for download.

- **usa_tones.ini** – Call progress tones INI file (used to create DAT file).

- **SIPgw_FXS.ini** – INI example file for VCX V7111 FXS Gateways.

- **SIPgw_FXO.ini** – INI example file for VCX V7111 FXO Gateways.

- **MP1xx_Coeff_FXS.dat** – Telephony interface configuration file for 8FXS and 4FXS Gateways.

- **MP10x_Coeff_FXO.dat** – Telephony interface configuration file for VCX V7111 FXO Gateways.

- **MIB library** – Library of SNMP MIBs.

Utilities:

- **INI file utility.xls** – Excel utility for creation of the INI file.

- **TPDMUtil.exe** – Call progress tones file generator utility.

- **Bootp_install.exe**– 3Com BootP and TFTP configuration utility.

VCX V7111 Initialization

VCX V7111 Gateways come with ready-installed software. The basic installation can be done using the 3Com Configuration Utility, or from a web browser, such as from Microsoft Internet Explorer.

To change network parameters, use a web browser, 3Com BootP and TFTP configuration utility, or use third party BootP server.

For setting the SIP parameters in the INI file, either edit the INI example file, or generate such a file using the Excel utility provided.

The INI file and other configuration files can be downloaded directly from a web browser using the HTTP protocol a 3Com-provided configuration utility, or any standard TFTP server. The Image software file, ram.cmp, is only used for software upgrades.

The Call Progress tone file USA_TONES.INI is used to define call progress tone levels, cadence, and their frequency. To change the tone's parameters, first modify the file, and then using the TPDMUtil.exe utility, convert the text INI file to binary DAT file. This procedure is described in "Using Call Progress Tones and Ringing" on page 93.

The Coeff_FXS.dat and Coeff_FXO.dat files can be used respectively to modify the VCX V7111 FXS and FXO telephony interface characteristics, such as DC and AC impedance, feeding current, and ringing voltage. For more information, see "The coeff.dat Configuration File" on page 101.

Quick Setup Procedure

The following procedure describes how to set up the VCX V7111 Gateway with basic parameters using standard web browser (such as Microsoft Explorer). It is assumed that the IP address of the Gateway is known. If the IP address is unknown, use the 3Com Configuration Utility (or any standard BootP server) to set the Gateway IP address and subnet mask.

Usually the VCX V7111 Gateways are shipped with the following network parameters:

- VCX V7111 FXS Gateway IP address: 10.1.10.10

- VCX V7111 FXO Gateway IP address: 10.1.10.11

- Subnet: 255.255.0.0

- Default Gateway: 0.0.0.0

For quick VCX V7111 setup, follow these steps:

1 Power-up the Gateway; after self-testing, in about 20 seconds the Ready LED on the front panel turns to green. Any malfunction changes the Ready LED to red.

2 Before using your web browser to access the Gateway's Embedded Web Server, change your PC's IP address and subnet mask to correspond with the VCX v7111 factory default IP address and subnet mask.

3 Open any standard web-browsing application, such as Microsoft Internet Explorer (Version. 5.0 and higher) or Netscape Navigator (Version. 7.0 and higher), and specify the IP address of the Gateway in the Address field (for example, http://10.1.10.10 for FXS); the Embedded Web Server Enter Network Password screen appears, Figure 18.

Figure 18   Web Browser Screen

4 Enter the user name and password (default: Admin, Admin). Note that the User Name and Password fields are case-sensitive. Click *OK*; the Quick Setup screen is accessed.

5 In the Quick Setup screen, complete the Gateway's new IP address, NAT IP Address, Subnet Mask, and Default Gateway IP Address fields.

Set the basic SIP Gateway Parameters, as shown in Figure 19.

Figure 19   SIP Quick Setup Screen

IP Configuration	
IP Address :	10.8.201.158
NAT IP Address :	
Subnet Mask :	255.255.0.0
Default Gateway IP Address :	10.8.0.1
SIP Parameters	
Gateway Name :	10.8.8.10
Working with Proxy :	No
Proxy IP address :	10.8.8.10
Proxy Name :	
Enable Registration:	No
▬ Coder :	g711Alaw64k(20ms
Tables	
Tel to IP Routing Table:	OPEN
Endpoints' Phone Numbers Table:	OPEN

6   Open the Endpoints' Phone Numbers table and fill in the Gateway phone numbers. For
example, for the 8FXS, enter 1-8 in Channel(s) field and a starting Phone number, such
as 101, in adjacent Phone Number field. The 8FXS physical ports are associated with
phone numbers from 101 through 108. You do not need to fill the Hunt Group ID column.
Click *Submit* and close the window.

Figure 20   Endpoints' Phone Numbers

	Channel(s)	Phone Number	Hunt Group Id
1	1-8	101	
2			
3			
4			
5			
6			
7			
8			

7 When working without a proxy server, the Tel to IP Routing table needs to be defined: Enter in the Destination Phone Prefix (prefix of the called number) and the associated IP address.

In the following example, an outgoing (Tel->IP) call with the prefix number of 512 is sent to IP address 10.2.201.11; while all other numbers starting with 51 are sent to IP address 10.2.201.12. Assuming that the Gateway IP address is 10.2.1.1, all local calls starting with 10 (such as 101-108) are routed back to the Gateway. When the proxy server is used, the routing table can be used optionally as fallback in case of failure.

8 Click *Submit* and close the window.

Figure 21 Tel to IP routing table

	Destination Phone Prefix	IP Address
1	10	10.2.1.1
2	512	10.2.201.11
3	51	10.2.201.12
4	5	10.2.201.13
5		
6		
7		
8		
9		
10		

9 Click *Reset* and click *OK* in the prompt; wait approximately 60 seconds and refresh the web page.

10 Restore your PC's IP address and subnet mask to their original values.

11 Using the Advanced Configuration and Protocol Management screens, you can modify all SIP and other Gateway parameters. See "Web Management" on page 106 for detailed directions of operating under web management control.

12 Finally you can establish phone calls between phones connected to the Gateway ports (for FXS Gateway) by dialing numbers from 101 through 108.

BootP and TFTP Procedures

If the Gateway IP address is known, you can use a web browser for Gateway configuration and provisioning. Otherwise, you can use BootP (or DHCP) and TFTP protocols for initialization and software download.

Each time the VCX V7111 Gateway is powered-on, it performs the standard BootP procedure.

If the DHCPEnable = 1 line is included in the Gateway's INI file and if the BootP server was not found, the Gateway initiates its standard DHCP procedure to configure the Gateway network parameters (IP address, subnet mask, and default router address). If DHCP procedure is used, you need to find the new Gateway IP address allocated by DHCP server. Usually the System Administrator can provide this information.

If the BootP/DHCP server has not been found, the VCX V7111 Gateway starts working from its internal flash memory.

Usually the application software already resides in the VCX V7111 flash memory, therefore there is no need to use the BootP or TFTP procedure. Their download need only be used for changing the VCX V7111 configuration or for a new software upgrade.

The BootP protocol enables the network administrators to manage the configuration of the VCX V7111 Gateway from a central configuration server – the BootP/DHCP server.

The following RFCs (IETF Requests for Comment) describe BootP in detail: RFC 951, RFC 1542, and RFC 2132.

The VCX V7111 download the image file using the Trivial File Transfer Protocol (TFTP). The TFTP protocol is described in RFC 906 and RFC 1350.

Although DHCP and BootP servers are very similar in operation, the DHCP server includes some differences that might prevent its operation with BootP clients. However, many DHCP servers, such as Windows NT DHCP server, are backward-compatible with BootP protocol and can be used for VCX V7111 configuration.

> *The BootP server is normally used to configure the VCX V7111 initial parameters. Once this information has been provided, the BootP server is no longer needed. All parameters are stored in non-volatile memory and used when the BootP server is not accessible. The BootP server is required again if, for example, the VCX V7111 IP address is to be changed.*

Using BootP procedure, the following parameters are configured:

- **Boot and INI File Names** – Optional; the boot file name can contain one or two file names. The file ram.cmp is used to download the application image and the file MP1XX.INI is used for VCX V7111 provisioning. Either one, two, or no file names can appear in the Boot file name field. To use both file names, use a semi-colon (;) separator (without blank spaces) between the xxx.cmp and the yyy.ini files (for example, ram.cmp;SIPgw.ini).

- **Local IP Address** – IP address of your VCX V7111 Gateway.

- **Gateway IP Address** – If a default Gateway/Router is required, complete this field; otherwise, enter the address 0.0.0.0.

- **Subnet Mask** – Set the subnet mask fields *to correspond to your network IP address settings.*

- **TFTP Server IP address** – Usually TFTP and BootP servers are installed on the same host. However, when using the 3Com Configuration utility or a Microsoft DHCP server, it is possible to set the IP address of TFTP server (Boot Server Host Name field), and in this case BootP and TFTP servers can run from different hosts.

Configuring the TFTP Server

To configure the TFTP Server, follow these steps:

1   Set the default directory where the image file resides (C:\3Com\…).

2   Copy the image file (such as, ram.cmp) to the TFTP default directory on your host PC.

3   Copy the INI file and other optional configuration files (Call Progress tones and Coeff.dat files) to the TFTP default directory on your host PC. Ensure the correct coeff.dat file is used. Two different coefficient files are provided, for the VCX V7111 FXS and VCX V7111 FXO Gateways.

4   Set the TFTP timeout to 3 seconds and number of retransmissions to 20.

Using the 3Com BootP/TFTP Configuration Utility

The 3Com Configuration utility provides an easy way to configure the VCX V7111 Gateway. Similar to other BootP and TFTP servers, the application can be installed on Windows 98 or Windows NT/2000. With 3Com's BootP/TFTP Server Configuration utility, it is possible to use the integrated TFTP server (part of the Utility) or to install TFTP server on a different host. The utility enables remote reset of the VCX V7111 for triggering the initialization procedure (BootP and TFTP). For details of the procedure, see the 3Com *Software Utilities Manual*.

Configuration Utility Main Features

- BootP server supporting hundreds of entries.

- Integrated TFTP server.

- Common Log window displaying BootP and TFTP status.

- Contains all data required for provisioning of 3Com products.

- Provides the TFTP server address, enabling network separation of TFTP and BootP servers.

- Tools for backup and for restoring the local database.

- Templates.

- User-defined names for each entity.

- Option for changing the Gateway MAC (Media Access Control) address.

- Protection against entering fault information.

- Unicast or Broadcast BootP response.

Configuring the Windows NT DHCP Server

If a Microsoft Windows NT DHCP server is used in your organization, the server can be used in reservation mode to provide an IP address and other necessary information to the VCX V7111 Gateway.

To configure the Microsoft Windows NT DHCP Server to assign IP address information to BootP clients, add a reservation for each BootP client.

For information about how to add a reservation, see "the Managing Client Reservations" Help topic in the DHCP Manager.

The reservation builds an association between the Media Access Control address (12 digits, provided in VCX V7111 documentation) and the IP address. Windows NT Server provides the IP address based on the VCX V7111 Media Access Control (MAC) address in the BootP request frame.

To configure the Microsoft Windows NT DHCP server to provide boot file information to BootP clients, edit the BootP Table in DHCP Manager. The BootP Table is located in the Server Properties dialog box that can be accessed from the Server menu. For information about how to edit the BootP Table, view the BootP Table Help topic in DHCP Manager.

The following parameters must be entered:

- **Local IP address** – The IP address of your VCX V7111 Gateway.

- **Subnet mask** – See "BootP and TFTP Procedures" for the mask limits.

- **Gateway IP address** – The default Gateway IP address

- **Boot File name** – Optional, however, the boot file name normally should not be used. This field is only used for software upgrade, see "VCX V7111 Software Upgrade".

See "Appendix D: DHCP Server Configuration" on page 157 for detailed description of configuration of Windows NT DHCP server.

VCX V7111 Software Upgrade

General Upgrade Procedure

VCX V7111 Gateways include on-board flash memory already programmed with application software. The following procedure replaces the old stored software with the new version. To run this procedure, BootP and TFTP servers are required. A web browser can be used instead of the BootP server (see "Web Management" on page 106).

> *The file extensions CMP and INI should be written in lower case letters. When upgrading the internal software (either from a TFTP Server or from the Embedded Web Server) it is mandatory to download the ram.cmp file and all other related files: INI, coefficient, and call progress tone.*

To upgrade the integral software, follow these steps:

1 Start the TFTP and BootP servers.

2 Copy the new ramxxx.cmp file, mp108.ini file (for example, SIPgw.ini), and optional configuration files to the default TFTP server directory.

3 Set the Boot file and INI file names in the web browser Network settings page or in the BootP server: ramxxx.cmp -fb;mp108.ini. Other network parameters (such as the IP address and the subnet mask) stay unchanged. If so required, it is possible to update only the mp108.ini parameters. For this option, set the boot file name to: mp108.ini (without preceding ramxxx.cmp). After you cycle power, the INI parameters are downloaded using the TFTP procedure and stored in the non-volatile memory.

4 Reset the VCX V7111. Wait about 20 seconds until the Ready LED lights green.

5 After accomplishing BootP and TFTP procedures, the new software is downloaded and stored in the Gateway's flash memory.

> *The parameter -fb added to the Boot file name is used to specify the burning of flash memory with new software image. To test new software version without replacing the old version, skip the -fb parameter. In this case, the new software is downloaded directly to RAM, and not permanently stored into flash.*

Upgrade Procedure Using the 3Com Configuration Utility

The following procedure describes how to upgrade VCX V7111 software using the 3Com Configuration Utility.

To upgrade the Software using the 3Com Configuration Utility, follow these steps:

1 Download and install the 3Com Configuration Utility with your web browser.

2 Open the 3Com Configuration Utility from *Start>Programs>BootP*; the 3Com Configuration Utility main screen opens:

Figure 22   3Com Configuration Utility Main Screen

3 Click the *Edit* tab to open the *Edit* menu.

4 Select *Preferences* to open Preferences window shown in .

Figure 23   Preferences Screen

5   In the Directory field, click >> and navigate to the directory of the source *.CMP and *.INI files. All downloaded files should reside in this folder, including ram.cmp, mp108.ini, Coeff.dat, and Call Progress tone.dat files.

6   Click *OK* to return to the main screen.

7   In the *Services* menu, choose *Clients*. This opens the Client Configuration screen shown in Figure 24. The parameter fields displayed on the right side of the screen constitute the VCX V7111 software profile configuration. The parameter fields are all blank in the case of a Client Not Found.

Figure 24   Client Configuration

8 Complete the Client MAC address and Client Name fields.

9 Enter the IP address (such as 10.2.37.1).

10 Enter the subnet (such as 255.255.0.0); set the subnet to a valid value in accordance with the IP address.

11 Enter the IP address of the default Gateway; it can be any address within the subnet.

12 Select the required Boot and INI files.

13 To permanently store the new image file in the VCX V7111 flash memory, add –fb suffix to Boot file name, such as ram.cmp –fb. After entering the file names, click *Apply & Reset*.The following status messages are displayed in the 3Com BootP/TFTP Server main screen:

Figure 25   3Com Configuration Utility – TFTP download

Reader's Notes

CHAPTER 5: PROFILING AND OPERATION

> *All VCX V7111 Gateways have similar functionality except for the number of channels (the VCX V7111 24 FXS and 2FXS support only FXS), and all versions are referred to collectively in this document as the VCX V7111.*
>
> *VCX V7111 FXS refers only to the 24FXS, 8FXS, 4FXS, and 2FXS Gateways.*
>
> *VCX V7111 FXO refers only to 8FXO and 4FXO Gateways.*

SIP Profile

Supported SIP Features

The VCX V7111 complies with RFC 3261 IETF standard.

The VCX V7111/SIP main features are:

- Works with or without a proxy. When no proxy is specified an internal routing table is used instead.

- Fallback to internal routing table if communication with the proxy server fails.

- Supports two redundant proxy servers. If main proxy fails, the VCX V7111 automatically switches to the first redundant proxy.

- Proxy and Registrar Authentication (handling 401 and 407 responses) using either Basic or Digest (MD5) methods.

- Proxy Registration, such as:

  ```
  REGISTER sip:proxyname SIP/2.0
  VIA: SIP/2.0/UDP 212.179.22.229;branch=z9hG4bRaC7AU234
  From: <sip:101@sipgatewayname>;tag=1c29347
  To: <sip:101@sipgatewayname>
  Call-ID: 10453@212.179.22.229
  Seq: 1 REGISTER
  Expires: 3600
  Contact: <sip:101@212.179.22.229;user=phone>;expires=3600
  Content-Length: 0
  ```

  Where the *proxyname* and *sipgatewayname* strings are defined in the VCX V7111 INI file (or configured from the web).

  By default the REGISTER request will be sent to the proxy IP address. It is possible to change the registration address by configuring the REGISTRAR IP address.

Usually the VCX V7111 FXS Gateway sends up to eight REGISTER messages, for each port, using the port's phone number on the left side of the SIP URL and the common Gateway password.

The VCX V7111 FXO Gateway usually registers just once, using the username parameter in the SIP URL, such as sip: *username@sipgatewayname*. The username parameter can be defined in the INI file or from the Embedded Web Server.

It is also possible to configure a separate username and password for each Gateway port.

- Supports the following methods: INVITE, CANCEL, BYE, ACK, REGISTER, OPTIONS, INFO, and REFER.

- Modifying connection parameters for an already established call (re-INVITE).

- Working with a Redirect server and handling 3xx responses.

- Early Media (supporting 183 Session Progress).

- PRACK reliable provisional responses (RFC 3262).

- Diversion Header (draft-levy-sip-diversion-05.txt).

- Call Transfer supplementary services using REFER, Refer-To, Referred-By, Replaces, and NOTIFY.

- Session Timer (draft-ietf-sip-session-timer-10.txt).

- Remote party ID (draft-ietf-sip-privacy-04.txt).

- DTMF Payload for DTMF Digits (RFC 2833).

- DTMF out of band transfer using INFO method (draft-choudhuri-sip-info-digit-00.txt).

- SIP URL: sip:*phone_number@IP address* (such as 122@10.1.2.4, where 122 is the phone number of the source or destination) or sip:*phone_number@domain name*, such as 122@myproxy.com.

- Reliable UDP transport, with retransmissions.

- Supported codecs:

  - G.711 A-law

  - G.711 μ-law

  - G.723

  - G.729A

  - G.726

  - NetCoders (from 6.4 through 8.8 Kbps)

- Can negotiate codec from a list of given codecs

- SIP Responses:

  - Informational Responses: 100, 180, 181,182, 183

  - Successful responses: 200 OK

  - Failure Responses: 4xx

For detailed information, see the latest *VCX V7111/SIP Release Notes*.

Using SIP Gateway Features

Details of how to use and configure the SIP Gateway Parameters are shown in Table 16.

Table 16   Using SIP Gateway Features

Feature	Parameter	Sheet of Excel Utility	Value
SIPGatewayName	SIP Gateway name. If specified, is used on the right side of the SIP URL in the FROM header. Otherwise, the Gateway IP address is used.	SIPgw	
Use SIP Proxy	IsProxyUsed	SIPgw	1
	ProxyIp	SIPgw	IP
	Proxy IP address		
ProxyIP	IP address of an external proxy server. Can either be an explicit IP address or a domain name. Two redundant proxy servers are supported, (define three lines each with a different IP address in the INI file). The Gateway starts working with the main proxy (first on the list), in case of a failure; the Gateway tries to communicate with the first redundant proxy (second on the list). Once a proxy is found, the Gateway will work with it, until the next failure occurs. If neither of the proxies responds, the Gateway goes over the list again. The Gateway monitors connection with the proxy using keep-alive messages (OPTIONS).	SIPgw	IP
ProxyName	Proxy domain name used in SIP Request-URI. If this parameter is not specified, the Proxy IP address will be used instead in SIP URL.	SIPgw	
No Proxy	IsProxyUsed	SIPgw	0

Table 16   Using SIP Gateway Features

Feature	Parameter	Sheet of Excel Utility	Value
Use routing table	Define routing table using: a. PREFIX = <prefix, IP address> list b. IP = *,<IP address> When a proxy is not used, it is necessary to define IP routing table, to enable VCX V7111 Gateway to find destination IP address, according to received dial number. The routing table is defined in INI file, using PREFIX or per phone number definitions. The Gateway first searches for Phone table to find a destination IP address, than it looks for Prefix parameter, and later for Prefix = *,<IP address> definition. Prefix = *, <IP address> defines destination IP address for any other phone number.	Phone or/and Prefix Tables	Phone numbers, Prefixes, and IPs
Set numbers to end points	Channel2Phone= <channel>,<phone> or ChannelList = port, number, phone or using Hunt Group definition The easiest way to define endpoint phone number is to use ChannelList parameter. For example, to define 101 – 107 numbers for an 8FXS, use a single line: ChannelList = 0,8,101. The first parameter (0) indicates the first endpoint number. The second parameter (8) indicates the number of endpoints. The third parameter (101) indicates the first endpoint phone number. Up to ten such ChannelList definitions can appear in the same INI file. One or more phone numbers in the ChannelList can be modified by using Channel2Phone definition, following the ChannelList parameter in the INI file.	EndPoints	Phone numbers
Dial plan	MaxDigits Maximal number of digits in dialed number.	General	Max digits
	TimeBetweenDigits Timeout between dialed digits, used to terminate dialing. Usually it is set to four seconds.	General	Time in seconds
Choose Coder	CoderName	General	Preferred coder name

Table 16   Using SIP Gateway Features

Feature	Parameter	Sheet of Excel Utility	Value
Several Coders	CoderName	General	List of coders
	In this mode, several codecs are sent in SDP message. On receiving the remote response (200 OK) with its SDP, a process of matching coders is done between the local set of coders (from the INI file) and the remote set. The local coders are the preferred ones, and if the first local coder is included in the remote SDP response, then it is selected; otherwise, next local coder is tested for match.		
Automatic dialing	IsDialNeeded	General	0
	TargetOfChannel	Automatic Dialing	Phone numbers to dial
	This is used to perform automatic dialing once OFF HOOK is detected in FXS Gateway or ringing is detected on FXO port. There is no need to dial in this mode. For each channel, define destination phone number, using TargetOfChannel<channel> = phone number definition.		
One Stage Dialing,	IsTwoStageDial	SIPgw	0
	IsUseFreeChannel	General	1
IP → FXO calls	VCX V7111 FXO seizes the next available FXO line, and dials the destination phone number received in INVITE message. Use the IsWaitForDialTone parameter to specify whether the dialing comes after detection of dial tone, or immediately after seizing the line. The FXO Gateway releases the call if busy or fast busy (reorder) tone is detected on the FXO port.	General	

Table 16   Using SIP Gateway Features

Feature	Parameter	Sheet of Excel Utility	Value
Two Stage Dialing, IP → FXO calls	IsTwoStageDial	SIPgw	1
	IsUseFreeChannel	General	1
	For Two Stage Dialing the VCX V7111 FXO seizes the next available PSTN/PBX line, without performing any dial, the remote user is connected over IP to PSTN/PBX, and all further signaling (dialing and call progress tones) is done directly with the PBX without Gateway intervention. Usually the phone number received in INVITE message is not used, however if IsUseFreeChannel = 0, the phone number received in INVITE, is used for seizing specific FXO line that has same number. The FXO Gateway releases the call if busy or fast busy (reorder) tone is detected on the FXO port.		

Table 16   Using SIP Gateway Features

Feature	Parameter	Sheet of Excel Utility	Value
Using Hunt Groups	TrunkGroup_x = a-b, <starting number>	TG	Range 1– 99
	IsUseFreeChannel	General	
	PSTNPrefix = <prefix>, <Hunt Group ID>	TGRoute	
	AddTrunkGroupAsPrefix		
	This feature defines groups of Gateway ports, called hunt groups, for routing outgoing IP→Tel calls. For example:		
	TrunkGroup_1 = 1–4,100		
	TrunkGroup_1 = 8,200		
	TrunkGroup_2 = 5–7,300		
	In this example, hunt group number 1 is composed of channels 1 – 4 and 8.		
	To use the hunt group feature, it is also required to set IsUseFreeChannel parameter to 1, and to define routing rules, using the PSTNPrefix.		
	The routing IP→Tel rules define for each called number, according to its prefix, the Hunt Group ID to where the call is sent.		
	For example: PSTNPrefix = 101,1;an outgoing IP→Tel call, with called number starting with 101 is allocated to a free channel in Hunt group number 1.		
	This feature applies mostly to FXO Gateways, but in some cases it also applies to FXS Gateways.		
	Add Hunt Group ID, as a prefix, to destination number, (AddTrunkGroupAsPrefix = 1) for Tel → IP calls. This feature enables users to differentiate incoming IP calls and apply routing and number manipulation rules, based on the hunt group from where the call arrived.		
	When using Hunt groups ChannelList and Channel2Phone parameters are ignored.		

Getting Started SIP Gateway Example

In this section, two 8FXS Gateways are configured to be used as SIP Gateways. The end-point numbers are from 101 through 108 for the first Gateway and from 201 through 208 for the second Gateway. After finishing the configuration, you can perform telephone calls between telephones connected to a single 8FXS Gateway, or between both VCX V7111 Gateways. SIP Proxy is not used in this example, and call routing is performed using the internal Tel to IP Routing table.

To configure the call, follow these steps:

1 Check connections and tools setup (TFTP, BootP, and HyperTerminal).

2 Build the INI file.

The INI file is a text file containing a list of parameters for the 8FXS. The file can be written manually or generated by the Excel utility provided. To use the Excel utility, first install the Microsoft Office 2000 Excel application.

In this example, the EXCEL utility is used:

- Invoke the EXCEL utility.

- On the Endpoints sheet, define local phone numbers for each 8FXS Gateway. For the first Gateway, define local phone numbers: from 101 through 108. For the second Gateway, define local phone numbers: from 201 through 208.

- On the Phones Prefix Routing Table sheet, define routing IP addresses for each dialed number. (This is required when a proxy is not used.)

- On the SipGW sheet, define that SIP Proxy is not used.

- Click the *Generate SIP INI File* button in the General sheet.

- Check that the SipGW.ini file was generated in the folder C:\SIPgw\.

Figure 26   Example of INI File for the First 8FXS Gateway

```
MGControlProtocolType = 8

MaxDigits = 3
CoderName = g711Alaw64k

IsProxyUsed = 0

; Phone of each end point
ChannelList = 0,8,101

; Logger information
EnableSyslog = 0
LoggerFormat = 0

; IP to Phones routing table

Prefix = 10,10.2.37.10
Prefix = 20,10.2.37.20
```

3 Download the INI file using TFTP and BootP procedures and check (viewing the RS-232 terminal) that there are no errors.

4 Pick up the phone connected to port 1 of the first 8FXS Gateway and dial 102, to the phone connected to port 2 of the same Gateway. Check for progress tones in the calling end-point and for ringing in the called end-point. Answer in the called end-point and check for voice quality.

5 Dial 201 from the phone connected to port 1 of the first 8FXS Gateway; the phone connected to port 1 of the second 8FXS will ring. Answer the call and check for voice quality.

SIP Call Flow

The following call flow describes SIP messages exchanged between two 8FXS Gateways during simple call.

1 Phone 6000 dials 2000, sending an INVITE message to Gateway 10.8.201.161.

Figure 27   SIP Call Flow


```
F1  10.8.201.158 ==> 10.8.201.161     INVITE
INVITE sip:6000@10.8.201.161;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.158;branch=z9hG4bKacolwbzYF
From: <sip:2000@10.8.201.158>;tag=1c3535
To: <sip:6000@10.8.201.161>
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
CSeq: 20214 INVITE
Contact: <sip:2000@10.8.201.158;user=phone>
Supported: 100rel,em
Accept-Language: en
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO
```

```
Content-Type: application/sdp
Content-Length: 208

v=0
s=Phone-Call
t=0 0
o=3ComGW 87943 43401 IN IP4 10.8.201.158
c=IN IP4 10.8.201.158
m=audio 6000 RTP/AVP 8 96
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20

F2  10.8.201.161 ==> 10.8.201.158     180 RINGING
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.8.201.158;branch=z9hG4bKacolwbzYF
From: <sip:2000@10.8.201.158>;tag=1c3535
To: <sip:6000@10.8.201.161>;tag=1c29715
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
CSeq: 20214 INVITE
Supported: 100rel,em
Content-Length: 0
```

2 Phone 2000 answers the call, and sends 200 OK message to Gateway 10.8.201.158.

```
F3  10.8.201.161 ==> 10.8.201.158     200 OK
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.8.201.158;branch=z9hG4bKacolwbzYF
From: <sip:2000@10.8.201.158>;tag=1c3535
To: <sip:6000@10.8.201.161>;tag=1c29715
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
CSeq: 20214 INVITE
Contact: <sip:6000@10.8.201.161;user=phone>
Supported: 100rel,em
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO
Content-Type: application/sdp
Content-Length: 208

v=0
s=Phone-Call
t=0 0
o=3ComGW 30762 37542 IN IP4 10.8.201.161
c=IN IP4 10.8.201.161
m=audio 4040 RTP/AVP 8 96
a=rtpmap:8 pcma/8000
a=ptime:20
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
F4  10.8.201.158 ==> 10.8.201.161     ACK
ACK sip:6000@10.8.201.161;user=phone;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.158;branch=z9hG4bKachoWSQxD
From: <sip:2000@10.8.201.158>;tag=1c3535
To: <sip:6000@10.8.201.161>;tag=1c29715
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
CSeq: 20214 ACK
Supported: 100rel,em
Content-Length: 0
```

3 Phone 6000 goes onhook, Gateway 10.8.201.161 sends BYE to Gateway 10.8.201.158.

```
F5  10.8.201.161 ==> 10.8.201.158     BYE
BYE sip:2000@10.8.201.158;user=phone;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.161;branch=z9hG4bKacLBzZgmA
From: <sip:6000@10.8.201.161>;tag=1c29715
To: <sip:2000@10.8.201.158>;tag=1c3535
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
CSeq: 34541 BYE
```

```
Supported: 100rel,em
Content-Length: 0


F6 10.8.201.158 ==> 10.8.201.161     200 OK
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.8.201.161;branch=z9hG4bKacLBzZgmA
From: <sip:6000@10.8.201.161>;tag=1c29715
To: <sip:2000@10.8.201.158>;tag=1c3535
Call-ID: 2123353775377NrpL-2000--6000@10.8.201.158
CSeq: 34541 BYE
Supported: 100rel,em
Content-Length: 0
```

SIP Authentication Example

The 8FXS Gateway supports basic and digest (MD5) authentication types, according to the SIP standard. A proxy server might require authentication before forwarding an INVITE message. A registrar server may also require authentication for client registration. A proxy replies to an unauthenticated INVITE with a 407 Proxy Authorization Required response, containing a Proxy-Authenticate header with the form of the challenge. After sending an ACK for the 407, the User Agent can then resend the INVITE with a Proxy-Authorization header containing the credentials.

User Agent, redirect or registrar servers typically use 401 Unauthorized response to challenge authentication containing a WWW-Authenticate header, and expect the re-INVITE to contain an Authorization header.

The following example describes the Digest Authentication procedure including computation of User Agent credentials.

The REGISTER request is send to registrar server for registration, as follows:

```
REGISTER sip:10.2.2.222 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c17940
To: <sip: 122@10.1.1.200>
Call-ID: 634293194@10.1.1.200
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
```

On receiving this request the registrar returns 401 Unauthorized response.

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.2.1.200
From: <sip:122@10.2.2.222 >;tag=1c17940
To: <sip:122@10.2.2.222 >
Call-ID: 634293194@10.1.1.200
Cseq: 1 REGISTER
Date: Mon, 30 Jul 2001 15:33:54 GMT
Server: Columbia-SIP-Server/1.17
Content-Length: 0
WWW-Authenticate: Digest realm="3com.com",
nonce="11432d6bce58ddf02e3b5e1c77c010d2",
stale=FALSE,
algorithm=MD5
```

According to the sub-header present in the WWW-Authenticate header the correct REGISTER request is formed.

Since the algorithm used is MD5, take:

- The username is equal to the endpoint phone number: 122.

- The realm return by the proxy: 3com.com.

- The password from the INI file: 3Com.

- The equation to be evaluated: (according to RFC this part is called A1).

- 122:3com.com:3Com.

- The MD5 algorithm is run on this equation and stored for future usage.

- The result is: a8f17d4b41ab8dab6c95d3c14e34a9e1.

Next we need to evaluate the par called A2. We take:

- The method type REGISTER

- Using SIP protocol sip

- Proxy IP from INI file 10.2.2.222

The equation to be evaluated:

```
REGISTER:sip:10.2.2.222.
```

The MD5 algorithm is run on this equation and stored for future usage. The result is: a9a031cfddcb10d91c8e7b4926086f7e.

The final stage:

- The A1 result

- The nonce from the proxy response: 11432d6bce58ddf02e3b5e1c77c010d2

- The A2 result

The equation to be evaluated:

```
"A1:11432d6bce58ddf02e3b5e1c77c010d2:A2".
```

The MD5 algorithm is run on this equation. The outcome of the calculation is the response needed by the Gateway to be able top register with the proxy. The response is: b9c45d0234a5abf5ddf5c704029b38cf.

At this time a new REGISTER request is issued with the response:

```
REGISTER sip:10.2.2.222 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c23940
To: <sip: 122@10.1.1.200>
Call-ID: 654982194@10.1.1.200
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
Authorization: Digest, username: 122,
```

```
realm="3com.com",
nonce="11432d6bce58ddf02e3b5e1c77c010d2",
uri="10.2.2.222",
response="b9c45d0234a5abf5ddf5c704029b38cf"
```

On receiving this request, if accepted by the proxy, it will return a 200 OK response closing the REGISTER transaction.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c23940
To: <sip: 122@10.1.1.200>
Call-ID: 654982194@10.1.1.200
Cseq: 1 REGISTER
Date: Thu, 26 Jul 2001 09:34:42 GMT
Server: Columbia-SIP-Server/1.17
Content-Length: 0
Contact: <sip:122@10.1.1.200>; expires="Thu, 26 Jul 2001 10:34:42 GMT";
action=proxy; q=1.00
Contact: <122@10.1.1.200:>; expires="Tue, 19 Jan 2038 03:14:07 GMT";
action=proxy; q=0.00
Expires: Thu, 26 Jul 2001 10:34:42 GMT
```

Remote Extension with FXO and FXS Gateways Example

This application explains how to demonstrate remote extension by means of IP, using 8FXO and 8FXS Gateways. In this configuration, PBX incoming calls are routed to Remote Extension using the 8FXO and 8FXS Gateways.

Requirements

- One 8FXO Gateway

- One 8FXS Gateway

- Analog phones (POTS)

- PBX – one or more PBX loop start lines

- LAN

Connect the 8FXO ports directly to PBX lines as shown in the following figure:

Figure 28   8FXS and 8FXO Layout

Dialing from Remote Extension When the Phone Is Connected to a VCX 7111 8FXS

To configure the call, follow these steps:

1   Take the handset off, to hear the dial tone coming from PBX, as if the phone was connected directly to PBX.

2   8FXS and 8FXO establish a voice path connection from the phone to the PBX immediately after the phone handset was raised.

3   Dial the destination number (the DTMF digits are sent, over IP, directly to PBX).

4   All tones heard are generated from PBX (such as ringback, busy, or fast busy tones).

5   There is one-to-one mapping between 8FXS ports and PBX lines.

6   The call is disconnected when the phone connected to 8FXS goes on-hook.

Dialing from other PBX line, or from PSTN

To configure the call, follow these steps:

1   Dial the PBX subscriber number the same way as if the user's phone was connected directly to PBX.

2 Immediately as PBX rings into the 8FXO, the ring signal is send to phone connected to the 8FXS.

3 Once the phone's handset, connected to the 8FXS, is raised, the 8FXO seizes the PBX line and the voice path is established between the phone and the PBX line.

4 There is a one to one mapping between PBX lines and 8FXS ports. Each PBX line is routed to the same phone (connected to the 8FXS).

5 The call is disconnected when phone connected to the 8FXS goes on-hook.

8FXS Configuration (using the FXS INI file)

To configure the 8FXS INI file, follow these steps:

1 Configure in the 8FXS INI file the endpoint numbers from 100 through 107.

2 Configure TargetOfChannel table to include phone numbers of the 8FXO Gateway: such as TargetOfChannel0 = 200. (When phone connected to port 0 goes off-hook, the FXS Gateway automatically dials 200 number).

3 Configure IP to phone table, to IP address and numbers of the FXO Gateway: such as Prefix=20,10.1.10.2 (where 10.1.10.2 is an IP address of the 8FXO).

4 Set IsDialNeeded = 0 to activate automatic dialing, when the handset goes off-hook.

```
IsDialNeeded = 0
;----------------------------------
; Phone of each end point
;----------------------------------
ChannelList = 0,8,100

;----------------------------------
; Automatic dialed numbers
;----------------------------------
TargetOfChannel0 = 200
TargetOfChannel1 = 201
TargetOfChannel2 = 202
TargetOfChannel3 = 203
TargetOfChannel4 = 204
TargetOfChannel5 = 205
TargetOfChannel6 = 206
TargetOfChannel7 = 207

; Phones to IP routing table
;----------------------------------
Prefix = 20,10.1.10.2
```

8FXO configuration (using the FXO INI file)

To configure the 8FXO INI file, follow these steps:

1 Configure in the 8FXO INI file the endpoint numbers from 200 through 207.

2 Configure TargetOfChannel table to include phone numbers of the 8FXS Gateway: such as TargetOfChannel0= 100 (when ringing signal is detected at port 0 of FXO Gateway, the FXO Gateway automatically dials 100 number).

3 Configure IP to phone table, to IP address and numbers of the FXS Gateway: such as Prefix=10, 10.1.10.3 (where 10.1.10.3 is an IP address of the 8FXS).

4 Set IsDialNeeded = 0 to activate automatic dialing when ringing is detected at FXO port.

```
IsDialNeeded = 0
;----------------------------------
; Phone of each end point
;----------------------------------
ChannelList = 0,8,200
;----------------------------------
; Automatic dialed numbers
;----------------------------------
TargetOfChannel0 = 100
TargetOfChannel1 = 101
TargetOfChannel2 = 102
TargetOfChannel3 = 103
TargetOfChannel4 = 104
TargetOfChannel5 = 105
TargetOfChannel6 = 106
TargetOfChannel7 = 107


;----------------------------------
; Phones to IP routing table
;----------------------------------
Prefix = 10,10.1.10.3
```

Reader's Notes

CHAPTER 6: PROVISIONING

> *All VCX V7111 Gateways have similar functionality except for the number of channels (the VCX V7111 24 FXS and 2FXS support only FXS), and all versions are referred to collectively in this document as the VCX V7111.*
>
> *VCX V7111 FXS refers only to the 24FXS, 8FXS, 4FXS, and 2FXS Gateways.*
>
> *VCX V7111 FXO refers only to 8FXO and 4FXO Gateways.*

Provisioning for SIP Operation

Initial configuration of the VCX V7111 is provided through a web browser or by loading of the mp108.ini configuration file. The configuration INI file can be downloaded with a web browser using HTTP or TFTP protocols or by using the 3Com configuration utility. The INI file name is provided in the field Boot File Name of the BootP server.

To create an INI file, it is recommended to use the Excel utility provided. To use the Excel utility, first install the Microsoft Office 2000 Excel application.

The INI file contains the following information:

- Basic and Logging Parameters shown in Table 17 on page 68.

- Channel Parameters shown in Table 18 on page 71.

- SIP parameters shown in Table 19 on page 76).

- Names for optional Call Progress Tone file. For detailed information, see "Using Call Progress Tones and Ringing" on page 93.

- Name for optional Telephony Interface (Coeff.dat) Configuration file. For the VCX V7111 FXS and VCX V7111 FXO two different files should be used. See "The coeff.dat Configuration File" on page 101 for more details.

> *The names of Call Progress and Coeff.dat files in* INI *file must be enclosed in quotation marks ('…').*

All INI file data is downloaded at startup and stored in non-volatile memory. The provisioning procedure should be used again only to modify VCX V7111 parameters; otherwise, BootP and TFTP is not needed again.

The Default Channel Parameters are applied to all VCX V7111 channels.

Users do not have to specify all parameters, as each unspecified parameter is set to its default value. Using the INI file resets all unspecified parameters to their default values.

The Channel Parameters define the DTMF/MF, Fax and Modem transfer modes. See "Appendix C: DTMF, Fax, and Modem Modes" on page 153 for a detailed description of these modes.

Basic, Logging, and Web Parameters

> In *Table 17, where there are parameters shown in brackets they refer to the format in the Embedded Web Server *.*

Table 17   Basic and Logging Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
MGControlProtocolType	8 = for SIP Gateway
DSPVersionTemplateNumber	0 = Firmware DSP version supports PCM/ADPCM, G723 and G729 Coders (default) 1 = Firmware DSP version supports PCM/ADPCM, and NetCoder coders 2 = Same as 0 but with voice and energy detectors 3 = Same as 1 but with voice and energy detectors DSP templates 2 or 3 should be selected for FXO Gateway Disconnect supervision feature, enabled by: EnableSilenceDisconnect = 1
EthernetPhyConfiguration	0 = 10 Base-T half-duplex 1 = 10 Base-T full-duplex 2 = 100 Base-T half-duplex 3 = 100 Base-T full-duplex 4 = auto-negotiate (Default) Auto-negotiate falls back to half-duplex mode (HD) when the opposite port is not in auto-negotiate, but the speed (10 Base-T, 100 Base-T) in this mode is always configured correctly.
DNSPriServerIP (DNS Primary Server IP)	IP address of primary DNS server
DNSSecServerIP (DNS Secondary Server IP)	IP address of secondary DNS server

Table 17   Basic and Logging Parameters

INI File Field Name GUI Parameter Name*	Valid Range and Description
DHCPEnable (Enable DHCP)	0 – Disable (default) 1 – Enable After the Gateway is powered up it will try first to communicate with BootP server; if BootP server is not responding and DHCPEnable =1 the Gateway will send DHCP request to configure its IP address and other network parameters from enterprise DHCP server. The DHCPEnable is special Hidden parameter. Once defined, and saved in flash it will be used even if it does not appear next time in INI file.
BootPRetries	1 = Single BootP request 2 = 2 BootP retries - (3 seconds) 3 = 3 BootP retries - (default, 6 seconds) 4 = 10 BootP retries - (30 seconds) 5 = 20 BootP retries - (60 seconds) 6 = 40 BootP retries - (120 seconds) 7 = 100 BootP retries - (300 seconds) 15 = BootP retries forever. Number of BootP retries, and then DHCP retries (if DHCPEnable = 1) at Gateway startup. The BootPRetries is special Hidden parameter. Once defined, and saved in flash it will be used even if it does not appear next time in INI file. Note that BootPRetries parameter becomes active after the VCX V7111 is reset and INI file is loaded. To change the parameters, first modify the INI file, and then reset the Gateway.
EnableDiagnostics	0 = No diagnostics (default) 1 = Perform diagnostics
WatchDogStatus	0 = Disable Gateway's watchdog 1 = Enable Gateway's watchdog (default)
EnableLanWatchDog	0 – Disable LAN Watchdog (default) 1 – Enable LAN Watchdog VCX V7111 restarts if LAN failure is detected

Table 17   Basic and Logging Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
SysLogServerIP (Network Settings/SysLog Server IP Address)	IP address in dotted format notation, for example, 192.10.1.255
EnableSyslog (Network Settings/Enable SysLog)	0 = Disable SysLog (default) 1 = Enable SysLog If SysLog is disabled all Logs and error messages are sent to RS-232 serial port if DisableRS232 = 0
DisableRS232	0 – RS-232 serial port is enabled (default) 1 – RS-232 serial port is disabled To enable sending of all log and error messages to the RS-232 serial port, define EnableSyslog = 0 and DisableRS232 = 0.
LoggerFormat	0 = name + msg 1 = time + msg 2 = name + time + msg 3 = SysLog prefix + msg (default)
DisableWebTask	0 = Enable Web management (default) 1 = Disable Web management
ResetWebPassword	Allows resetting to default of Web password to: Username: Admin Password: Admin
Disable WebConfig	0 = Enable changing parameters from Web (default) 1 = Operate Web server in read only mode
SNMPManagerIP (Network Settings/SNMP Manager IP)	IP address of SNMP Manager. The SNMP manager is used for receiving SNMP Traps. For example: SNMPManagerIP = 10.2.1.10
DisableSNMP (Network/Settings/Enable SNMP)	0 = SNMP is enabled (default) 1 = SNMP is disabled
HTTPport	HTTP port used for Web management (default = 80)

Channel Parameters

> In _Table 18_, parameters that are shown in brackets refer to the format in the Embedded Web Server [*].

Table 18   Channel Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
DJBufMinDelay (Dynamic Jitter Buffer Minimum Delay)	0 – 150 ms (default = 70) Dynamic Jitter Buffer Minimum Delay (described in the 3Com _VoIPLib Reference Library User Manual_).
DJBufOptFactor (Dynamic Jitter Buffer Optimization Factor)	0 – 12 (default = 7) Dynamic jitter buffer frame error/delay optimization.
BaseUDPPort (RTP Base UDP Port)	Range 6000 –64000 (default 6000) Lower boundary of UDP ports to be used by the Gateway for RTP, RTCP and T.38 channels. The upper boundary is the BaseUDPPort + 10*(number of Gateway's channels). For details see "Appendix A: BootP/TFTP Configuration Utility" on page 137.
ECHybridLoss	0 = 6 dB (default) 1 = 9 dB 2 = 0 dB 3 = 3 dB Sets the four wire to two wire worst case hybrid loss, the ratio between the signal level sent to the hybrid and the echo level returning from the hybrid.
FaxModemBypassM (Fax Modem Bypass Packing Factor)	Number of 20 ms payloads to be used for generating one RTP fax/modem bypass packet. (Range 1–2, default = 1)
FaxModemRelayVolume	-18 to -3, corresponding to -18 dBm to -3 dBm in 1 dB steps. (Default = -12 dBm) Fax gain control.
FaxRelayECMEnable (Fax Relay ECM Enabled)	0 = Disable using ECM mode during Fax Relay 1 = Enable using ECM mode during Fax Relay. (default)

Table 18   Channel Parameters

INI File Field Name GUI Parameter Name*	Valid Range and Description
FaxRelayEnhanced RedundancyDepth (Fax Relay Enhanced Redundancy Depth)	0 - 4 (default =2) Number of repetitions applied to control packets when using T.38 standard.
FaxRelayRedundancyDepth (Fax Relay Redundancy Depth)	0 – 2 (default =0) Number of repetitions to be applied to each fax relay payload when transmitting to network (applicable only when T38ProtectionMode = 0).
FaxRelayMaxRate (Fax Relay Max Rate (bps))	Limits the maximum rate at which fax messages are transmitted. 0 = 2.4 Kbps 1 = 4.8 Kbps 2 = 7.2 Kbps 3 = 9.6 Kbps 4 = 12.0 Kbps 5 = 14.4 Kbps, (default)
FaxTransportMode (Fax transport Mode)	Sets the Fax transport 0 = disable 1 = relay, (default) 2 = bypass
UseT38orFRF11	0 = Use proprietary FRF.11 syntax to send/receive fax relay. 1 = Use T.38 protocol to send/receive fax relay, (default).
V21ModemTransportType (V21 Transport Type)	0 = Transparent, (default) 2 = ModemBypass
V22ModemTransportType (V22 Transport Type)	0 = Transparent 2 = ModemBypass, (default)
V23ModemTransportType (V23 Transport Type)	0 = Transparent 2 = ModemBypass, (default)

Table 18   Channel Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
V32ModemTransportType (For V.32 and V.32bis modems) (V32 Transport Type)	0 = Transparent 2 = ModemBypass, (default)
V34ModemTransportType (For V.34 and V.90 modems) (V34 Transport Type)	0 = Transparent 2 = ModemBypass, (default)
FaxModemBypassCoderType (Fax/Modem Bypass Coder Type)	Coder to be used while performing fax/modem bypass. See acTCoders enumeration. Usually, high bit rate coders such as G.711 and G.726/G.727 should be used. 0 = G711 A-law =0, (default) 1 = G711 μ-law=1 4 = G726_32 11 = G727_32
FaxBypassPayloadType (Fax Bypass Payload Type)	Fax Bypass RTP dynamic payload type (Default = 102)
T38ProtectionMode	0 = Use redundancy packets for protecting T.38 fax relay stream, (default) 1 = Use Forward Error Correction (FEC) algorithm to protect T.38 fax relay stream (is not implemented)
DTMFVolume (DTMF Volume)	-31 to 0, corresponding to -31 dBm to 0 dBm in 1 dB steps (default = -11 dBm) DTMF gain control.
DTMFTransportType (DTMF Transport Type)	0 = erase digit from voice stream, do not relayed to remote. 1 = erase digit from voice stream, relay to remote. (Default) 2 = digits remains in voice stream. 3 = erase digit from voice stream, relay to remote according to RFC 2833 standard. 7 = digits are sent using RFC 2833 standard, but received RFC 2833 digits are muted from the audio stream.

Table 18   Channel Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
MFTransportType (MF Transport Type)	0 = erase MFs from voice transport, not relayed to remote. 1 = erase MFs from voice transport, relay to remote. (default) 2 = MFs are not erased from voice, not relayed to remote.
RFC2833PayloadType (RFC 2833 Payload Type)	The RFC 2833 DTMF relay dynamic payload type. Range: 96 – 99, 106 – 127; Default = 96 The 100, 102 – 105 range is allocated for 3Com proprietary usage. The same payload type should be used for receive and transmit.
InputGain (Input Gain)	-31 to 31 corresponding to -31 dB to +31 dB in 1 dB steps. (Default = 1 dB). PCM input gain.
RTPRedundancyDepth (RTP Redundancy Depth)	0 = Disable redundancy packets generation (default). 1 = Enable generation of RFC 2198 redundancy packets.
VoiceVolume (Voice Volume)	-31 to 31, corresponding to -31 dB to +31 dB in 1 dB steps (Default = 1 dB). Voice gain control
M (Packing Factor)	Number of codec payloads (5, 10, 20, or 30 ms, depending on selected codec) to be used for generating one RTP packet. M = n payloads (n = 1, 2, or 3); M = 1 (default)
SCE (Silence Suppression)	0 = silence compression disabled (default) 1 = silence compression enabled
ECE (Echo Canceller)	0 = Echo Canceler disabled 1 = Echo Canceler Enabled (default)
IPDiffServ (Network Settings/RTP IP Diff. Serv)	0 – 63 value for setting the Diff Services Code Point (DSCP). If defined it will override the IP TOS and IP Precedence settings. Applies only to RTP packets.
IPPrecedence (Network Settings/RTP IP Precedence)	0 – 7 (default 0) Sets the value of the IP precedence field in the IP header for all RTP packets

Table 18   Channel Parameters

INI File Field Name GUI Parameter Name*	Valid Range and Description
IPTOS (Network Settings/RTP IP TOS)	0 – 15 (default 0) Sets the value of the IP Type Of Service field in the IP header for all RTP packets
FlashHookPeriod	300 – 1500 (default 700) Flash Hook time in ms. The parameter is used for Flash Hook detection in VCX V7111 FXS Gateways and for Flash Hook generation in VCX V7111 FXO Gateways.
MinimumFlashHookTime	25 – 300, (default = 250) Minimum threshold in ms + 50ms for detection of HookFlash. Relevant only for VCX V71111 FXS Gateways.
DTMFDetectionPoint	0 = DTMF event is reported when button is pressed 1 = DTMF event is reported on button release (default) The parameter is used for out of band dialing (using SIP INFO messages)
DTMFDigitLength	Time in ms for generating DTMF to TEL side Default = 100 ms
DTMFInterDigitInterval	Time in ms between generated DTMFs to TEL side Default = 100 ms
CallerIDType (Caller ID Type)	0 = Bellcore GR-30-CORE Type 1(default) 1 = ETSI Type 1 Both Bellcore and ETSI Type 1 Caller ID signals are generated/detected between the first and the second rings.
TestMode	0 = CoderLoopback, encoder-decoder loopback inside DSP. 1 = PCMLoopback, loopback the incoming PCM to the outgoing PCM. 2 = ToneInjection, generates a 1000 Hz tone to outgoing PCM. 3 = NoLoopback, (default).

SIP Parameters

> In *Table 19*, parameters shown in brackets refer to the format in the Embedded Web Server *.

Table 19   SIP Parameters

INI File Field Name GUI Parameter Name*	Valid Range and Description
GatewayVersion (Version ID)	Version of Gateway, for example, GatewayVersion = 4.2 Beta.
SIPGatewayName (Gateway Name)	VCX V7111 Gateway Domain Name, if specified the name is used in right side of SIP URL. If not specified, the Gateway IP address is used instead (default).
StaticNatIP (NAT IP Address)	Static NAT IP address. Global Gateway IP address. Define if static Network Address Translation (NAT) device is located between the Gateway and the Internet.
IsProxyUsed (Enable Proxy)	0 = no Proxy used [internal phones table used] (default) 1 = Proxy is used
ProxyIp ProxyIP ProxyIP (Proxy IP)	IP addresses (or fully qualified domain name) of main and optionally two redundant SIP outbound Proxies. Used if IsProxyUsed = 1 The Gateway starts working with the main Proxy, in case of a failure; the Gateway tries to communicate with the first redundant proxy (second on the list). Once a proxy is found, the Gateway will work with it, until the next failure occurs. If none of the proxies respond, the Gateway goes over the list again. The Gateway monitors connection with a proxy using keep-alive messages (OPTIONS).
ProxyName (Proxy Name)	Home Proxy Domain Name. If specified, the name is used as Request-URI in REGISTER, INVITE, and other SIP messages. If the proxy name is not specified, the Proxy IP address is used instead.
EnableProxyKeepAlive (Enable Proxy Keep Alive)	0 = Disable (default) 1 = Keep alive with Proxy, every 60 seconds sends an OPTIONS SIP message

Table 19 SIP Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
UserName (User Name)	User name used for Registration and for BASIC/DIGEST authentication process with Proxy. It is applicable only for single gateway registration, if AuthenticationMode = 1
Password (Password)	Password used for BASIC/DIGEST authentication process with Proxy. Single password is used for all Gateway ports.
Cnonce (Cnonce)	String used by the server and client to provide mutual authentication. (Free format that is Cnonce = 0a4f113b)
IsRegisterNeeded (Enable Registration)	0 = Gateway will not register to Proxy or to Registrar (default) 1 = Gateway will register to Proxy or to Registrar at power up The Gateway registers according to AuthenticationMode parameter.
RegistrarIP (Registrar IP)	IP address of Registrar server (optional). If not specified, the Gateway sends the Register request to IP address of the proxy server (ProxyIP).
RegistrationTime (Registration Time)	Registration expired timeout (seconds). The value is used in Expires = header. Typically a value of 3600 is assigned, for one hour registration. The Gateway resumes registration before the timeout expires.
AuthenticationMode	0 = Registration and Authentication separately for each Endpoint. 1 = Single Registration and Authentication for the Gateway. 2 = Registration and Authentication according to selection of IsUseFreeChannel parameter (default). Usually Authentication, on a per Endpoint basis, is used for FXS Gateways in which each Endpoint registers (and authenticates) separately with its own username and password. Single Registration and Authentication (AuthenticationMode=1) is usually defined for FXO Gateways. Selecting AuthenticationMode=2 and IsUseFreeChannel=0 leads to a separate registration of each Endpoint. Selecting AuthenticationMode=2 and IsUseFreeChannel=1 leads to a single Gateway registration.

Table 19　SIP Parameters

INI File Field Name GUI Parameter Name*	Valid Range and Description
Authentication_x	Username, Password for each VCX V7111 FXS Gateway port x = Gateway port, range 0 – 23 Authentication_0 = david,14325 Authentication_1 = Alex,18552 The parameter enables the user to specify, for each port, a username-password combination for authentication purposes. If not specified, the port's phone number and global password are used instead of username and password respectively. Using the sign $$ enables the user to omit the username. For instance, Authentication_5 = $$, 18552. In this case the Endpoint's phone number is used instead.
EnableEarlyMedia (Enable Early Media)	0 = Early Media is disabled (default) 1 = Enable Early Media If enabled, the VCX V7165 Digital Media Server sends a 183 Session Progress response (instead of 180 ringing), allowing the setup of the media stream prior to the answering of the call.
IsPrackRequired (Enable PRACK)	0 = PRACK is not used (default) 1 = PRACK is used by Gateway
SipSessionExpires (Session-Expires Time)	0 = not activated (default) Timeout [seconds] for Keeping a re-INVITE message alive within a SIP session
EnableRPIheader	0 = Disable (default) 1 = RPI header is generated in SIP INVITE message for both called and calling numbers.
EnableTransfer (Enable Transfer)	0 = Call transfer is not allowed 1 = The Gateway initiates Call Transfer (using REFER) on Hook-Flash
EnableHold (Enable Hold)	0 = Hold service disabled (default) 1 = Hold (or unhold) is activated in FXS Gateway using HookFlash signaling
SIPDestinationPort (SIP Destination Port)	SIP UDP destination port for sending SIP messages. Default = 5060
LocalSIPPort	Local UDP port used to receive SIP messages (default = 5060)

Table 19   SIP Parameters

INI File Field Name GUI Parameter Name*	Valid Range and Description
SecureCallsFromIP (IP Security)	0 = Accept all SIP Calls (default) 1 = Accept SIP calls only from the IP addresses which are defined in the Prefix Routing table. Gateway rejects calls from unknown IP addresses. Specifying the IP address of a proxy server in the Prefix Routing tables enables the Gateway to only accept calls originating in the Proxy server and rejects all other calls.
IsUserPhone (Use user=phone in SIP URL)	0 = Does not use user=phone string in SIP URL 1 = user=phone string is part of the SIP URL (default)
IsSpecialDigits	0 = # digit will terminate DTMF dialing (Default) 1 = # digit will not terminate dialing
SipT1Rtx (SIP T1 Retransmission Timer [ms])	Timer T1 value for retransmission in ms. SipT1Rtx = 500
SipT2Rtx (SIP T2 Retransmission Timer [ms])	Timer T2 value for retransmission in ms. SipT2Rtx = 4000

Table 19   SIP Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
CoderName (Coders)	Supported Coders g711Ulaw64k G.711 μ-law, 20 ms g711Alaw64k G.711 A-law, 20 ms g711Ulaw64k,5 G.711 μ-law, 5 ms g711Alaw64k,5 G.711 A-law, 5 ms g711Ulaw64k,10 G.711 μ-law, 10 ms g711Alaw64k,10 G.711 A-law, 10 ms g7231 G.723 6.3 Kbps (default) g7231r53 G.723 5.3 Kbps g726 G.726 ADPCM 16 Kbps (Payload Type = 35) g726r16 G.726 ADPCM 16 Kbps, Cisco mode (Payload Type = 23) g726r32 G.726 ADPCM 32 Kbps (Payload Type = 2) g729 G.729A, 20 ms g729,10 G.729A, 10 ms NetCoder6_4 NetCoder 6.4 Kbps NetCoder7_2 NetCoder 7.2 Kbps NetCoder8 NetCoder 8.0 Kbps NetCoder8_8 NetCoder 8.8 Kbps This parameter can appear up to five times.
TrunkGroup_x (Hunt Group Table)	TrunkGroup_x = a-b, c a, b, c or just a, c x = Hunt group ID, starting with 1 a = Starting port b = Ending port (up to 24 for 24FXS) c = phone number allocated for the first port For example: TrunkGroup_1 = 1–4,100 TrunkGroup_2 = 5–8,200

Table 19   SIP Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
PSTNPrefix (IP to Trunk group routing table)	PSTNPrefix = a, b a = Destination number prefix (for IP→Tel calls) b = Hunt group ID Outgoing IP→Tel calls starting with the prefix a are routed to Hunt group number b.
RemovePrefix	0 = Do not remove prefix (default) 1 = Remove PSTN prefix (defined in the routing table) from a telephone number of an incoming IP call, before forwarding it to an Analog port. Applicable only if number manipulation is performed after call routing for IP→Tel calls (RouteModeIP2Tel = 0).
ChannelList	List of phone numbers for VCX V7111 channels a, b, c a = first channel b = number of channels starting from a c = the phone number of the first channel example: ChannelList = 0,8,101 Defines phone numbers 101 – 108 for up to eight VCX V7111 8FXS or 8FXO channels. The INI file can include up to ten ChannelList = entries The ChannelList = can be used instead or in addition to Channel2Phone parameter.
Channel2Phone	Phone number of channel. Its format: Channel2Phone = <channel>, <number> <channel> is 0...23. Example: Channel2Phone = 0, 1002 Appears once for each channel: eight times for VCX V7111 8FXS or 8FXO Gateways, four times for VCX V7111 4FXS or 4FXO Gateways, and twice for VCX V7111 2FXS Gateways. For 8-port and 24-port Gateways it is suggested to use ChannelList = parameter, where in a single line, all Gateway's phone numbers can be defined. The Channel2Phone can be used instead or in addition to ChannelList = parameter.

Table 19 SIP Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
Prefix	Mapping phone number to IP address, using phone number prefix Example: Prefix = 20,10.2.10.2 Any dialed number that starts with 20 is routed to IP address 10.2.10.2. Needed when Proxy server is not used. Can appear up to 20 times. Maximal prefix size is seven digits.
NumberMapTel2IP (Manipulation Tables: TEL → IP Destination Numbers)	NumberMapTel2IP = a,b,c,d a = Phone number prefix b = Number of digits to remove c = New prefix to be added, or none d = Number of digits to leave, from the right The parameter defines set of rule(s) for changing the destination phone number of incoming Tel→ IP calls. Each set of rules applies to calls which their telephone number correspond with the prefix a. The b, c and d actions apply to that number. The actions are executed in the following order: b, d, and then c.
NumberMapIP2Tel (Manipulation Tables: IP → TEL Destination Numbers)	NumberMapIP2Tel = a,b,c,d a = Phone number prefix b = Number of digits to remove c = New prefix to be added, or none d = Number of digits to leave, from the right The parameter defines set of rule(s) for changing the destination phone number of outgoing IP→Tel calls. Each set of rules applies to calls which their telephone number correspond with the prefix a. The b, c and d actions apply to that number. The actions are executed in the following order: b, d, and then c.
SourceNumberMapTEL2IP (Manipulation Tables: TEL → IP Source Numbers)	Identical rules to those of destination number
SourceNumberMapIP2TEL (Manipulation Tables: IP → TEL Source Numbers)	Identical rules to those of destination number

Table 19 SIP Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
AddTrunkGroupAsPrefix (Add Trunk Group ID as Prefix)	0 = not used 1 = For Tel→IP incoming calls, Hunt Group ID is added as a prefix to destination phone number. Applicable only if Hunt groups ID are defined. Enable users to define various routing rules based on the hunt group from which the call arrived.
RouteModeIP2Tel (IP to TEL routing mode)	0 = Route calls before number manipulation (default) 1 = Route calls after number manipulation Defines the execution order of number manipulation and routing rules, for outgoing IP→Tel calls.
RouteModeTel2IP (TEL to IP routing mode)	0 = Route calls before number manipulation (default) 1 = Route calls after number manipulation Defines the execution order of destination number manipulation as opposed to the routing rules, for incoming Tel→IP calls. Not applicable if proxy routing is used.
PlayRBTone2Tel (Play Ringback Tone to TEL)	0 = Do not play if 183 response was received 1 = Play 2 = Play according to 180/183 response If 183 session progress with SDP message was received, the Gateway cuts through the voice channel and does not play ringback tone.
PlayRBTone2IP (Play Ringback Tone to IP)	0 = Do not play (default) 1 = The Gateway plays ringback tone to IP after sending SIP 183 session progress response.
IsDialNeeded (Enable Automatic Dialing)	0 = no dial needed (automatic dialing) 1 = dial needed (default) If 0 = TargetOfChannel parameters define the automatic dialed number. This parameter is applicable for both FXS and FXO Gateways. If DialisNeeded =1 the FXO Gateway will seize the line (after detecting the ringing signal), play a dial tone, collect DTMF digits and send INVITE to IP destination.

Table 19   SIP Parameters

INI File Field Name GUI Parameter Name*	Valid Range and Description
TargetOfChannel# (Automatic Dialing Table)	Automatic dialed phone number. The automatic dialed number, used if OFF HOOK detected in FXS channel, or ringing signal is detected in FXO channel. Applicable when IsDialNeeded = 0. Its format: TargetOfChannel<channel> = <number>. Example: TargetOfChannel1 = 123 The parameter, if used, should be defined per Gateway FXS or FXO port (channel).
IsUseFreeChannel (Selects Next Available Channel)	0 = Select the FXO channel according to destination phone number received in INVITE message, (default) 1 = Select the next available FXO channel Used for IP→ VCX V7111 FXO calls The next available FXO channel is selected, out of the Gateway channels defined in Hunt group table. When using one stage dialing, (IsTwoStageDial =0), IsUseFreeChannel should be equal to 1. For one stage dialing the VCX V7111 FXO selects the next free channel, and dials into the FXO line the destination phone number received in INVITE message.
IsTwoStageDial	0 = One stage dialing 1 = Two Stage Dialing (default) Used for IP → VCX V7111 FXO calls For Two Stage Dialing the VCX V7111 FXO seizes the PSTN/PBX line, without performing any dial, the remote User is connected over IP to PSTN/PBX, and all further signaling (dialing and call progress tones) is done directly with the PBX without Gateway intervention. For One Stage Dialing VCX V7111 FXO seizes the next available channel (IsUseFreeChannel should be 1), and dials the destination phone number received in INVITE message. Use the IsWaitForDialTone parameter to specify whether the dialing should come after detection of dial tone, or immediately after seizing of the line.

Table 19   SIP Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
IsWaitForDialTone	0 = do not wait for dial tone
	1 = Wait for dial tone (default)
	Used for VCX V7111 FXO, for One Stage Dialing.
	If IsWaitForDialTone = 0, VCX V7111 FXO dials phone number immediately after seizing the PSTN/PBX line, without listening to dial tone.
	If IsWaitForDialTone = 1, VCX V7111 FXO dials phone number only after it detects a dial tone (it can take 3-5 seconds to detect a dial tone).
	The correct dial tone parameters should be configured in call progress tone file.
FXOWaitForDialTime	FXO dialing timeout [in ms], after seizing the line and before start dialing. (Default =1000)
	Applicable for VCX V7111 FXO for single stage dialing.
EnableSilenceDisconnect	1 = Enables FXO Gateways to disconnect calls which have silence in both directions for more than FarEndDisconnectSilencePeriod consecutive seconds.
	0 = Feature disabled (default)
FarEndDisconnectSilencePeriod	Duration of silence period (in seconds) prior to call disconnection. (Default = 120). Applicable to FXO Gateways, which use DSP templates 3 or 4.
	Range 10–1000 seconds, with a 5 seconds deviation.
MaxDigits (Max Digits in Phone Number)	2 – 19 (default 4). Maximum number of digits that can be dialed. Dialing ends when maximum number of digits dialed or timeout between digits expired (TimeBetweenDigits parameter), or # is dialed.
TimeBetweenDigits (Interdigit Timeout [seconds])	0 – 5 (default 4) Inter-digit timeout in seconds, used to terminate dialed numbers.
IsDTMFUsed (Use INFO for DTMF)	0 = not used
	1 = INFO message is used to transfer DTMF digits, implementing IETF draft draft-choudhuri-sip-info-digit-00
	To use out of band DTMF transfer, disable in band DTMF (DTMFTransportType=0, erase digit and do not relay).

Table 19   SIP Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
InfoDTMFFormat (Use Info for DTMF)	1 = Sends DTMF digits according with IETF draft-choudhuri-sip-info-digit-00 2 = Sends DTMF digits according with Cisco format (default) Currently the Cisco method of sending DTMF digits in INFO messages is supported only for incoming Tel → IP calls.
IsHookFlashUsed (Use Info for Hook-Flash)	0 = not used (default) 1 = Sends INFO message with HookFlash indication when Hookflash is detected Received INFO message with HookFlash signal generates HookFlash at FXO channel
CallerDisplayInfo# (Caller ID table)	Caller DisplayInfo table is used to send Caller Identification information per FXS Gateway port to remote IP terminal. This parameter can appear up to eight times (for 8FXS or 8FXO GatewayS), with up to 18 characters per string. For the 24FXS, this parameter can appear up to 24 times with up to ten characters per string. For example: CallerDisplayInfo0 = 3Com 1001 CallerDisplayInfo1 = 3Com 1002 ; ; CallerDisplayInfo8 = 3Com 1008
EnableCallerID (Enable Caller ID)	0 = Do not send CallerID signal to Gateway's FXS port (default) 1 = Calling number and Display text is sent to Gateway FXS port, between first and second rings, to be displayed on phone's caller ID display for incoming call. In FXO Gateway if EnableCallerID=1, the Caller ID signal will be detected and send to IP in SIP INVITE message (as Display element).
EnableReversalPolarity (Enable Polarity Reversal)	0 = The line polarity is not changed on answer (default) 1 = The line polarity is changed on call answer and then changed back on call release Applicable for VCX V7111 FXS Gateways.

Table 19   SIP Parameters

INI File Field Name GUI Parameter Name[*]	Valid Range and Description
EnableCurrentDisconnect (Enable Current Disconnect)	0 = Current disconnect is disabled (default) 1 = FXS Gateway generates a current disconnect pulse of 900 ms after a call is released from IP Applicable only to VCX V7111 FXS Gateways.
TimeForReorderTone	Duration of played reorder tone in seconds (default 5 seconds). Applicable for FXO port. The tone is played before releasing the FXO line.
TimeForDialTone (Dial Tone Duration [seconds])	Duration of played dial tone (default 16 seconds). The dial tone is played at FXS Gateway port, after phone is picked up, or after the FXO Gateway seizes the line in respond to ringing. During play of the dial tone, Gateway waits for DTMF digits. Applicable for both FXS and FXO Gateways when Automatic dialing feature is disabled, IsDialNeeded = 0.
MaxActiveCalls	The parameter is used to limit number of calls in the Gateway. When number of concurrent calls reaches MaxActiveCalls, the Gateway rejects calls from both Tel and IP.
EnableCDR (Enable Syslog CDR)	0 = CDR is not used 1 = Call Detail Record is sent to the Syslog server at the end of each call The CDR Syslog message complies with RFC 3161 and is identified by: Facility = 17 (local1) and Severity = 6 (Informational)
EnableBusyOut	0 = Not used (default) 1 = If LAN or Proxy are not responding, the Gateway plays a reorder tone when the phone is OffHooked.

Loading Configuration Files

You can use the INI file to specify Call Progress Tone table files and Line Characteristics control file to be downloaded to the VCX V7111 during the configuration phase, either directly from the web browser or by using the TFTP procedure. It is also possible to define whether the downloaded files are stored in non-volatile memory so the TFTP process is not required every time the Gateway boots up.

The following INI file fields are related to this operation:

- **CallProgressTonesFilename** – The name (and path) of the file containing the call progress tones definition. See "Call Progress Tone and Ringing Generation and Download Procedure" on page 99 for additional information on how to create and download this file.

- **FXSCoefFileName** – The name (and path) of the file providing the FXS line characteristic parameters.

- **FXOCoefFileName** – The name (and path) of the file providing the FXO line characteristic parameters.

- **BurnCallProgressTonesFile** – Stores the call progress tones configuration file in non-volatile memory, if set to 1.

- **BurnCoefFile** – Stores the line characteristics file in non-volatile memory, if set to 1.

The INI File Structure

The INI file can contain any number of parameters. The parameters are divided into groups by their functionality. The general form of the INI file is shown in the following figure.

Figure 29   INI File Structure

```
[Sub Section Name]

Parameter_Name = Parameter_Value
Parameter_Name = Parameter_Value
.
..

; REMARK

 [Sub Section Name]
```

The INI File Structure Rules

- Lines beginning with a semi-colon (;) (as the first character) are ignored.

- Carriage Return must be the final character of each line.

- Number of spaces before and after = is not relevant.

- If there is a syntax error in the parameter name, the value is ignored.

- Syntax errors in the parameter value field can cause unexpected errors (because parameters may be set to the wrong values).

- Sub-section names are optional.

- The File name String parameters, should be placed between two inverted commas ('…'). For example CallProgressTonesFileName = 'cpusa.dat'.

- The parameter field is NOT case sensitive.

- Parameter values should be entered only in decimal format, except for the Call Agent IP address.

- The INI file should be ended with one or more carriage returns.

- [Files] line should precede the CallProgressTonesFileName, and these lines, if used, should be placed at the end of INI file; (see the INI file example, <u>Figure 30</u>).

The INI File Example

An example of an INI file for an SIP Gateway is shown in the following figure.

Figure 30 SIP INI File Example

```
MGControlProtocolType = 8

[Channel Params]
DJBufferMinDelay = 75
RTPRedundancyDepth = 1

IsProxyUsed = 1
ProxyIp = 192.168.122.179
MaxDigits = 3
CoderName = g7231

; Phone of each end point
Channel2Phone = 0, 101
Channel2Phone = 1, 102
Channel2Phone = 2, 103
Channel2Phone = 3, 104

EnableSyslog = 0
LoggerFormat = 0

[Files]

CallProgressTonesFilename = 'CPUSA.dat'
BurnCallProgressTonesFile = 1
 FXSCOEFFILENAME = 'coeff.dat'
 BurnCoefFile = 1
```

> *Using Windows Properties Display, verify that the MS-DOS name of the INI file is in fact mp108.ini, and* not *by mistake mp108.ini.ini, or mp108~.ini.*
>
> *To restore VCX V7111 default configuration parameters, use the mp1xx.ini file without any valid parameters or with semicolon (;) character preceding all lines in the file.*

Excel Utility for INI File Generation

The Excel Utility enables easy generation of VCX V7111 Gateway INI files. To use the Excel utility, first install the Microsoft Office 2000 Excel application.
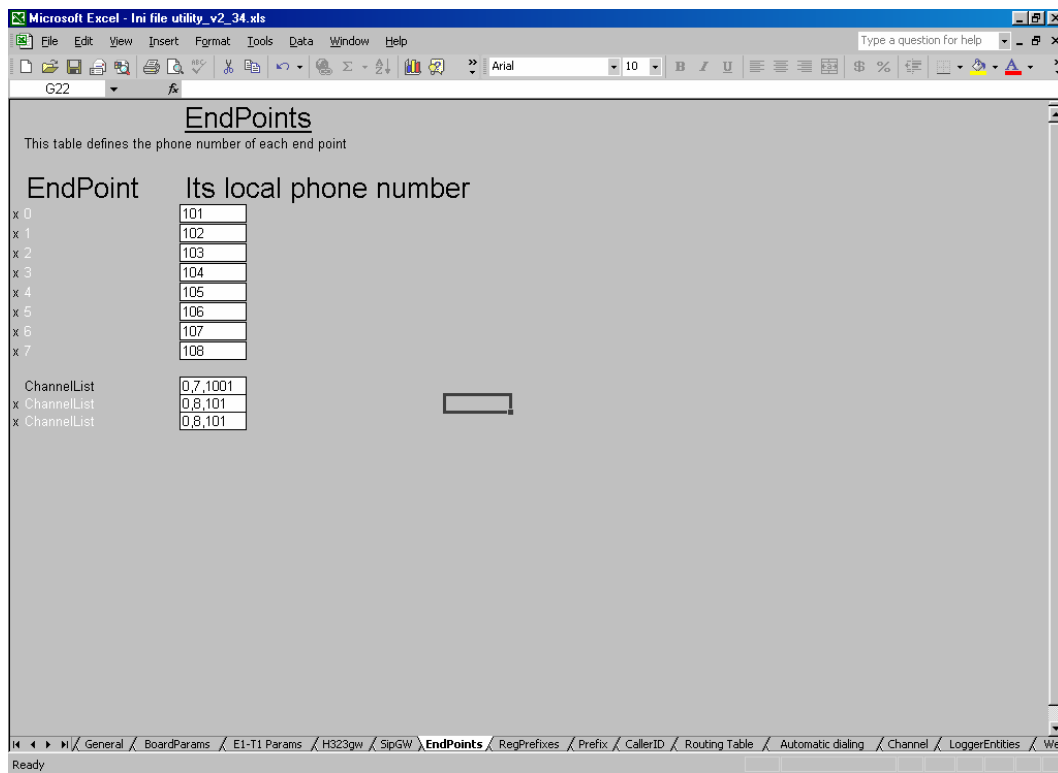
General Data Sheet

Figure 31 General Data Sheet

End Points Page

Figure 32   End Points Page

Phones to IP Routing Table

Figure 33   Phones to IP Routing Table

Using Call Progress Tones and Ringing

The Call Progress Tones Configuration File contains the definitions of the call progress tones and characteristics of ringing signal to be detected/generated by the VCX 7111. You can use either VCX V7111, one of the configuration files supplied by 3Com, or construct their own file.

The Call Progress Tones Configuration File used by the VCX V7111 is a binary file (with the extension DAT). You can construct your own configuration file by starting from the tone.ini file format, then modifying the file, and finally converting it into binary format using the Download conversion utility supplied with the VCX V7111 package. Select the *Convert dBm Values* check box in the Conversion Utility.

To download the Call Progress Tones File to the VCX V7111, a correct definition should be used in the mp108.ini file. See "Call Progress Tone and Ringing Generation and Download Procedure" on page 99 for the description of the procedure on how to generate and download the Call Progress Tones file.

Format of the Call Progress INI File

The Call Progress Tones section of the INI file format starts from the following string:

- **[NUMBER OF CALL PROGRESS TONES]** – Number of Call Progress Tones defining the number of call progress tones to be defined in the file.

- **[CALL PROGRESS TONE #X]** – Containing the Xth tone definition (starting from 1 and not exceeding the number of call progress tones defined in the first section) using the following keys:

    - **Tone Type** – Call Progress tone type

        1 - Dial Tone

        2 - Ringback Tone

        3 - Busy Tone

        4 - Congestion Tone

        5 - Special Information Tone

        6 - Warning Tone

        7 - Reorder Tone

        8 - Confirmation Tone

        9 - Call Waiting Tone

        16 - Off Hook Warning Tone

        17 - Call Waiting Ringback Tone

        23 - Hold Tone

    - **Low Freq [Hz] –** Frequency in hertz of the lower tone component in case of dual frequency tone, or the frequency of the tone in case of single tone.

    - **High Freq [Hz]** – Frequency in hertz of the higher tone component in case of dual frequency tone, or zero (0) in case of single tone.

    - **Low Freq Level [-dBm]** – Generation level 0 dBm to –31 dBm in [dBm].

    - **High Freq Level** – Generation level 0 to –31 dBm. The value should be set to 32 in the case of a single tone.

    - **First Signal On Time [10 msec]** – Signal On period (in 10 ms units) for the first cadence on-off cycle.

    - **First Signal Off Time [10 msec]** – Signal Off period (in 10 ms units) for the first cadence on-off cycle.

    - **Second Signal On Time [10 msec]** – Signal On period (in 10 ms units) for the second cadence on-off cycle.

▪ **Second Signal Off Time [10 msec]** – Signal Off period (in 10 ms units) for the second cadence on-off cycle.

Using this configuration file, you can create up to 16 different call progress tones using up to 15 different frequencies (in the range from 300 Hz through 2000 Hz). Each one of the call progress tones is specified by the following two parameters: the tone frequency (either single or dual frequencies are supported) and the tone cadence. This is specified by two sets of ON/OFF periods, but you can discard the use of the first On/Off cycle by setting the relevant parameters to zero. When the tone is made up of a single frequency, the second frequency field should be set to zero.

For a continuous tone (such as dial tone), only the First Signal On time should be specified. In this case, the parameter specifies the detection period. For example if it equals 300, then the tone is detected after 3 seconds (300 x 10 ms).

> *When defining several continuous tones, the First Signal On Time parameter should have the same value for all tones.*
>
> *The tones frequency should differ by at least 40 Hz from one tone to other defined tones.*

Default Template for Call Progress Tones

The VCX V7111 is initialized with the default Call Progress Tones configuration template shown in Table 20. If you need to change one of the tones, edit the default call progress.txt file.

For example: to change the dial tone to 440 Hz only, replace the #Dial tone section in Table 20 with the following text:

```
#Dial tone
[CALL PROGRESS TONE #1]
Tone Type=1
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=10 (-10 dBm)
High Freq Level [-dBm]=32 (use 32 only if a single tone is required)
First Signal On Time [10msec]=300; the dial tone is detected after 3 sec
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
```

You can specify several tones of the same type using Tone Type definition. These additional tones are used only for tone detection. Generation of specific tone is according to the first definition of the specific tone. For example, you can define an additional dial tone by appending the second dial tone definition lines to the tone INI file. The VCX V7111 reports dial tone detection if either one of the two tones has been detected.

Table 20   Call Progress Tones Template

```
[NUMBER OF CALL PROGRESS TONES]
Number of Call Progress Tones=9
#Dial tone
[CALL PROGRESS TONE #0]
Tone Type=1
Low Freq [Hz]=350
High Freq [Hz]=440
Low Freq Level [-dBm]=13
High Freq Level [-dBm]=13
First Signal On Time [10msec]=300
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
#Dial tone
[CALL PROGRESS TONE #1]
Tone Type=1
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=10High Freq Level [-dBm]=32
First Signal On Time [10msec]=300
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
#Ringback
[CALL PROGRESS TONE #2]
Tone Type=2
Low Freq [Hz]=440
High Freq [Hz]=480
Low Freq Level [-dBm]=19
High Freq Level [-dBm]=19
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=200
Second Signal Off Time [10msec]=400
#Ringback
[CALL PROGRESS TONE #3]
Tone Type=2
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=16
High Freq Level [-dBm]=32
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=100
Second Signal Off Time [10msec]=300

#Busy
[CALL PROGRESS TONE #4]
Tone Type=3
Low Freq [Hz]=480
High Freq [Hz]=620
Low Freq Level [-dBm]=24
High Freq Level [-dBm]=24
First Signal On Time [10msec]=50
First Signal Off Time [10msec]=50
Second Signal On Time [10msec]=50
Second Signal Off Time [10msec]=50
```

Table 20   Call Progress Tones Template

```
#Busy
[CALL PROGRESS TONE #5]
Tone Type=3
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=20
High Freq Level [-dBm]=32
First Signal On Time [10msec]=50
First Signal Off Time [10msec]=50
Second Signal On Time [10msec]=50
Second Signal Off Time [10msec]=50
#Reorder tone
[CALL PROGRESS TONE #6]
Tone Type=7
Low Freq [Hz]=480
High Freq [Hz]=620
Low Freq Level [-dBm]=24
High Freq Level [-dBm]=24
First Signal On Time [10msec]=25
First Signal Off Time [10msec]=25
Second Signal On Time [10msec]=25
Second Signal Off Time [10msec]=25
#Confirmation tone
[CALL PROGRESS TONE #7]
Tone Type=8
Low Freq [Hz]=350
High Freq [Hz]=440
Low Freq Level [-dBm]=20
High Freq Level [-dBm]=20
First Signal On Time [10msec]=10
First Signal Off Time [10msec]=10
Second Signal On Time [10msec]=10
Second Signal Off Time [10msec]=10
#Call Waiting Tone
[CALL PROGRESS TONE #8]
Tone Type=9
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=20)
High Freq Level [-dBm]=32
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=30
Second Signal Off Time [10msec]=900
```

Format of the Ringing Definition

The ringing pattern configures the ringing tone frequency and up to four ringing cadences. It is applicable for VCX V7111 FXS Gateways. Only single ringing pattern can be defined, if not a default ringing pattern applies. The ringing frequency can be configured in the range from 10 Hz through 200 Hz with a 5 Hz resolution. Each ringing cadence period can be defined as single ringing burst. See the following examples.

The distinctive ringing section of the INI file format contains the following strings:

- [NUMBER OF DISTINCTIVE RINGING PATTERNS]

  - Number of Ringing patterns = 1

  - [Ringing Pattern #0]

  - Ring Type =0

  - **Freq [Hz]** – Frequency in hertz of the ringing tone.

  - **First Ring On Time [10 msec]** – Ring On period (in 10 ms units) for the first cadence On-Off cycle.

  - **First Ring Off Time [10 msec]** – Ring Off period (in 10 ms units) for the first cadence On-Off cycle.

  - **Second Ring On Time [10 msec]** – Ring On period (in 10 ms units) for the second cadence On-Off cycle.

  - **Second Ring Off Time [10 msec]** – Ring Off period (in 10 ms units) for the second cadence On-Off cycle.

  - **Third Ring On Time [10 msec]** – Ring On period (in 10 ms units) for the third cadence On-Off cycle.

  - **Third Ring Off Time [10 msec]** – Ring Off period (in 10 ms units) for the third cadence On-Off cycle.

  - **Fourth Ring On Time [10 msec]** – Ring Off period (in 10 ms units) for the fourth cadence On-Off cycle.

  - **Fourth Ring Off Time [10 msec]** – Ring Off period (in 10 ms units) for the fourth cadence On-Off cycle.

  - **Burst** – Configures the ringing signal to be a single ringing burst comprised of all specified cadences. The Burst string is defined per each ringing cadence and it must appear between First/Second/Third/Forth string and the Ring On/Off Time.

Examples of Various Ringing Signals

```
#Regular North American Ringing Pattern: 20 Hz, 2 sec On, 4 sec Off
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=20
First Ring On Time [10msec]=200
First Ring Off Time [10msec]=400

#GR-506-CORE Ringing Pattern 3: 20 Hz ringing comprised of three cadences
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=20
First Ring On Time [10msec]=40
First Ring Off Time [10msec]=20
Second Ring On Time [10msec]=40
Second Ring Off Time [10msec]=20
Third Ring On Time [10msec]=80
Third Ring Off Time [10msec]=400

#EN 300 001 Ring - Finland: informative ringing nr. 3: three ringing bursts
followed by repeated ringing of 1 sec on and 3 sec off.
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=25
First Burst Ring On Time [10msec]=30
First Burst Ring Off Time [10msec]=30
Second Burst Ring On Time [10msec]=30
Second Burst Ring Off Time [10msec]=30
Third Burst Ring On Time [10msec]=30
Third Burst Ring Off Time [10msec]=30
Fourth Ring On Time [10msec]=100
Fourth Ring Off Time [10msec]=400
```

Call Progress Tone and Ringing Generation and Download Procedure

To generate and download the Call Progress Tone file, follow these steps:

1 Prepare the tone.ini file including call progress tones and ringing parameters.

2 Use the Download conversion utility to generate binary tone.dat file. See Figure 34.

Figure 34   Download Selection Screen

3   Click *Process a new file*.

4   Select an input file such as usa_tone.ini and fill the Vendor and Version fields.

Figure 35   File Selection Screen

5   Select the *Convert Code value into dBms* check box.

6   Click *Make File* and then close the application.

7   Edit the mp-1xx.ini file and add the following two lines:

```
CallProgressTonesFilename = 'usa_tone.dat'
BurnCallProgressTonesFile = 1
```

8   Save the usa_tone.dat and mp108.ini files in TFTP folder.

9 Set the Boot file name in the BootP server: mp108.ini.

10 Activate the BootP and TFTP servers and reset the VCX V7111 Gateway (See "Provisioning for SIP Operation" on page 67, describing VCX V7111 provisioning).

The coeff.dat Configuration File

The purpose of the coeff.dat configuration file is to provide best feed and transmission quality adaptation for different phone line types. Two different coeff.dat files are needed for VCX V7111 FXS and FXO Gateways. The file consists of a set of parameters for the signal processor of the loop interface devices. This parameter set provides control of the following AC and DC interface parameters:

- DC (battery) feed characteristics

- AC impedance matching

- Transmit gain

- Receive gain

- Hybrid balance

- Frequency response in transmit and receive direction

- Hook thresholds

- Ringing generation and detection parameters

This means, for example, that changing impedance matching or hybrid balance requires no hardware modifications, so that a single device is able to meet requirements for different markets. The digital nature of the filters and gain stages also ensures high reliability, no drifts (over temperature or time) and simple variations between different line types.

The coeff.dat configuration file is produced by 3Com for each market after comprehensive performance analysis and testing, and can be modified on request. The current file supports US line type of 600 ohm AC impedance and 40 V RMS ringing voltage for REN = 2.

In future software releases, it will be expanded to consist of different sets of line parameters, which can be selected in the INI file, for each port.

To support different types of countries and markets, it is necessary to support loading of a new Coefficients.ini file. This file consist of AC and DC line parameters for the peripheral devices. This file is loaded into the VCX V7111 using the TFTP, in the same way as for the tones.dat file.

Reader's Notes

3Com VCX V7111 VoIP SIP Gateways User Manual

CHAPTER 7: DEVICE MANAGEMENT

i *All VCX V7111 Gateways have similar functionality except for the number of channels (the VCX V7111 24 FXS and 2FXS support only FXS), and all versions are referred to collectively in this document as the VCX V7111.*

VCX V7111 FXS refers only to the 24FXS, 8FXS, 4FXS, and 2FXS Gateways.

VCX V7111 FXO refers only to 8FXO and 4FXO Gateways.

SNMP Management

SNMP Overview

SNMP (Simple Network Management Protocol) is a standard network-based client/server-based control protocol to manage devices in the network. The client program (called the Network Manager) makes connections to a server program, called the SNMP Agent. The SNMP Agent, embedded on a remote network device, serves information to the Network Manager regarding the device's status. The database used by the Agent to retrieve information, is referred to as the SNMP Management Information Base (MIB), and is a standard set of statistical and control values. Apart from the standard MIBs documented in IETF's RFCs, SNMP additionally allows the usage of private MIBs, containing non-standard information set.

Directives, issued by the network manager client to an SNMP Agent, consist of the identifiers of SNMP variables (referred to as MIB object identifiers or MIB variables) along with instructions to either get the value for the identifier, or set the identifier to a new value.

The definitions of MIB variables supported by a particular agent are incorporated in descriptor files, written in Abstract Syntax Notation (ASN.1) format, made available to network management client programs so that they can become aware of MIB variables and their usage.

The VCX V7111 contains an embedded SNMP Agent supporting both general network MIBs (such as the IP MIB), VoIP-specific MIBs (such as RTP) and a proprietary MIB (known also as 3Com MIB) enabling a deeper probe into the inter-working of the Gateway. All the supported MIBs files are supplied as part of the release.

SNMP Message Standard

Four types of SNMP messages are defined:

- **Get** – Request that returns the value of a named object.

- **Get-Next** – Request that returns the next name (and value) of the next object supported by a network device given a valid SNMP name.

- **Set** – Request that sets a named object to a specific value.

- **Trap** – Message generated asynchronously by network devices. It notifies the network manager of a problem apart from polling of the device.

Each of the following message types fulfills a particular requirement of network managers:

- **Get Request** – Specific values can be fetched using the get request to determine the performance and state of the device. Typically, many different values and parameters can be determined using SNMP without the overhead associated with logging into the device, or establishing a TCP connection with the device.

- **Get Next Request** – Enables the SNMP standard network managers to walk through all SNMP values of a device (using the get-next request) to determine all names and values that the device supports. This is accomplished by beginning with the first SNMP object to be fetched, fetching the next name with a get-next, and repeating this operation until an error is encountered (indicating that all MIB object names have been walked).

- **Set Request** –The SNMP standard provides a method of effecting an action associated with a device (using the set request) to accomplish activities such as disabling interfaces, disconnecting users, clearing registers, etc. This provides a way of configuring and controlling network devices using SNMP.

- **Trap Message** – The SNMP standard furnishes a mechanism by which devices can reach out to a network manager on their own (using the trap message) to notify the manager of a problem with the device. This typically requires each device on the network to be configured to issue SNMP traps to one or more network devices that are awaiting these traps. The Trap messages are send to SNMP Manager. The IP address of SNMP Manager is defined in the INI file or using a web browser (in Network Settings).

SNMP MIB Objects

The SNMP MIB is arranged in a tree-structured fashion, similar in many ways to a disk directory structure of files. The top level SNMP branch begins with the ISO Internet directory, which contains four main branches:

▪ The mgmt SNMP branch contains the standard SNMP objects usually supported (at least in part) by all network devices.

▪ The private SNMP branch contains those extended SNMP objects defined by network equipment vendors.

▪ The experimental and directory SNMP branches, also defined within the internet root directory, are usually devoid of any meaningful data or objects.

This tree structure is an integral part of the SNMP standard; however, the most pertinent parts of the tree are the leaf objects of the tree that provide actual management data regarding the device. Generally, SNMP leaf objects can be partitioned into two similar but slightly different types that reflect the organization of the tree structure:

▪ **Discrete MIB Objects** – Contain one precise piece of management data. These objects are often distinguished from Table items by adding a .0 (dot-zero) extension to their names. The operator must know the name of the object.

▪ **Table MIB Objects** – Contain multiple pieces of management data. These objects are distinguished from Discrete items by requiring a . (dot) extension to their names that uniquely distinguishes the particular value being referenced. The . (dot) extension is the instance number of an SNMP object. In the case of Discrete objects, this instance number is zero. In the case of Table objects, this instance number is the index into the SNMP table. SNMP tables are special types of SNMP objects, which allow parallel arrays of information to be supported. Tables are distinguished from scalar objects, in that tables can grow without bounds. For example, SNMP defines the ifDescr object (as a standard SNMP object) that indicates the text description of each interface supported by a particular device. Since network devices can be configured with more than one interface, this object could only be represented as an array.

By convention, SNMP objects are always grouped in an Entry directory, within an object with a Table suffix. (The ifDescr object described resides in the ifEntry directory contained in the ifTable directory).

SNMP Extensibility Feature

One of the principal components of any respectable SNMP manager is a MIB Compiler that allows new MIB objects to be added to the management system. When a MIB is compiled into an SNMP manager, the manager is made aware of new objects that are supported by agents on the network. The concept is similar to adding a new schema to a database.

Typically, when a MIB is compiled into the system, the manager creates new folders or directories that correspond to the objects. These folders or directories can typically be viewed with a MIB Browser, which is a traditional SNMP management tool incorporated into virtually all network management systems.

The act of compiling the MIB allows the manager to know about the special objects supported by the agent and access these objects as part of the standard object set.

VCX V7111 Gateway Supported MIBs

The VCX V7111 Gateway contains an embedded SNMP Agent supporting the following MIBs:

▪ **The Standard MIB (MIB-II)** – The various SNMP values in the standard MIB are defined in RFC 1213. The standard MIB includes various objects to measure and monitor IP activity, TCP activity, UDP activity, IP routes, TCP connections, interfaces, and general system description.

▪ **RTP MIB** – The RTP MIB is supported per the RFC 2959. It contains objects relevant to the RTP streams generated and terminated by the Gateway and to the RTCP information related to these streams.

▪ **AcBoard MIB** – This proprietary MIB contains objects related both to the configuration of the Gateway and channels as well as to run-time information. Through this MIB, you can set up the Gateway configuration parameters, reset the Gateway, and monitor the Gateway's operational robustness and quality of service during run-time.

Web Management

Overview

The VCX V7111 Gateway contains an Embedded Web Server to be used both for Gateway configuration, including downloading of configuration files, and for run-time monitoring. The Web server can be accessed from any standard web browser, such as Microsoft Internet Explorer or Netscape Navigator. Specifically, you can employ this facility to set up the Gateway configuration parameters needed to configure the Gateway. You also have the option to reset the Gateway to apply the new set of parameters.

Access to the Embedded Web Server is controlled by protection and security mechanisms described in the following sections.

> *The 8FXS, 8FXO, 4FXS, 4FXO, and 2FXS Gateways have identical functionality (the 2FXS supports FXS only), except for the number of channels, and are referred to collectively in this manual as the VCX V7111.*

Password Control

The Embedded Web Server is protected by a unique username-password combination. The first time a browser request is made, you are requested to provide its username-password so that you can obtain access. Subsequent requests are negotiated by the browser on behalf of the User, so that you do not have to re-enter the username-password for each request, but the request is still authenticated.

An additional level of protection is obtained by a restriction that no more than three IP addresses can access the Embedded Web Server concurrently. With this approach, a fourth user is told that the server is busy, even if the correct username-password was provided.

The Embedded Web Server Username-Password

The default username-password for all Gateways is:

- Username = Admin

- Password = Admin

Change the Web password using the *Advanced Configuration Menu > Change Password* selection and then follow the pop-up window directives. The password can be a maximum of seven characters. The new password is active only after restarting the Gateway using the *Reset* button of the Embedded Web Server. Otherwise, the old password is still active.

You can reset the Web username-password (to the default values) using an INI file parameter called ResetWebPassword. The Web password is automatically the default password.

Web Configuration

The Embedded Web Server can be configured using INI file parameters.

Read-only Mode

The Embedded Web Server can be initialized to read-only mode by setting the DisableWebConfig = 1 INI file parameter (the default state is read-write mode). In this mode, all web pages are presented in read-only mode. By selecting this mode, you disable the capability to modify the configuration data. In addition, you do *not* have access to the Change Password page or to the Reset page. When the Gateway is controlled through PCI, the Embedded Web Server is always in read-only mode.

Disable/Enable Embedded Web Server

To deny access to the Gateway through HTTP protocol, you have the capability to disable the Embedded Web Server task. To disable the Web task, use an INI file parameter called DisableWebTask. The default is to Web task enabled. When the Gateway is controlled through PCI, the Embedded Web Server is always activated.

Using the Embedded Web Server

This section explains how to use the Embedded Web Server. After the initial IP address is set to the Gateway it is possible to connect with the integral web-based configuration application.

To access this web application, follow these steps:

1   Invoke any standard web-browsing application, such as Microsoft Internet Explorer (Version. 5.0 and higher) or Netscape Navigator (Version. 7.0 and higher).

2   Specify the IP address of the Gateway in the Address field; the Embedded Web Server screen appears, as shown in Figure 36.

3   Enter the user name and password (default: Admin, Admin).

4   Click *OK*; the Quick Setup screen is accessed.

To configure the Gateway parameters you can use either the *Quick Setup* menu or go directly to *Protocol Management* or *Advanced Configuration* menus. Quick Setup provides a basic set of Gateway configuration settings. An example of the Quick Setup configuration is described in "Quick Setup Procedure" on page 38.

Figure 36   Embedded Web Server – Home Page

                                             3Com VCX V7111 VoIP SIP Gateways User Manual

Set Up Gateway Configuration Parameters

Clicking the *Advanced Configuration* button leads to the following screen.

Figure 37   Embedded Web-Server - Gateway Parameters

Selecting each of the sub menus shows the active configuration of each section and the values of the relevant parameters. From this menu you can select the following web pages:

- Network Settings

- Channel Settings

- Configuration File

- Regional Settings

- Change Password

Set up Gateway Network Parameters

To change the Gateway network settings, select the *Network Settings* tab shown in the following figure.

Figure 38   Web Server – Network Settings

From network settings page you can define:

- IP settings including the Gateway IP address and subnet mask.

- Logging settings, such as IP address of SysLog Server. If the SysLog Server is disabled, the logging data is sent to the Gateway's serial RS-232 port.

- SNMP settings.

- RTP settings, including RTP Base UDP port, RTP IP Differentiated Services or RTP IP TOS, and RTP IP Precedence QOS parameters.

- Ethernet status (read only).

Channel Settings

From Channel Settings page, you can modify the following channel associated parameters:

- Voice Setting

- Fax/Modem/CID Settings

- RTP Settings

- IP Media Settings

Configuration Files

Selecting the *Configuration Files* tab enables you to send new INI file to the Gateway or to upload from Gateway to PC the current INI file the Gateway is using. The uploaded INI file will include only parameters that were modified for the default values.

Figure 39   Configuration Files

Regional Settings

From the Regional Settings page, you can download the following files to the Gateway:

- Call Progress Tone

- Coefficient file (different file for VCX V7111 FXS and FXO Gateways)

- Voice Prompt file – Currently not applicable for VCX V7111 Gateways

In addition, you can set and view from this page the Gateway internal date and time.

Figure 40   Regional Settings

Change Password

To change the username and password, select the *Change Password* tab as shown in the following figure.

Figure 41   Change Password

Set Up Gateway SIP Parameters

To set the Gateway's SIP parameters, select the *Protocol Management* tab shown in the following figure.

Figure 42   Protocol Management Parameters

From this menu, you can select one of the following configuration web pages:

- Protocol Definition page, enabling to configure SIP Gateway various parameters

- Manipulation tables

- EndPoints' Phone Number table

- Automatic Dialing table

- Caller ID table

- Tel To IP Routing table

- IP to Hunt Group Routing table

- Call Forward table (For VCX V7111 FXS Gateways); currently not supported on SIP Gateways

Set Up SIP Protocol Definition Parameters

Protocol Definition page enables you to define SIP configuration parameters. The SIP protocol definition page includes several sections:

- General, Proxy Server, and Authentication Parameters shown in Figure 43.

Figure 43 General, Proxy Server and Authentication Parameters

General	
Gateway Name :	10.8.8.10
Enable PRACK:	Yes
Session-Expires Time:	0
Use Info for DTMF:	No
Use Info for Hook-Flash:	No
Proxy Server and Authentication	
Enable Proxy:	No
Proxy Name :	
Proxy IP :	10.8.8.10
Redundant Proxy IP Address:	
Registrar IP:	10.8.8.10
Enable Registration:	No
Registration Time:	180
Enable Proxy Keep Alive:	Yes
Password :	**************
Cnonce :	Default_Cnonce
User Name :	

- VoIP Coders shown in the following figure.

Figure 44 Coders

* Coders	
* 1st Coder :	g711Alaw64k(20ms
* 2nd Coder:	
* 3rd Coder:	
* 4th Coder:	
* 5th Coder:	

- DTMF and Dialing Parameters shown in the following figure.

Figure 45 DTMF and Dialing Parameters

DTMF and Dialing Parameters	
Enable Automatic Dialing :	No
*Max Digits In Phone Num:	4
* Interdigits Timeout[sec]:	4
* Dial Tone Duration [sec]:	16

- Early Media Parameters shown in the following figure.

Figure 46   Early Media Parameters

Early Media Parameters	
* Enable Early Media:	No
* Play Ringback Tone to IP:	Dont Play
* Play Ringback Tone to Tel:	Play

- Number Manipulation and Routing Modes Parameters, shown in the following figure. Add Hunt Group Id as Prefix feature adds the Hunt Group ID to destination number for Tel → IP calls. The routing and number manipulation rules can then be applied to the new number, enabling you to define routing rules based on the hunt group from where the call arrived. The two routing mode parameters define the execution order of number manipulation as opposed to the routing rules.

Figure 47   Number of Manipulation and Routing Modes

Number Manipulation and Routing Modes	
* Add Trunk Group Id as Prefix:	No
* IP To Tel Routing Mode:	Route calls after mar
* Tel To IP Routing Mode:	Route calls after mar

- Supplementary Services Parameters. These parameters enable Call Hold/Unhold and Call Transfer services shown in the following figure.

Figure 48   Supplementary Services

Supplementary Services	
Enable Hold:	No
Enable Transfer:	No

- Other Miscellaneous parameters shown in the following figure.

Figure 49   Miscellaneous Parameters

Misc parameters	
* Select Next Available Channel:	Yes
Enable Caller ID:	No
Caller ID Type:	Bellcore
Enable Polarity Reversal:	No
Enable Current Disconnect:	No
SIP Destination Port:	5060
* IP Security:	Yes
Enable Remote Party ID:	No
Use "user=phone" in sip URL:	Yes
SIP T1 retransmission timer[msec] :	4000
SIP T2 retransmittion timer [msec] :	6000
Enable Syslog CDR:	Yes

Configuration of Number Manipulation Tables

Manipulation Tables tab defines four manipulation tables used to modify destination and source numbers in IP→ Tel and Tel→ IP directions. Manipulation table for IP → Tel calls is shown in the following figure.

Figure 50   Phone Number Manipulation Table for IP → Tel calls

	Prefix	Num of stripped digits	Prefix to add	Number of digits to leave
1	03	0	972	
2	1001	4	5	
3	123451001	0	8	4
4				
5				
6				
7				
8				
9				
10				

In Figure 50, the first rule adds 972 prefix to all numbers starting with 03. The second rule first deletes four digits from all numbers starting with 1001 and then the digit 5 is added as a prefix. The third rule deletes all digits except for the last four, from all numbers starting with 123451001 and adds the digit 8 as a prefix.

For example, number 035000 is changed to 972035000, number 1001876 is changed to 5876 and number 1234510012001 is changed to 82001.

Set Up Gateway Endpoints' Phone Numbers

Endpoints' Phone Numbers table allocates phone numbers to Gateway ports, and to enable/disable Gateway ports. The table defines phone numbers for Gateway endpoints. The endpoints that are not defined are disabled.

In Channel(s) field a range of endpoints can be entered, such as from 1 through 8 for the 8FXS or 8FXO or from 1 through 24 for the 24FXS. For a single endpoint, a single number can be entered in the channel(s) field.

Hunt Group ID can optionally be configured to define a group of channels that will be used for routing IP → Tel calls with common rules. Figure 51 shows the Endpoints' Phone Number table.

Figure 51   Endpoint's Phone Number Table

	Channel(s)	Phone Number	Hunt Group Id
1	1-4	101	
2	5	201	
3	6	202	
4	7	203	
5	8	204	
6			
7			
8			

Set Up Automatic Dialing Table

Automatic Dialing table defines destination numbers for Tel ➔ IP calls. It can be used if Automatic Dialing feature is enabled (IsDialNeeded = 1). The table is applicable for FXS and FXO analog Gateways, for outgoing, Tel ➔ IP calls. The table contains pre-configured phone numbers per Gateway port. The number is automatically dialed if phone is picked up (for FXS Gateway) or ring signal is applied to FXO Gateway port, see Figure 52.

Figure 52   Automatic Dialing Table

GW Port	Destination Phone Number
Port 1	1002
Port 2	1003
Port 3	1004
Port 4	1101
Port 5	1102
Port 6	1105
Port 7	1234
Port 8	2222

Set Up of Caller ID Display Table

Caller ID Display Info table, shown in the following figure, contains Caller ID display information per VCX V7111 FXS/FXO Gateway port. This information is sent to remote party, for Tel ➔ IP calls. Remote party can use this display information for caller identification. The caller ID string can contain up to 18 characters.

Figure 53   Caller ID Table

Gateway Port Number	Caller ID/Name
Port 1	Zohar
Port 2	Avi
Port 3	Michel
Port 4	Eran
Port 5	Surpin
Port 6	Doron
Port 7	Tania
Port 8	Ilan

Set Up of Tel to IP Routing Table

Tel to IP Routing & IP Security table is required if the Gateway operates without a proxy server. It contains up to 50 rows. Each row associates a called phone number prefix with destination IP address. The routing table can also be used for fallback routing if communication with the proxy server is lost. The IP addresses listed in the table can also be used for security. If security feature is enabled (SecureCallFromIP = 1) the Gateway accepts calls coming only from these addresses. The Phone to IP Routing Table is shown in the following figure.

Figure 54   Phone to IP Routing & IP Security Table

	Destination Phone Prefix	IP Address
1	25	10.2.201.11
2	16	10.2.201.22
3	4	10.2.201.22
4	12345	10.2.0.3
5	2040	10.2.0.4
6		
7		
8		
9		
10		

Set Up of IP to Hunt Group Routing Table

IP to Hunt Group Routing table can be used to route incoming IP calls to a group of channels. In the example that follows, IP incoming calls with called numbers starting with 10 or 20 will be routed to the first Hunt Group, while called numbers starting with 302 will be sent to the second Hunt group. In each Hunt group the next available Gateway port will be allocated. To use Hunt Groups it is necessary to enable Select Next Available Channel in Protocol Definition web page (or configure IsUseFreeChannel = 1 in the INI file). The IP to Hunt Group Routing Table is shown in the following figure.

Figure 55   IP to Hunt Group Routing Table

Prefix	Hunt Group Id
10	1
20	1
302	2
401	3

Channel Status Menu

Selecting the *Status & Diagnostics* tab and then the Channel Status page provides real time monitoring of the current channel status, as shown in the example in Figure 56.

Figure 56   Channel Status

Active Channels are green. To view the status of a specific channel, click it; a channel status screen appears (Figure 57).

Figure 57   Channel Status Detail

Reader's Notes

CHAPTER 8: DIAGNOSTICS

> **i** All VCX V7111 Gateways have similar functionality except for the number of channels (the VCX V7111 24 FXS and 2FXS support only FXS), and all versions are referred to collectively in this document as the VCX V7111.
>
> VCX V7111 FXS refers only to the 24FXS, 8FXS, 4FXS, and 2FXS Gateways.
>
> VCX V7111 FXO refers only to 8FXO and 4FXO Gateways.

Diagnostics Overview

3Com provides several diagnostic tools to enable you to identify an error condition and to provide a solution or work around when working with VCX V7111 Gateways.

- LED Indication of channel activity status, data, control, and LAN status.

- VCX V7111 Self-Testing on hardware initialization.

- RS-232 Terminal Notification Messages.

- SysLog Event Notification Messages.

- Solutions to Common Problems.

VCX V7111 Gateway Alarms and SNMP Traps

LED Visual Indicator Status and Alarms

Table 21   Indicator LEDs on the VCX V7111 Front Panel

Label	Type	Color	State	Meaning
LAN	Ethernet Link Status	Green	ON	Valid Connection to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Sending and receiving SIP messages
		Red		Not supported in current release
Ready	Device Status	Green	ON	Device Powered, Self test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction

Table 22   VCX V7111 with Channel LEDs

VCX V7111 with 1 – 8 Channels

Label	Type	Color	State	Meaning
Activity	FXS Tel Port	Green	ON	Off-Hook/Ringing for Phone Port
	FXO Line Port	Green	ON	Line-Seize/Ringing State for Line Port

VCX V7111 Self-Testing

The VCX V7111 features two self-testing modes: rapid and detailed.

Rapid self-test mode is invoked each time the Gateway completes the initialization process. This is a short test phase in which the only error detected and reported is failure in initializing hardware components. All Status and Error reports in this self-test phase are reported through Network Interface ports, as well as indicated by the LED status indicators.

Detailed self-test mode is invoked when initialization of the Gateway is completed and if the configuration parameter EnableDiagnostics is set to 1 (this parameter can be configured through the INI file). In this mode, the Gateway tests all the hardware components (such as memory and DSP), outputs the status of the test results, and ends the test. To continue operational running, reset the Gateway again but this time configure the EnableDiagnostics parameter to 0.

RS-232 Terminal

The VCX V7111 status and error messages can be viewed using a terminal connected to the RS-232 management port.

To connect VCX V7111 to a HyperTerminal using a standard RS-232 straight cable (not a cross-over cable) with DB-9 connectors, connect the VCX V7111 RS-232 port (it is marked RS232) to either COM1 or COM2 RS-232 communication port on the PC. The connector pinout and gender are shown in <u>Figure 58</u>.

Figure 58   RS-232 Cable Wiring

To configure the HyperTerminal, follow these steps:

1   On a PC running a Windows operating system, open *Start>Programs>Accessories>Communications>HyperTerminal*. The Connection Description dialog opens.

2   Enter a name for the new connection in the Name field and click *OK*; the Connect To dialog opens.

3   In the Connect To dialog, enter COM1 or COM2, depending on the physical connection you performed when connecting the VCX V7111 to the PC with the RS-232 cable; the COM1/2 Properties dialog opens.

4   In the COM1/2 Properties dialog, enter the following settings for the serial communication port:

- Baud Rate: 115,200 bps

- Data bits: 8

- Parity: None

- Stop bits: 1

- Flow control: Hardware

5 Click *OK*; the HyperTerminal main screen opens.

After applying power or resetting the Gateway, the following information is printed on the terminal screen shown in [Figure 59](#).

Figure 59 Status and Error Messages

```
MAC address = 00-90-8F-01-00-9E
Local IP address = 10.1.37.6
Subnet mask = 255.255.0.0
Default gateway IP address = 10.1.1.5
TFTP server IP address = 10.1.1.167
Boot file name = ram35136.cmp
INI file name = mp108.ini
Call agent IP address = 10.1.1.18
Log server IP address = 0.0.0.0
Full/Half Duplex state = HALF DUPLEX
Flash Software Burning state = OFF
Serial Debug Mode = OFF
Lan Debug Mode = OFF

BootLoad Version 1.75
Starting TFTP download ... Done.
MP108 Version 3.80.00
```

SysLog Support

Overview

SysLog protocol is an event notification protocol that allows a machine to send event notification messages across IP networks to event message collectors - also known as SysLog servers. The SysLog protocol is defined in RFC 3164 IETF standard.

Since each process, application, and operating system was written somewhat independently, there is little uniformity to SysLog messages. For this reason, no assumption is made on the contents of the messages other than the minimum requirements of its priority.

SysLog uses UDP as its underlying transport layer mechanism. The UDP port that has been assigned to SysLog is 514.

The SysLog message is transmitted as an ASCII message. The message starts with a leading < (less-than character), followed by a number, which is followed by a > (greater-than character). This is optionally followed by a single ASCII space.

The number with these brackets is known as the Priority and represents both the Facility and Severity. The Priority number consists of one, two, or three decimal integers.

Example

```
<37> Oct 11 16:00:15 mymachine su: 'su root' failed for lonvick on
/dev/pts/8
```

SysLog Operation

Sending the SysLog Messages

The 3Com SysLog client, embedded in the firmware of the VCX V7111, sends error reports/events generated by the VCX V7111 unit application to a SysLog server, using IP/UDP protocol. 3Com does *not* provide a SysLog server as several are provided as shareware that can be downloaded from the Internet.

Examples of SysLog Servers downloadable from the Internet:

- Kiwi Enterprises: http://www.kiwi-enterprises.com/software_downloads.htm

- The US CMS Server: http://uscms.fnal.gov/hanlon/uscms_server/

- TriAction Software: http://www.triaction.nl/Products/SyslogDaemon.asp

- Netal SL4NT 2.1 Syslog Daemon: http://www.netal.com/

A typical SysLog server application enables filtering of the messages according to priority, IP sender address, time, and date.

Setting the SysLog Server IP Address

A SysLogServerIP Address parameter is supplied with the web browser or from an INI file in order to determine the address of the SysLog server.

Controlling the Activation of the SysLog Client

Activation of the SysLog client is controlled by an EnableSyslog INI file parameter. Setting it to 1 enables the SysLog protocol log.

The INI *File Example for SysLog*

Figure 60   The INI File Example for SysLog
```
[Syslog]
SyslogServerIP=10.2.0.136
EnableSyslog =1
```

Solutions to Possible Problems

General

If there is a problem, check the following resources:

- Web browser status and channel parameter pages.

- Log messages of VCX V7111 in HyperTerminal screen.

- BootP and TFTP log messages (for startup problems).

- Log messages in SysLog server.

Possible Common Problems

Possible common problems are described in the following table.

Table 23   Possible Common Problems

Problem	Possible Cause	What to do
No communication	Software does not function in VCX V7111	Try to ping to VCX V7111. If ping fails, check for network problems/definitions and try to reset the VCX V7111.
	Network problem	Check cables.
	Network definitions	▪ Check if default Gateway can reach IP of box. ▪ Check if box got the correct IP (it can be seen in the HyperTerminal screen). ▪ Check the validity of IP address, subnet and default Gateway. ▪ If default Gateway is not used, enter 0.0.0.0
	BootP did not reply to box	▪ Check if BootP server replied to VCX V7111 at restart; it is seen in the BootP server log. ▪ Try to restart BootP server. ▪ Check the MAC address of the box in BootP server.
INI file was not loaded	TFTP server down	Check if TFTP server working.
	TFTP server did not get the request	Check this in its log.
	VCX V7111 did not request the file from your TFTP	Look in HyperTerminal for the TFTP server IP address that the VCX V7111 is trying to use.
	TFTP server bug	Try to restart TFTP server.
	BootP sent to MP wrong TFTP server address	Check in HyperTerminal screen the address of used TFTP.
	The INI file does not exists in default directory of TFTP	Check default directory of TFTP server and check that INI file exists there.
	Wrong INI file name	▪ Verify in windows explorer that file extensions are displayed and the INI file is not by mistake XXX.ini.ini. ▪ Verify that extension INI is in lowercase letters.
	TFTP have too short timeout	Verify that: ▪ Timeout = 2 seconds ▪ # of retransmission = 10

Table 23   Possible Common Problems

Problem	Possible Cause	What to do
Wrong INI file loaded	The INI file is not in the correct position	Old INI file was probably loaded. Check which INI file was loaded. This can be done using HyperTerminal screen. The Gateway displays contents of INI file before it began.
	The INI file is corrupted	Check INI file syntax.

Reader's Notes

CHAPTER 9: SPECIFICATIONS

> ℹ️ *All VCX V7111 Gateways have similar functionality except for the number of channels (the VCX V7111 24 FXS and 2FXS support only FXS), and all versions are referred to collectively in this document as the VCX V7111.*
>
> *VCX V7111 FXS refers only to the 24FXS, 8FXS, 4FXS, and 2FXS Gateways.*
>
> *VCX V7111 FXO refers only to 8FXO and 4FXO Gateways.*

VCX V7111 Specifications

Table 24   VCX V7111 Functional Specifications

VCX V7111 FXS Functionality	
FXS Capabilities	▪ Short or Long Haul up to 3,000 m (10,000 ft) using 24 AWG line cord.
	▪ Includes lightning and high voltage protection for outdoor operation.
	▪ Caller ID generation: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation or ETSI Type 1 (between first and second rings).
	▪ Programmable Line Characteristics: Battery feed, line current, hook thresholds, AC impedance matching, hybrid balance, Tx and Rx frequency response, Tx and Rx Gains.
	▪ Programmable ringing signal. Up to three cadences and frequency 10 – 200 Hz.
	▪ Drive up to four phones per port (total 32 phones) simultaneously in Off-hook and Ring states. VCX V7111 24 FXS REN = 2 VCX V7111 2FXS, 4FXS, 4FXO, 8FXS, or 8FXO REN = 5
	▪ Over-temperature protection for abnormal situations as shorted lines.
	▪ Loop-backs for testing and maintenance.

VCX V7111 FXO Functionality

FXO Capabilities (does not apply to the 2FXS)	▪ Short or Long Haul up to 7,000 m (24,000 ft) using 24 AWG line cord. ▪ Includes lightning and high voltage protection for outdoor operation. ▪ Programmable Line Characteristics: AC impedance matching, hybrid balance, Tx and Rx frequency response, Tx and Rx Gains, ring detection threshold, DC characteristics. ▪ Caller ID detection: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation or ETSI Type 1 (between first and second rings).

Voice and Tone Characteristics

Voice Compression	G.711 PCM at 64 Kbps µ-law/A-law G.723.1 MP-MLQ at 5.3 or 6.3 Kbps G.726 at 16 – 40 Kbps ADPCM and E-ADPCM G.729A CS-ACELP at 8 Kbps NetCoder at 6.4 – 8.8 Kbps, 800-bit increments (proprietary coder)
Silence Suppression	G.723.1 Annex A G.729 Annex B PCM and ADPCM - Proprietary Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) NetCoder
Echo Canceler	G.168, 25 ms
Gain Control	Programmable
DTMF Transport	Mute, transfer in RTP payload or relay in compliance with RFC 2833
DTMF Detection and Generation	Dynamic range 0 to –25 dBm, compliant with TIA 464B and Bellcore TR-NWT-000506.
Call Progress Tone Detection and Generation	15 tones: single tone or dual tones, programmable frequency and amplitude; 16 frequencies in the range 300 –2000 Hz, 1 or 2 cadences per tone, up to two sets of ON/OFF periods.
Output Gain Control	-31 dB to +31 dB in steps of 1 dB
Input Gain Control	-31 dB to +31 dB in steps of 1 dB

Fax/Modem Relay

Fax Relay	Group 3 fax relay up to 14.4 Kbps with auto fallback T.38 compliant, real time fax relay Tolerant network delay (up to 9 seconds round trip)
Modem Relay	Up to 14.4 Kbps V.32bis (optional)

Modem Transparency	Auto switch to PCM or ADPCM on V.34 or V.90 modem detection

Protocols

VoIP Signaling Protocol	SIP RFC3261
Communication Protocols	RTP/RTCP packetization
	IP stack (UDP, TCP, RTP)
	Remote Software download (TFTP and BootP support)
Line Signaling Protocols	Loop start, FXS and FXO

Interfaces

FXS Telephony Interface	2, 4, 8 or 24 Analog FXS phone or fax ports, loop start
FXO Telephony Interface	4 or 8 Analog FXO PSTN/PBX loop start ports
Network Interface	RJ-45 shielded connector, 10/100 Base-T
RS-232 Interface	RS-232 Terminal Interface for maintenance, diagnostic reports, and code tracing. DB-9 connector on rear panel
Life Line (8FXS, 4FXS, and 2FXS)	Life Line, connected to the unused pins on port 4 (port 2 for the 2FXS), with a relay to an analog line, even if the VCX V7111 FXS is powered off (See "Installation of the VCX V711 FXS Life Line" on page 21 for details). Does *not* function with the 24FXS and VCX V7111 FXO Gateways.

Connectors and Switches

Rear Panel

24 Analog Lines (24FXS)	50-pin Telco shielded connector
Eight Analog Lines (8FXS or 8FXO)	Eight RJ-11 connectors
Four Analog Lines (4FXS or 4FXO)	Four RJ-11 connectors
Two Analog Lines (2FXS)	Two RJ-11 connectors
Ethernet	10/100 Base-T, RJ-45 shielded connector
RS-232	Console port - DB-9

Front Panel

Reset	Resets the VCX V7111

Physical			
2 – 8-line Gateway Enclosure Dimensions	Width:	221 mm	8.7 in
	Height:	44.5 mm	1.75 in
	Depth:	240 mm	9.5 in
	Weight:	1.24 kg	2.5 lb
24FXS Enclosure Dimensions	1U, 19-inch Rack		
	Width:	445 mm	17.5 in
	Height:	44.5 mm	1.75 in
	Depth:	269 mm	10.6 in
	Weight:	2.24 kg	4.9 lb
Environmental	Operational:	0° to 45° C	32° to 113° F
	Storage:	-10° to 70° C	14° to 158° F
	Humidity:	10 to 90% non-condensing	
Installation	Desktop, shelf, or 19-inch rack mount with side brackets.		
Electrical	Universal 90–260 VAC, 1A, 47–63 Hz		
Type Approvals			
Telecommunication	FCC part 68 and CE CTR21		
Safety and EMC	UL 1950, FCC part 15 Class B		
	CE Mark (EN 60950, EN 55022, EN 55024)		
Management			
Configuration	Gateway configuration using Web browser, INI files or local RS-232 console		
Management and Maintenance	SNMP		
	SysLog, per RFC 3164		
	Local RS-232 terminal		
	Web Management		

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APPENDIX A: BOOTP/TFTP CONFIGURATION UTILITY

Introduction

The 3Com BootP/TFTP Configuration Utility enables easy configuration and provisioning of 3Com products. It contains BootP and TFTP servers with specific adaptations to 3Com requirements.

Key Features

- Internal BootP server supporting hundreds of entities.

- Internal TFTP server.

- Contains all required data for 3Com products in predefined format.

- Provides a TFTP server address, enabling network separation of TFTP and BootP servers.

- Tools to backup and restore the local database.

- Templates.

- User-defined names for each entity.

- Option for changing MAC address.

- Protection against entering faulty information.

- Remote reset (for this version and above).

- Unicast BootP respond.

- User-initiated BootP respond, for remote provisioning over WAN.

- Filtered display of BootP requests.

- Location of other BootP servers that contain the same MAC entity.

- Common log window for both BootP and TFTP sessions.

- Works with Windows 98, Windows NT, Windows 2000.

Specifications

- **BootP standards** – RFC 951 and RFC 1542

- **TFTP standards** – RFC 1350 and RFC 906

- **Operating System** – Windows 98, Windows NT, and Windows 2000

- **Max number of MAC entries** – 200

BootP Fields:

- Hardware address (MAC): 12 hex digits

- IP address

- Subnet

- Default Gateway

- TFTP server IP; (Using the TFTP server IP field enables the download of a software image from a different Host)

- Boot File

- INI File

- Call Agent IP

- New MAC (optional)

Screens:

- Logging screen.

- Preferences screen

- Client Configuration screen

- Template definition screen

BootP/TFTP Configuration Utility Installation

The BootP/TFTP Configuration Utility can be installed on a PC by downloading it from http://www.3com.com/:

- TrunkPack Boards (TP)

- VCX V7165 Platforms (IPM)

- TrunkPack Modules (TPM)

- MediaPack Series (MP)

To install the BootP/TFTP Configuration Utility, unzip the TP3.810.exe file and navigate to the BootP *.exe file. The installation procedure is facilitated by prompts. After completing the procedure, open *Start>Programs>BootP*; the BootP/TFTP Server main screen is displayed.

Logging Screen

The 3Com BootP/TFTP Configuration Utility main screen (Figure 61 on page 141) includes the Log line, printed per BootP request with the following parameters:

- Hardware (MAC) address

- Status (found or not found in cache)

- Date and Time

- Assigned IP address (if found)

- Client name

Clicking Log line shows all BootP reply parameters or enables entry to a new entity.

Preferences Window

The Preferences Window (Figure 62 on page 143) is used to define BootP and TFTP configuration parameters:

- TFTP directory

- INI File Mask

- Boot File Mask

- TFTP timeout and number of retransmissions

- BootP replay type (broadcast or unicast)

- BootP ARP mode (dynamic or static)

- Number of initiated BootP replies (send after remote reset), optionally used when the Gateway (2FXS, 4FXS, 4FXO, 8FXS, 8FXO, or 24FXS), is installed behind the firewall that blocks BootP broadcast requests.

Client Configuration Window

The Client Configuration Window (Figure 63 on page 144) shows:

- All client entities

- MAC

- Name

- IP per entity

Using this screen, you can:

- Add a new entry

- Delete an existing entry

- Modify an existing entry

- Test a selected client for finding all BootP servers that respond to a BootP request with a specific MAC address.

If a template is selected, any parameter can be entered manually or copied from the selected template, by marking the checkbox on the right side of the parameter. Usually, only an IP address is entered manually, while other parameters are copied from the template.

Template Window

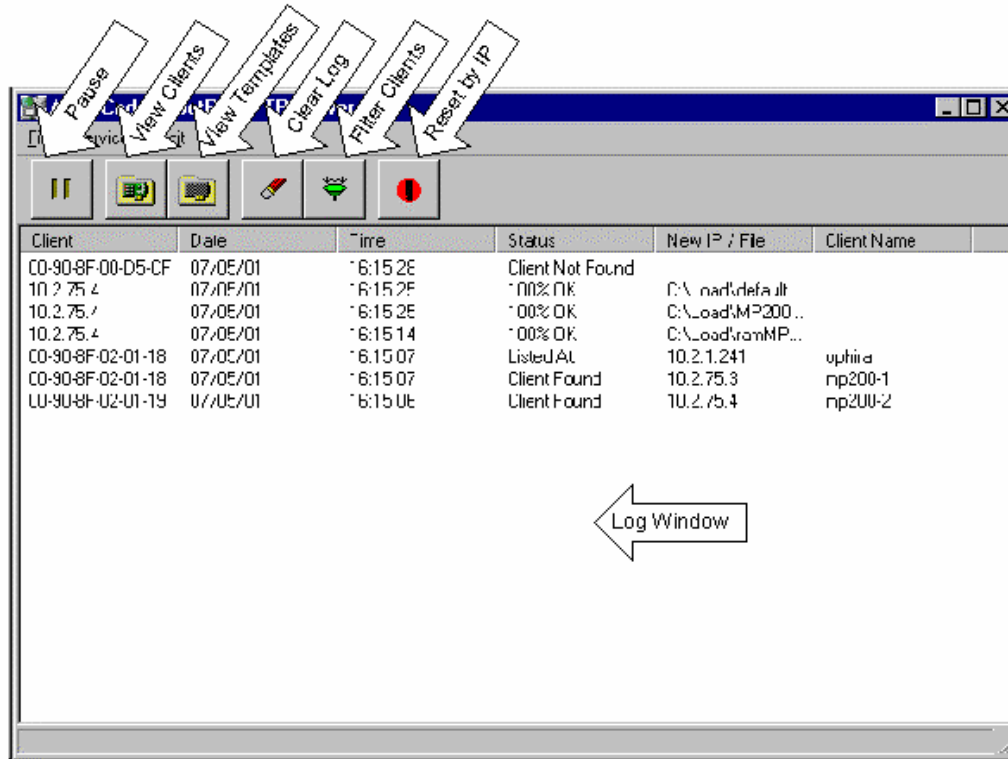The Template Window () enables you to add, modify, or delete templates.

The template includes:

- Subnet

- Gateway, TFTP server

- BootFile

- INI file

- Call Agent fields

- Server IP

Window Details

Main Window

Figure 61   Main Window

[Figure 61](#) is the main window of the program. It features the following controls:

- **Program State Button** – Pauses the program. When the program has paused, no replies to BootP requests are sent.

- **View Clients Button** – Opens the Clients Configuration window.

- **View Templates Button** – Opens the Templates Configuration window.

- **Clear Log Button** – Clears the log.

- **Filter Unknown Clients Button** – Filters all BootP requests that are not in the client configuration list.

- **Reset Button** – Opens a dialog box, where you can enter an IP of a client. The program sends a Reset command to that client.

- **Edit/Preferences** – Selecting *Preferences* from the *Edit* menu opens the Preferences window for defining BootP and TFTP parameters.

- **Log Window** – All BootP requests and TFTP sessions are displayed, including the time and date of the request. In addition, the response type is also displayed:

    - Client Not Found

    - Client Found

    - Client's Mac Changed

    - Client Disabled

    - Listed at (when using the test selected clients button)

    - For TFTP session, File name and Download status are displayed

- **Pop-Up Menu** – When you right-click a line in the log window, the pop-up menu opens. In this menu there two options:

    - **Reset** – When this option is selected, the program searches the database for the selected MAC. When the client is found, the program adds the client's MAC to the ARP table, and then sends a reset command to the client. Note that by performing the remote reset this way, you do not have to know the current IP of the client; however, you must have administrator privileges, or else an error message appears.

    - **View Client** – This option is the same as double-clicking on a line. When selected, the Clients Window opens. If the Client's MAC is found in the database, it is focused. If not, a new client is added, with the MAC filled out. You must only fill in the remaining fields.

Preferences Window

In the Preferences window, Figure 62, BootP and TFTP configuration parameters are defined.

Figure 62   Preferences Window

For the TFTP server, you can configure a TFTP directory and a value for TFTP Timeout and Maximum Retransmissions. Set these values to 2 and 10 as shown.

You can disable the TFTP server by clearing the *Enable* check box.

In the BootP section, you can select ARP mode: *Dynamic* or *Static*, and reply type: *Broadcast* or *Unicast*. For a typical application, use Dynamic ARP mode and Unicast, as shown.

This option requires you to have administrator privileges; otherwise, an error message appears. If you do not have administrator privileges, clear the *ARP Manipulation Enabled* check box in the Preferences Window, Figure 62.

The Number of Timed Replies (the number of initiated timed BootP replies) can be used when a VCX V7111 is installed behind a Firewall that blocks BootP broadcast requests. In a typical application, this feature can be disabled by entering 0 in this field. When selected, several BootP replies are sent to the VCX V7111 immediately after the remote reset command.

Client Configuration Window

is the Client Configuration Window in which clients are added and defined.

Figure 63 Client Configuration Window

In the left pane of the window is the client list. By clicking on a client in this list, the following parameters for this client are displayed on the right side of the window:

- **Client MAC** – This is the MAC address of the client. When you edit the MAC, a new client is added, with the same parameters as the previous client.

  The client can be disabled by clearing the check box on the right side of the Client MAC, causing the BootP server not to reply to the BootP request. The client can be enabled by selecting the check box. Click *Apply* each time the client enable check box is selected or cleared.

- **Client Name** – Free text for client description.

- **Template** – The template to be used for this client. When a template is selected, its parameters override all of the previous parameters.

- **IP, Subnet, Gateway** – Normal IP parameters.

- **TFTP Server IP** – The IP of the TFTP Server.

- **Boot File, INI File** – The files to request from the TFTP server.

- **Call Agent** – The IP of the Call Agent that will be controlling the Gateway.

Note the seven check boxes to the right of the parameters. These enable you to assign only the selected fields from the template. The rest can be unique for each client. When the field is assigned a value from the selected template, the field is grayed out.

Click *Apply* to save your changes. Clicking *Apply & Reset* saves the changes to the database, performs a remote reset to the client by adding the client's MAC to the ARP table, and then sends out a reset command. This option works only if *ARP Manipulation Enabled* check box in the Preferences window is selected (in Figure 62 on page 142); otherwise, an error message appears. It requires you to have administrator privileges. The remote reset is supported for software in this version and up.

When adding a new client, follow these steps:

1 Click *Add Client*; a client with blank parameters is displayed.

2 Fill out the parameters.

3 Click *Apply*; the client is added.

To find out if there is another BootP server on the net that contains a client with the same MAC address, follow these steps:

1 Click Test Selected Clients; in the log screen.

2 View the IP addresses of all BootP servers that contain the same MAC address in the status Listed At. In normal operation, BootP client MAC address should be listed only on a single BootP server. If the MAC address is listed in multiple BootP servers, it must be removed from other BootP servers.

Templates Window

Figure 64 shows the Templates window, which provides a fast way to configure a number of clients that have the same parameters (except for the IP address). To use the Templates window, create a template, and then apply the template to the client by selecting it.

Figure 64   Templates Window

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APPENDIX B: RTP/RTCP PAYLOAD TYPES

RTP Payload Types are defined in RFC 1889 and RFC 1890. 3Com has added new payload types to enable advanced use of other coder types. These types are reportedly not used by other applications.

See the VCX V7165 Digital Media Server Release Notes for the supported coders.

Packet Types Defined in RFC 1890

Table 25   Packet Types Defined in RFC 1890

Payload Type	Description	Basic Packet Rate [ms]
0	G.711 µ-Law	20
2	G.726-32	20
4	G.723 (6.3/5.3 Kbps)	30
8	G.711 A-Law	20
18	G.729	20
200	RTCP Sender Report	Randomly, approximately every 5 seconds (when packets are sent by channel)
201	RTCP Receiver Report	Randomly, approximately every 5 seconds (when channel is only receiving)
202	RTCP SDES packet	
203	RTCP BYE packet	
204	RTCP APP packet	

3Com Defined Payload Types

Table 26   3Com Defined Payload Types

Payload Type	Description	Basic Packet Rate [ms]
35	G.726 16 Kbps	20
36	G.726 24 Kbps	20
38	G.726 40 Kbps	20
39	G.727 16 Kbps	20
40	G.727 24-16 Kbps	20
41	G.727 24 Kbps	20
42	G.727 32-16 Kbps	20
43	G.727 32-24 Kbps	20
44	G.727-32 Kbps	20
45	G.727 40-16 Kbps	20
46	G.727 40-24 Kbps	20
47	G.727 40-32 Kbps	20
51	NetCoder 6.4 Kbps	20
52	NetCoder 7.2 Kbps	20
53	NetCoder 8.0 Kbps	20
54	NetCoder 8.8 Kbps	20
56	Transparent PCM	20
100	DTMF relay	20
101	Fax Relay	Different packet rates
102	Fax Bypass	20
103	Modem Bypass	20
104	RFC 2198 (Redundancy)	Same as channel's voice coder

Default RTP/RTCP/T.38 Port Allocation

Table 27 Default RTP/RTCP/T.38 Port Allocation

Channel Number	RTP Port	RTCP Port	T.38 Port
1	4000	4001	4002
2	4010	4011	4012
3	4020	4021	4022
4	4030	4031	4032
5	4040	4041	4042
6	4050	4051	4052
7	4060	4061	4062
8	4070	4071	4072
9	4080	4081	4082
10	4090	4091	4092
11	4100	4101	4102
12	4110	4111	4112
13	4120	4121	4122
14	4130	4131	4132
15	4140	4141	4142
16	4150	4151	4152
17	4160	4161	4162
18	4170	4171	4172
19	4180	4181	4182
20	4190	4191	4192
21	4200	4201	4202
22	4210	4211	4212
23	4220	4221	4222
24	4230	4231	4232

Note the changed port allocation from earlier releases, for channel 5 and above.

Reader's Notes

APPENDIX C: DTMF, FAX, AND MODEM MODES

DTMF Relay Settings

You can control the way DTMF digits are transported to the remote endpoint, using the DTMFTransport configuration parameter. The following four modes are supported:

- **DTMFTransportType= 0 (MuteDTMF)** – In this mode, DTMF digits are erased from the audio stream and are *not* relayed to the remote side. Instead silence is sent in the RTP stream.

- **DTMFTransportType= 1 (Proprietary DTMF Relay)** – In this mode, DTMF digits are erased from the audio stream and are relayed to the remote side using a proprietary RTP syntax.

- **DTMFTransportType= 2 (Transparent DTMF)** – In this mode, DTMF digits are left in the audio stream and the DTMF relay is disabled.

- **DTMFTransportType= 3 (RFC 2833 DTMF Relay)** – In this mode, DTMF digits are relayed to the remote side using the RFC 2833 Relay syntax.

Fax/Modem Settings

You can choose to use one of the following transport methods for Fax and for each modem type (V.21/V.22/V.23/Bell/V.32/V.34):

- **Fax relay** – Demodulation / modulation

- **Bypass** – Using a high bit rate coder to pass the signal

- **Transparent** – Passing the signal in the current voice coder

When the fax relay mode is enabled, distinction between fax and modem is not immediately possible at the beginning of a session. The channel is therefore in Answer Tone mode until a distinction is determined. The packets being sent to the network at this stage are fax relay packets (The packets can be either T.38-complaint, or FRF.11-based proprietary syntax, selected by setting the channel's configuration parameter UseT38orFRF11).

Configuring Fax Relay Mode

When FaxTransportType= 1 (relay mode), then on detection of fax, the channel automatically switches from the current voice coder to answer tone mode, and then to fax relay mode. The UseT38orFRF11 configuration parameter defines either T.38-compliant

network packets or proprietary FRF.11-based packets (the last mode should be used mostly for backward-compatibility with previous software versions).

When the fax transmission has ended, the reverse switching from fax relay to voice is performed. This mode switching automatically occurs at both the local and remote endpoints.

You can limit the fax rate using the FaxRelayMaxRate parameter and can enable/disable ECM fax mode using the FaxRelayECMEnable parameter.

When using T.38 mode, you can select between two protection strategies – redundancy packets or forward error correction (FEC). This selection is made using the T38FaxRelayProtectionMode configuration parameter. You can also control a special (proprietary) redundancy mode that was specially designed to improve protection against packet loss using the EnhancedFaxRelayRedundancyDepth parameter. Although this is a proprietary redundancy scheme, it is compatible with other T.38 decoders. When using FRF.11 mode, only redundancy packets are supported. The depth of the redundancy in both protocols (that is, the number of repetitions) is defined by the FaxRelayRedundancyDepth configuration parameter.

> *T.38 mode currently supports only the T.38 UDP syntax.*

Configuring Fax/Modem ByPass Mode

When VxxTransportType= 2 (FaxModemBypass, Vxx can be one of the following: V32/V22/V21/Bell/V34/Fax), then on detection of Fax/Modem, the channel automatically switches from the current voice coder to a high bit-rate coder, as defined by the user, with the FaxModemBypassCoderType configuration parameter.

If relay is enabled for one of the modes (Fax/Modem), then the Answer Tone mode packets are relayed as fax relay packets.

During the bypass period, the coder uses the packing factor (by which a number of basic coder frames are combined together in the outgoing WAN packet) set by the user in the FaxModemBypassM configuration parameter. The network packets generated and received during the bypass period are regular voice RTP packets (per the selected bypass coder) but with a different RTP Payload type.

When Fax/Modem transmission ends, the reverse switching, from bypass coder to regular voice coder, is carried out.

> *When Fax relay is enabled, V21TransportType must be set to disable (Transparent) mode.*

Supporting V.34 Faxes

V.34 faxes do not comply with the T.38 relay standard. 3Com therefore provides the optional modes described in "Using Bypass Mechanism for V.34 Fax Transmission " and "Using Relay Mode for Both T.30 and V.34 Faxes".

Note that the CNG detector is disabled (CNGDetectorMode=0) in all the following examples.

Using Bypass Mechanism for V.34 Fax Transmission

In this proprietary scenario, the Gateway uses a high bit-rate coder to transmit V.34 faxes, enabling the full utilization of its speed.

Use the following configurations as a guide:

```
FaxTransportMode = 2 (Bypass)
V34ModemTransportType = 2 (Modem bypass)
V32ModemTransportType = 2
V23ModemTransportType = 2
V22ModemTransportType = 2
V21ModemTransportType = 2
```

In this configuration, both T.30 and V.34 faxes work in Bypass mode.

Or

```
FaxTransportMode = 1 (Relay)
V34ModemTransportType = 2 (Modem bypass)
V32ModemTransportType = 2
V23ModemTransportType = 2
V22ModemTransportType = 2
V21ModemTransportType = 2
```

In this configuration, T.30 fax uses T.38 Relay mode while V.34 fax uses Bypass mode.

Using Relay Mode for Both T.30 and V.34 Faxes

In this scenario, V.34 fax machines are compelled to use their backward compatibility with T.30 faxes; as a T.30 machine, the V.34 fax can use T.38 Relay mode.

Use the following configuration as a guide:

```
FaxTransportMode = 1 (Relay)
V34ModemTransportType = 0 (Transparent)
V32ModemTransportType = 0
V23ModemTransportType = 0
V22ModemTransportType = 0
V21ModemTransportType = 0
```

Both T.30 and V.34 faxes use T.38 Relay mode. This configuration forces the V.34 fax machine to operate in the slower T.30 mode.

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APPENDIX D: DHCP SERVER CONFIGURATION

Windows NT DHCP Server Configuration in BootP compatible (Reserve) Mode
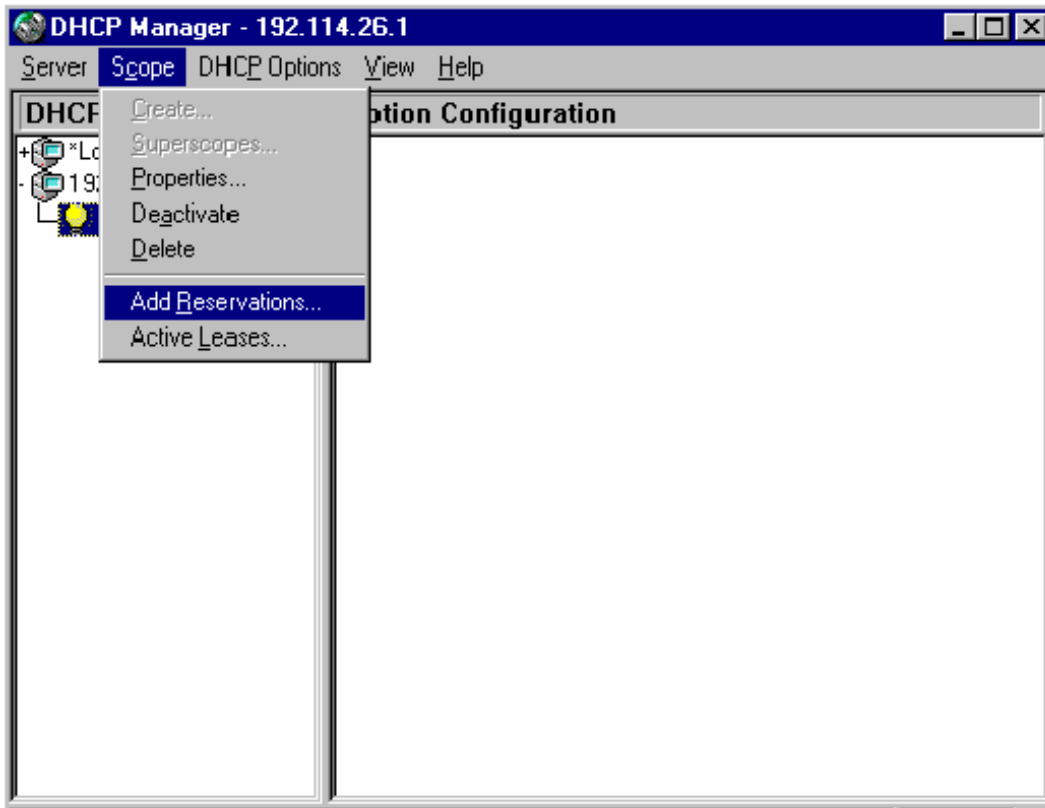
> *You need to install Windows NT4 services pack 4 (or higher)* after *enabling the DHCP server service on the NT server. This is required for correct operation with BootP clients.*

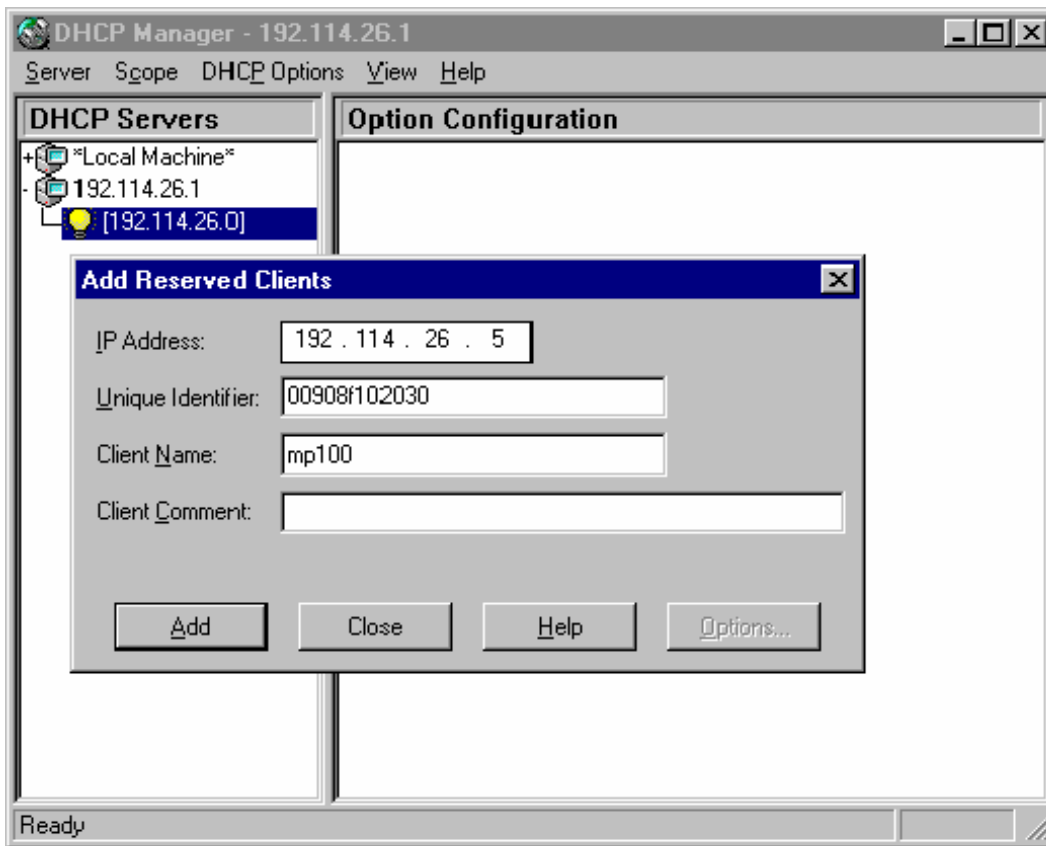To configure the Windows NT DHCP server, follow these steps:

1 Start the DHCP Manager on your computer. DHCP Manager opens the following screen:
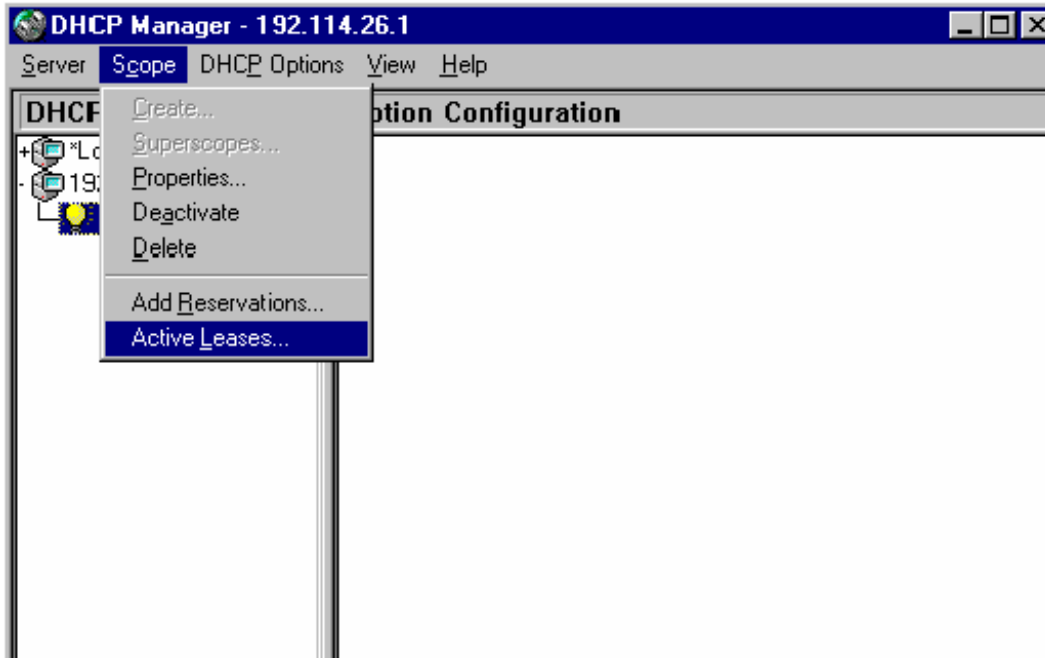
Figure 65 Scope Menu

2 Select *Add Reservations* from the *Scope* menu; the Add Reserved Clients screen opens.

Figure 66   Add Reserved Clients

3   Enter the IP address you want to provide to the 3Com Gateway.

4   The IP address reservation should be inside your DHCP Scope.

5   Enter the hardware MAC address, 12 digits, from your 3Com Gateway.

6   Enter Client name; it can be any free text.

7   Select *Add* and then *Close*.

8   From *Scope* menu in the main screen, select *Active Leases*.

Figure 67   Active Leases Select Screen

The Active Leases screen displays the following selection box:

Figure 68   Active Leases Selection Box

9   Click the *Properties* button.

Figure 69   Client Properties Screen

10 Click *Options* to display the DHCP Options: Reservation screen.

Figure 70   DHCP Options: Reservation Screen.

11 Add the following extension fields from the list in the left pane of the DHCP Options: Reservation screen:

- 003 Router (Default router/Gateway)

- 012 Host Name (BootP client name such as the 24FXS or other customer-selected name)

- 066 Boot Server Host Name (Domain name or IP address of the TFTP server)

- 067 Bootfile Name (Such as ram.cmp)

> *The file ram.hex is a loadable software image file. The file ram.cmp is a compressed version of the ram.hex, enabling faster download and reduced file size. The ram.cmp is the only version that can be transferred to flash memory.*

The selected extension fields are then displayed in the right pane of the DHCP Options: Reservation screen under Active Options, as the following figure shows.

Figure 71   Active Options

12 Select the 003 Router field and click the *Value* button.

13 Edit the IP Address Array in the IP Address Array Editor screen.

Reader's Notes

APPENDIX E: OBTAINING SUPPORT FOR YOUR 3COM PRODUCTS

3Com offers product registration, case management, and repair services through eSupport.3com.com. You must have a user name and password to access these services, which are described in this appendix.

Register Your Product to Gain Service Benefits

To take advantage of warranty and other service benefits, you must first register your product at:

http://esupport.3com.com/

3Com eSupport services are based on accounts that are created or that you are authorized to access.

Solve Problems Online

3Com offers these support tools:

- **3Com Knowledgebase** — Helps you to troubleshoot 3Com products. This query-based interactive tool is located at:

    http://knowledgebase.3com.com/

    It contains thousands of technical solutions written by 3Com support engineers.

- **Connection Assistant** — Helps you to install, configure, and troubleshoot 3Com desktop and server network interface cards (NICs), wireless cards, and Bluetooth devices. This diagnostic software is located at:

    http://www.3com.com/connectionassistant

Purchase Extended Warranty and Professional Services

To enhance response times or extend your warranty benefits, you can purchase value-added services such as 24x7 telephone technical support, software upgrades, onsite assistance, or advanced hardware replacement.

Experienced engineers are available to manage your installation with minimal disruption to your network. Expert assessment and implementation services are offered to fill resource gaps and ensure the success of your networking projects. For more information on 3Com Extended Warranty and Professional Services, see:

http://www.3com.com/

Contact your authorized 3Com reseller or 3Com for additional product and support information. See the table of access numbers later in this appendix.

Access Software Downloads

You are entitled to *bug fix / maintenance releases* for the version of software that you initially purchased with your 3Com product. To obtain access to this software, you need to register your product and then use the Serial Number as your login. Restricted Software is available at:

http://esupport.3com.com/

To obtain software releases that follow the software version that you originally purchased, 3Com recommends that you buy an Express or Guardian contract, a Software Upgrades contract, or an equivalent support contract from 3Com or your reseller. Support contracts that include software upgrades cover feature enhancements, incremental functionality, and bug fixes, but they do not include software that is released by 3Com as a separately ordered product. Separately orderable software releases and licenses are listed in the 3Com Price List and are available for purchase from your 3Com reseller.

Contact Us

3Com offers telephone, internet, and e-mail access to technical support and repair services. To access these services for your region, use the appropriate telephone number, URL, or e-mail address from the table in the next section.

Telephone Technical Support and Repair

To obtain telephone support as part of your warranty and other service benefits, you must first register your product at:

http://esupport.3com.com/

When you contact 3Com for assistance, please have the following information ready:

- Product model name, part number, and serial number

- A list of system hardware and software, including revision level

- Diagnostic error messages

- Details about recent configuration changes, if applicable

3Com VCX V7111 VoIP SIP Gateways User Manual

To send a product directly to 3Com for repair, you must first obtain a return materials authorization number (RMA). Products sent to 3Com without authorization numbers clearly marked on the outside of the package will be returned to the sender unopened, at the sender's expense. If your product is registered and under warranty, you can obtain an RMA number online at http://esupport.3com.com/. First-time users must apply for a user name and password.

Telephone numbers are correct at the time of publication. Find a current directory of 3Com resources by region at:

http://csoweb4.3com.com/contactus/

Country	Telephone Number	Country	Telephone Number
Asia, Pacific Rim — Telephone Technical Support and Repair			
Australia	1 800 678 515	Pakistan	+61 2 9937 5083
Hong Kong	800 933 486	Philippines	1235 61 266 2602 or 1800 1 888 9469
India	+61 2 9424 5179 or 000800 650 1111	P.R. of China	800 810 3033
Indonesia	001 803 61009	Singapore	800 6161 463
Japan	00531 616 439 or 03 5977 7991	S. Korea	080 333 3308
Malaysia	1800 801 777	Taiwan	00801 611 261
New Zealand	0800 446 398	Thailand	001 800 611 2000

You can also obtain support in this region at this e-mail address: apr_technical_support@3com.com

Or request a return material authorization number (RMA) by FAX using this number: +61 2 9937 5048

Europe, Middle East, and Africa — Telephone Technical Support and Repair			

From anywhere in these regions, call: +44 (0)1442 435529

From the following countries, call the appropriate number:

Country	Telephone Number	Country	Telephone Number
Austria	01 7956 7124	Luxembourg	342 0808128
Belgium	070 700 770	Netherlands	0900 777 7737
Denmark	7010 7289	Norway	815 33 047
Finland	01080 2783	Poland	00800 441 1357
France	0825 809 622	Portugal	707 200 123
Germany	01805 404 747	South Africa	0800 995 014
Hungary	06800 12813	Spain	9 021 60455

Country	Telephone Number	Country	Telephone Number
Ireland	01407 3387	Sweden	07711 14453
Israel	1800 945 3794	Switzerland	08488 50112
Italy	199 161346	U.K.	0870 909 3266

You can also obtain support in this region using this URL:
http://emea.3com.com/support/email.html

Latin America — Telephone Technical Support and Repair

Country	Telephone Number	Country	Telephone Number
Antigua	1 800 988 2112	Guatemala	AT&T +800 998 2112
Argentina	0 810 444 3COM	Haiti	57 1 657 0888
Aruba	1 800 998 2112	Honduras	AT&T +800 998 2112
Bahamas	1 800 998 2112	Jamaica	1 800 998 2112
Barbados	1 800 998 2112	Martinique	571 657 0888
Belize	52 5 201 0010	Mexico	01 800 849CARE
Bermuda	1 800 998 2112	Nicaragua	AT&T +800 998 2112
Bonaire	1 800 998 2112	Panama	AT&T +800 998 2112
Brazil	0800 13 3COM	Paraguay	54 11 4894 1888
Cayman	1 800 998 2112	Peru	AT&T +800 998 2112
Chile	AT&T +800 998 2112	Puerto Rico	1 800 998 2112
Colombia	AT&T +800 998 2112	Salvador	AT&T +800 998 2112
Costa Rica	AT&T +800 998 2112	Trinidad and Tobago	1 800 998 2112
Curacao	1 800 998 2112	Uruguay	AT&T +800 998 2112
Ecuador	AT&T +800 998 2112	Venezuela	AT&T +800 998 2112
Dominican Republic	AT&T +800 998 2112	Virgin Islands	57 1 657 0888

You can also obtain support in this region in the following ways:

- Spanish speakers, enter the URL: http://lat.3com.com/lat/support/form.html

- Portuguese speakers, enter the URL: http://lat.3com.com/br/support/form.html

- English speakers in Latin America, send e-mail to: lat_support_anc@3com.com

US and Canada — Telephone Technical Support and Repair

All locations:	Network Jacks; Wired or Wireless Network Interface Cards:	1 847-262-0070
	All other 3Com products:	1 800 876 3266