VH2

VH2 Quickstart Guide

The VH2 is a studio telephone interface device designed to work on VoIP telephone systems. This guide will give you the minimum you need to know to get the system hooked up and running. For more details on how to configure more complex arrangements, see the full user manual.

Specifications

The VH2 can handle one or two VoIP accounts (a single account can be assigned to both channels).

The unit has the ability to easily transfer the call on either channel back and forth to a companion VoIP phone.

Caller output audio is available of balanced XLR-M. Send input is available on balanced XLR-F. Clip levels on XLR feeds is +20dBu.

On-hold audio input is available on a ¼" TRS jack. Two channels may be fed to "on-hold" callers via "Tip-sleeve" (channel 1) and "Ring-sleeve" (channel 2). These inputs are unbalanced and the clip level is +20dBu.

VH2 has a single 10/1000 Ethernet jack for network connection. The companion phone should be located on the same physical LAN as the VH2 (no routers between them).

The approved companion phone (**Polycom IP 331**) can be powered from PoE networks. The VH2 cannot. The companion phone contains a 2 port Ethernet switch, so the units can be wired as follows:

PoE Ethernet Jack ->Polycom phone->VH2.

If a PoE is not available, you must use the provided power adapter with the Polycom phone.

Requirements

To configure and start using the VH2, you'll need the following information:

1 The user name, password, and server address of a cloud-based VoIP service, an available account on your PBX, or similar information in order to utilize VH2 on a gateway device.

2 An available static IP on your network, including gateway, netmask and DNS server information.

3 Knowledge of how the VH2 will be attached to your studio audio ports. The choices are:

   a VH2 can take a single "send" feed from your console, or two separate "send" feeds.

   b VH2 can deliver a single "caller" feed to your console, or two separate "caller" feeds.

   c VH2 can be fed a single "on hold" input, two separate "on hold" inputs, or none at all.

   d VH2 can be configured to allow callers to hear each other when "on-air", or they can be isolated from each other.

   e "Send" and "caller" feeds can be connected via analog or AES3 signals.

Steps

Here's a list of the steps that need to be accomplished to start using VH2. We'll outline each in detail below:

1. Using the Comrex **Device Manager** utility, set static IP info into the VH2.

2. If using the companion telephone set, enter the VH2 static IP info into the telephone set.

3. Using a computer browser, set up the provider details (cloud provider, pbx) into the VH2 **Toolbox** interface.

4. Using the **Toolbox** interface, configure the VH2 "behavior" to match your desired functions in the studio.

5. Understand how the front panel buttons and indicators work.

Step 1 - Static IP

While it's not technically required for VH2 to have a static IP address, it is required for use with the companion phone, and if the DHCP address changes you'll need to find the new address in order to configure the VH2 via the **Toolbox** interface.

VH2 is shipped from the factory set to DHCP mode, so it will find an address on your network if possible. The easiest way to find what that address is to use the Comrex **Device Manager** utility, available on our website for both Windows and Mac platforms. You can also use **Device manager** to upgrade the firmware on your VH2.

Install **Device Manager** on a computer on the same physical LAN as the VH2. Once installed and run, press the "**scan**" button in the upper left corner to scan your network for Comrex products. The results should look like this:
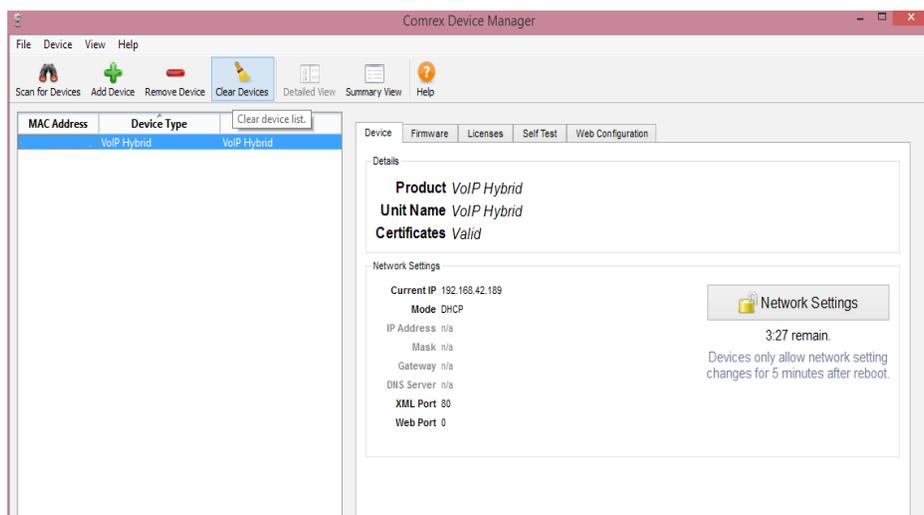
There are two ways to set the static IP information using **Device Manager**. The "rescue" mode allows you to change parameters during the first five minutes of operation, and is initiated using the "**Network Settings**" button on the right pane. This mode is most useful if you've forgotten the unit password. The normal mode of changing any settings in VH2 is by clicking the "**Web Configuration**" tab in the right pane. This opens the unit's **Toolbox** interface. **Toolbox** is a web-based interface that allows you to set all relevant parameters of the product. Once you know the unit's IP address, **Toolbox** can also be accessed via a browser by going to **http://[ip-address]/cfg**.

Toolbox requires that you log on with a user name and password. The default user name is '**admin**' and the default password is '**comrex**'. We recommend you change the default password, especially if your VH2 is exposed directly to the Internet.

Once logged in, choose the '**Network**' option. Under '**Network Mode**', you'll see the following options:
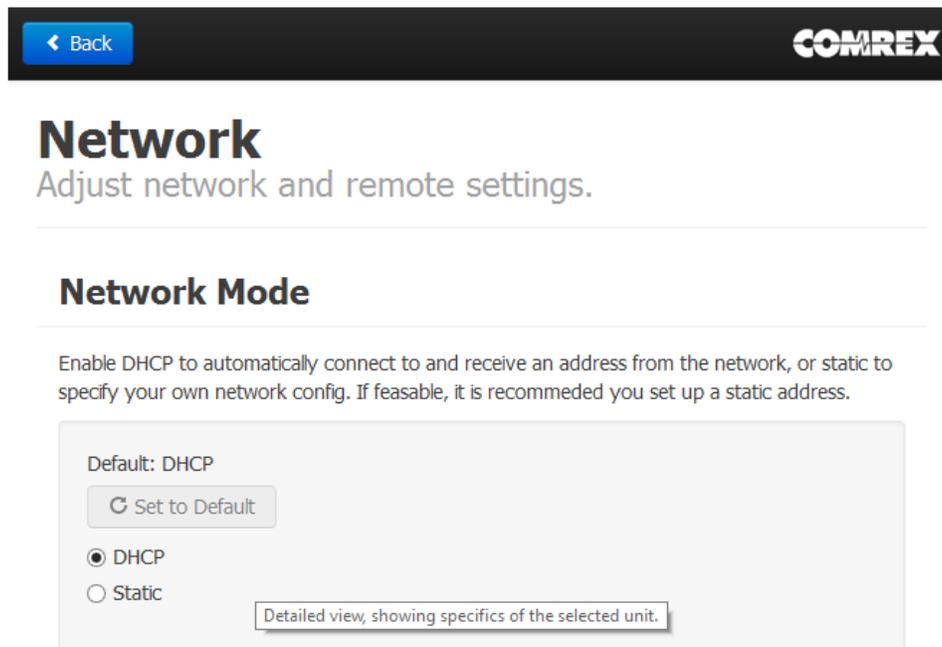
Change the mode to **Static**, and input the IP address, netmask, gateway and DNS information for your network. Click '**apply IP settings**' after all the information is correct. If you need to access **Toolbox** after this, you'll need to rescan and log in again (or open a browser to the new IP address).

Step 2 - Companion phone

If you have the **IP 331** IP phone for use with VH2, you'll need to set it up to be an extension of the VH2. The phone needs to be on the same physical LAN as the VH2, but it doesn't need a static IP address. You'll only need to program the static address of the VH2 into the phone.

Once the phone is on your LAN, it will get a DHCP address from the network. You can find what this is by pressing the menu button on the phone and navigating to the following using the direction cursor and enter key:

2)Status->2)Network->1)TCP/IP Parameters

Open a browser on your local computer and input the IP address of the phone into the URL bar. You will be prompted to log in to the phone's web interface. Default username is '**Polycom**' and default password is '**456**'.

Time setting

To stop the annoying time display from flashing an incorrect time and date, set a default SNTP server and time zone on the phone. Navigate to **General->Time** and set the fields to read the following:

SNTP Server-> **pool.ntp.org** (or any other SNTP server of your choosing)

GMT Offset-> The offset of your time zone in hours from GMT (e.g. US Eastern = -5, US Pacific = -8)

Time	
Synchronization	
SNTP Server	pool.ntp.org
GMT Offset	-5 ▼
SNTP Resync Period	86400
Daylight Savings	

VH2 settings

To put the details of your VH2 into the phone, choose the '**Lines**' tab on the top menu of the configuration page:
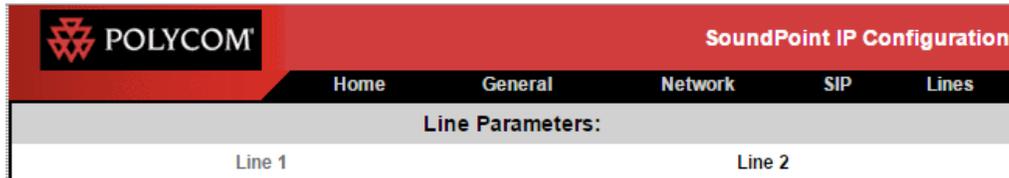
Note the two sub-sections, **Line 1** and **Line 2**. You will need to program information into *both* of these pages.

On the Line 1 page, set the following fields:

Under *Identification*:

Display Name -> 'Line 1'

Address-> '1100@<VH2_ip_address>' (remove ' and <>)

Authentication User ID -> '1100'

Authentication Password -> (default="456" - can be changed in Toolbox)

Label ->'Line 1'

Type->Private

Third Party Name->'none'

Number of Line Keys->1

Calls per Line->1

Under *Server 1*:

Address-><VH2_ip_address> (remove <>)

Port->5170

Transport->DNSnaptr

Expires->100

Register->1

Retry Timeout->0

Retry Maximum Count->3

Line Seize Timeout->30

(leave the rest of the fields at default)

Your Line 1 Page should read like this (but with your VH2 IP address)

Line 1	
Identification	
Display Name	Line 1
Address	1100@192.168.42.15
Authentication User ID	1100
Authentication Password	••••
Label	Line 1
Type	⦿ Private ◯ Shared
Third Party Name	none
Number Of Line Keys	1
Calls Per Line	1
Server 1	
Address	192.168.42.15
Port	5170
Transport	DNSnaptr ▼
Expires	100
Register	1
Retry Timeout	0
Retry Maximum Count	3
Line Seize Timeout	30

Click Submit on the bottom of the page, then go to line 2

On the Line 2 page, set the following fields:

Under *Identification*:

Display Name -> 'Line 2'

Address-> '1200@<VH2_ip_address>' (remove ' and <>)

Authentication User ID -> '1200'

Authentication Password -> (default="456" - can be changed in Toolbox)

Label ->'Line 2'

Type->Private

Third Party Name->'none'

Number of Line Keys->1

Calls per Line->1

Under *Server 1*:

Address-><VH2_ip_address> (remove <>)

Port->5170

Transport->DNSnaptr

Expires->100

Register->1

Retry Timeout->0

Retry Maximum Count->3

Line Seize Timeout->30

(leave the rest of the fields at default)

Your Line 2 Page should read like this (but with your VH2 IP address)

Line 2	
Identification	
Display Name	Line 2
Address	1200@192.168.42.15
Authentication User ID	1200
Authentication Password	••••
Label	Line 2
Type	⦿ Private ○ Shared
Third Party Name	none
Number Of Line Keys	1
Calls Per Line	1
Server 1	
Address	192.168.42.15
Port	5170
Transport	DNSnaptr ▾
Expires	100
Register	1
Retry Timeout	0
Retry Maximum Count	3
Line Seize Timeout	30

Click the Submit button on the bottom of the page.

You are finished with the telephone setup.

Step 3 - SIP Details

To use VH2, you must input the account details for at least one voice provider or PBX extension. You can have up to two providers, and it's also possible to establish two accounts with the same provider, or on the same PBX.

Provider input is done in the **Toolbox** interface, accessible either by the Comrex **Device Manager** utility (see step 1) or pointing a browser to **http://[ip_address]/cfg**. As a reminder, the default login credentials are **admin/comrex**.

In **Toolbox**, navigate to **Channel Configuration->VoIP Providers->Add Provider**. Here you'll be given a choice between the types of VoIP accounts VH2 can handle. We'll focus here on setting up a PBX or cloud provider account (other types are described in the manual) so choose the "**SIP Provider**" option.

Here you will enter the SIP credentials for your account. The name field is just for reference and can be anything. Under "**SIP Provider**", you can choose from several popular providers, and the menus will change to help you with the most common settings. If your provider is not listed (or you're using a PBX), choose "**Generic Provider**".

At a minimum, a provider or PBX account needs a *user name*, *password*, and *server address* to register. The server address can take the form of a domain name (myprovider.net) or an IP address. Some providers require additional details like *outgoing proxy* and *auth username*. If required, these fields are available under the "show advanced" button, or may be exposed based on the type of provider chosen.

Once all your account information is correct, the "**apply changes**" button will appear. Select this to begin the registration process. Note the VH2 SIP subsystem will reset, so don't do this with a call "on-air". Once the system has reset, VH2 will attempt to register with your provider and display status on the provider list.

Once you're registered with at least one provider, you can assign the VH2's "channels" to that provider.

Important: Simply registering is not enough. VH2's channel(s) must be assigned to your provider(s) before you can use it.

You can have up to two providers, and each provider can be assigned to one or both channels. Channel assignment is done in **Toolbox** via **Channel Configuration->Channel Assignments**.

The figure shows each channel assigned to a different provider. Each channel could also be assigned to the same provider, and the usual behavior is for calls to "hunt" between the channels (first call on Channel 1, second on 2).

Step 4 - Setting up behavior

Back at the main **Toolbox** page, selecting '**System Behavior**' allows you to set how the VH2's audio I/O interacts with your studio. The options are:

Auto-Answer

You can individually set each channel to auto-answer or not. If set to auto-answer, you can choose whether each channel puts each call to "on-air" state or "on-hold" state.

Studio Audio I/O

Here you can select how many input and output audio ports are used on the VH2. The choices are:

- **Single In/Single Out** - Only the channel 1 input and channel 1 outputs are used. All callers hear channel 1 in and all callers audio is output on channel 1 out.

- **Single In/Dual Out** - The channel 1 input is used, but both channel 1 and 2 outputs are used. Caller audio appears separately on these channels (to be sent to two separate console faders, for example). Both callers hear audio sent to the channel 1 in.

- **Dual In/Dual Out** - Both channel 1 and channel 2 inputs and outputs are used. Caller 1 hears channel 1 in and appears on channel 1 out. Caller 2 hears channel 2 in and appears on channel 2 out.

Note that none of these settings has any effect on whether callers hear each other. That function is chosen in the next option.

About Mix Minus

When we refer to "send" audio to the caller, we're talking about the feed that it attached to the VH2 channel 1 and 2 inputs. This is the audio that the caller hears when "on-air". It is essential that these feeds are specially mixed so that the caller output is not part of that mix. Many consoles provide for a special audio bus especially for this function. On some other consoles, you can use the secondary or audition bus (with callers deselected) to create this feed. But under no circumstances must incoming callers hear themselves sent back, or they will experience an annoying echo.

Caller Mix

Determines whether the callers can hear each other in the "conference" built within the VH2. If your console provides individual mix-minus for telephone channels, this should be off, as the mix will be done in your console. If the console doesn't handle this, and you are using a single mix-minus for both channels, turn this function on so the callers will be conferenced within the VH2.

AGC

Selects whether VH2 applies AGC to the caller outputs, helping minimize large level changes between calls.

Caller Ducking

Selects whether VH2 applies an algorithm to the caller audio to reduce it when the host voice is detected. This allows the host to "dominate" the conversation. Ducking, if enabled, has three choices (**Low, Medium, High**) that allows selection of how much the caller is reduced when the host speaks.

Caller On-Air Tone

Selects whether the caller hears a short "beep" when put "on-air". This is sent to the caller only and is not heard "on-air".

Contact Closures

See full user manual for details.

Step 5 - Operating the VH2

DIP Switch Settings

The front panel DIP switches have the following function:

1   Analog in (down) / AES3 in (up)

2   Analog out (down) / AES3 out (up)

3   unused

4   Level Indicators reflect Send audio (down)

    Level Indicators reflect Caller audio (up)

5, 6   Unused

7, 8   Factory Reset (see manual)

ON/OFF Button (ch 1 and 2)

This is a toggle which determines if the call is active on the selected channel. When pressed, an incoming call will be answered. This button will also "grab" a currently active call on the companion telephone and place it on the desired channel. Calls selected this way are considered "on-air" and caller audio is routed via the selected behavior settings. Pressing this button when a call is active drops the call, whether "on-air" or "on-hold".

HOLD/XFR (ch 1 and 2)

This is a toggle that places an active call "on-hold" from the "on-air" state. The caller will be removed from the main audio ports, and hear only the audio presented to the VH2 "on-hold" input. Pressing this button while an incoming call is ringing will send the caller directly to "on-hold" state. Pressing this button while a call is "on-hold" will send the call back to "on-air" state.

If a call is active (either "on-hold" or "on-air") a long press of this button (over two seconds) will transfer the call to the companion telephone set. The set will ring, but the call will remain in its previous state ("on-hold" or "on-air") until the handset is answered.

Indications

The **ON/OFF** and **HOLD/XFR** buttons are lit to indicate the state of each particular channel

ON/OFF	HOLD/XFR	Function
Off	Off	Idle-registered and ready for call
Blink Green	Off	Ringing
Green	Off	On-Air
Green	Green	Hold
Blink Green	Blink Green	On Handset
Red	Off	Auto-Answer
Red	Red	No Provider Assigned
Blink Red	Off	Provider Assigned, but not registered

The "Ready" Indicator is used to signal the overall network condition. Its states are:

Off - System is starting up

Fast Blink - Loading application

Red - No Internet connectivity detected

Green - System has IP address on network

General Operation

Incoming calls

Calls will ring the handset (if used) and blink the **ON/OFF** indicator for the channel with the incoming call. Calls can be answered via the handset, or placed directly "on-air" or "on-hold" via the front panel button. Calls on the handset can be put "on-air" or "on-hold" via the same buttons.

Outgoing calls

Place calls on either channel 1 or 2 via the handset. Place them "on-air" or "on-hold" with the appropriate channel buttons.

Transferring to Handset

From "on-air" or "on-hold", long press the **HOLD/XFR** button. The handset will ring. Answer it and the call will be transferred (note the call is still in the current state until handset is answered).

Ending calls

From "on-air" or "on-hold" state, press the **ON/OFF** button to end the call. From the handset, simply hang up.