Cisco CE Console for CE 8.0

CUSTOMIZING AUDIO CONNECTIONS FOR AN ENHANCED AUDIO EXPERIENCE

 | © 2016 Cisco Systems, Inc. All rights reserved.

Cisco CE Console

WHAT'S IN THIS USER GUIDE?

Cisco CE Console for CE 8.0 1

Cisco CE Console 3
- Audio Console 3
- Starting the CE Console 4
- Defining a new project 5
- Synchronizing with the codec 6
- About the Audio Console 7
- Invoking the Audio Console 7
- About virtual and physical modules 8
- Connecting to a physical source 8
- Housekeeping–removing unused sources 8
- This applies to output modules as well 8
- Introduction 9
- Global settings 9
- Individual settings 9
- About the equalizer functionality 10
- A tour of the equalizer editor 11
- Creating a simple peaking filter 12
- Adding another peaking filter 13
- Using the numerical settings 14
- About outputs 15
- Add and delete connections 15
- About the gain setting 16
- Add new modules or rename existing? 17
- Adding an input module 17
- Add new modules or rename existing? 18
- Adding an output module 18
- Assign physical outputs to the module 18
- The Settings 18

General purpose input/output 19
- About GPIO 19

 | © 2016 Cisco Systems, Inc. All rights reserved.

Cisco CE Console

CUSTOMIZING AUDIO CONNECTIONS FOR AN ENHANCED VIDEO EXPERIENCE

The Cisco CE Console

The Cisco CE Console tool lets you customize the audio routing of the Codec SX80.

It will let you create setup profiles that can be applied to the codec at a later time or you can configure the system in real time.

Audio Console

The Audio Console helps you configure the audio system of the codec. It will allow you to change the default mixing, routing and equalization as well allow you to set various input and output connector properties.

 | © 2016 Cisco Systems, Inc. All rights reserved.

Starting the CE Console

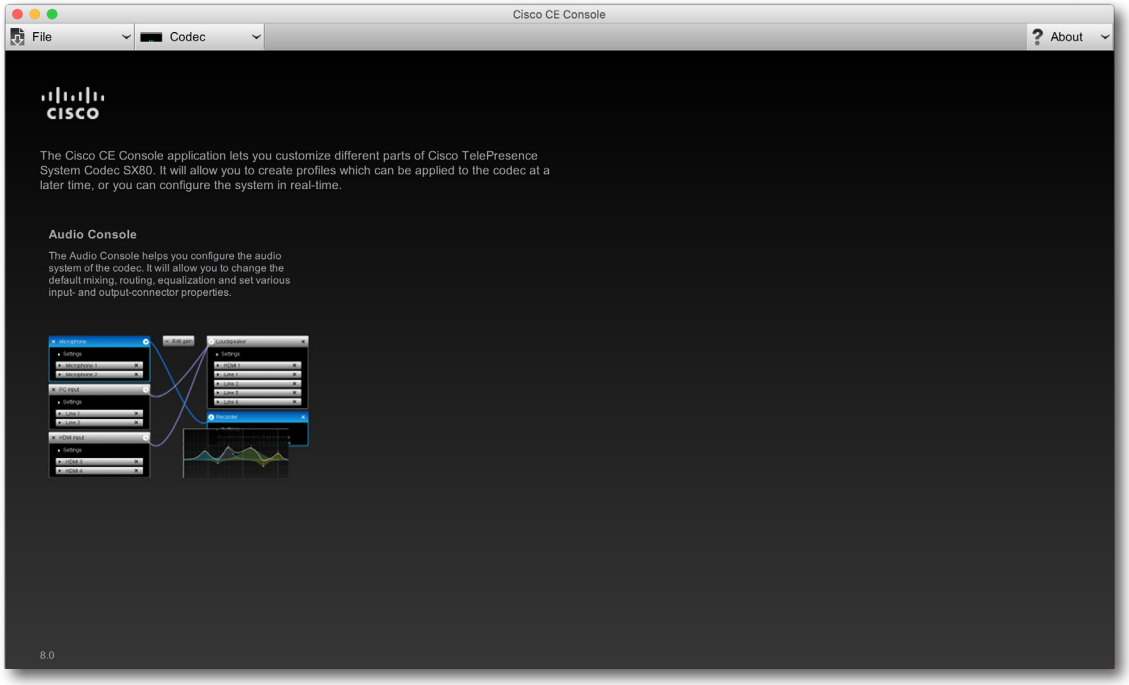
The Cisco CE Console can be downloaded for free from developer.cisco.com.

Once started, the welcome screen will look as shown at right.

There is no need to have the codec connected while you do the setup. You can if you wish, but you don't have to. Your settings can be uploaded to the codec when everything is set and ready.

Any setup previously stored on the codec may, of course, be downloaded, modified and then uploaded back to the codec, as well.

The result of your work will inevitably change the codec's behavior (and that is intentional). The settings defined using the CE Console will run as a stand-alone utility within the codec and become an integral part of the codec behavior until it is changed again.

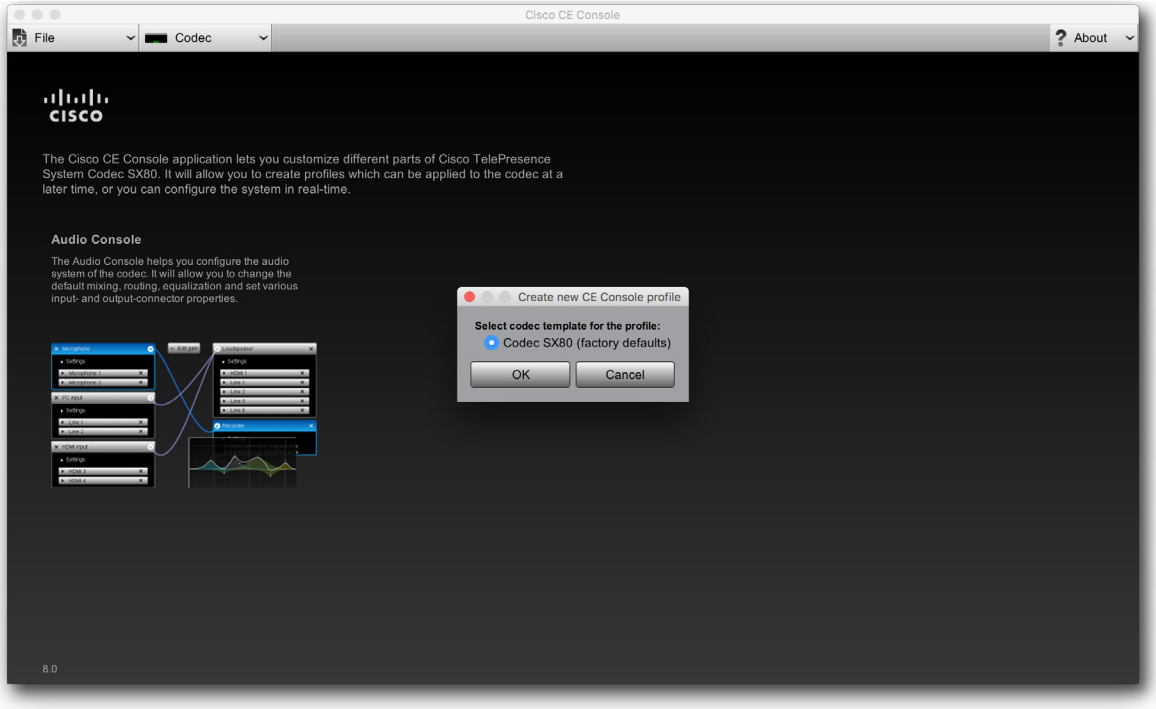


Defining a new project

The first time you run the CE Console utility you will start by clicking on File > New and select which codec you are going to create this for. In this version SX80 is the only codec available.

The template you now choose will be based on factory default settings.

Later you will be able to base a new profile on any existing profile. You will then use Open instead of New.



 | © 2016 Cisco Systems, Inc. All rights reserved.

Synchronizing with the codec

When the CE Console utility is connected to a codec, synchronization will take place automatically. This means that any changes made in the Audio Console will immediately be uploaded to the codec through the synchronization process.

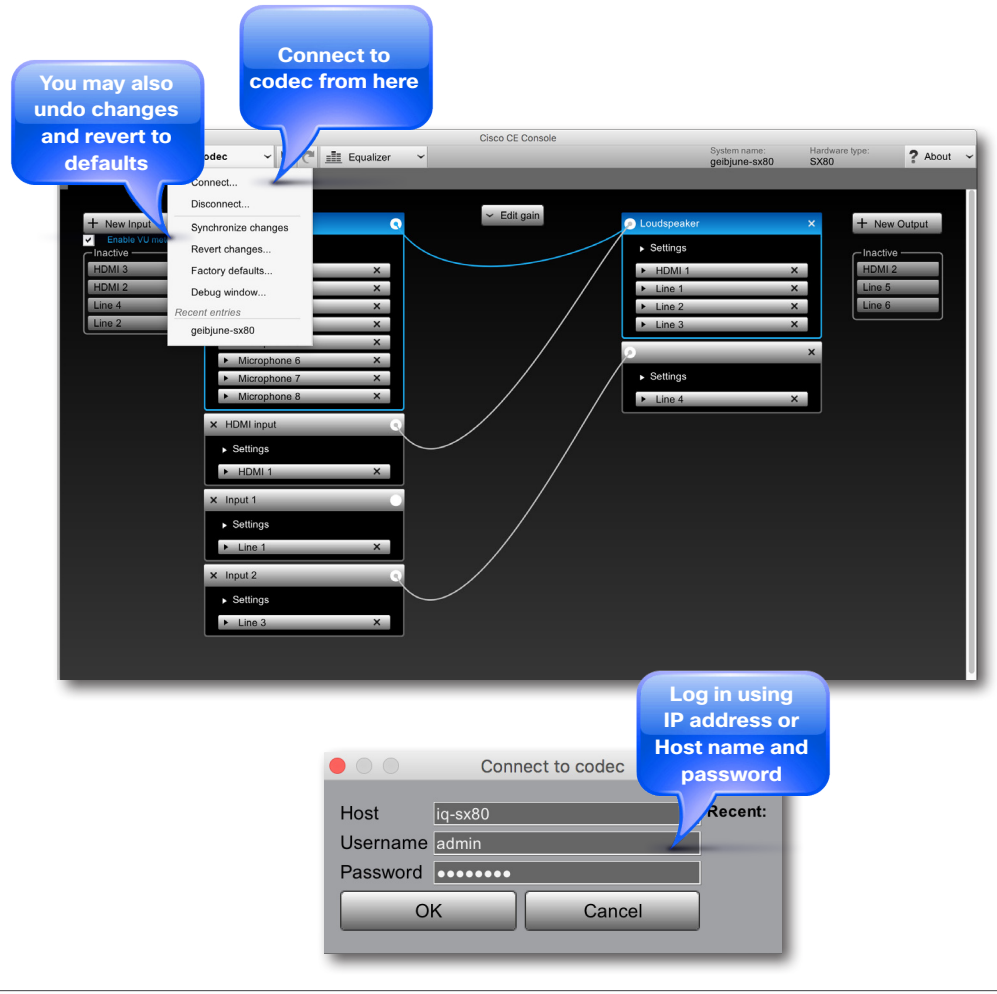
This works both ways; as soon as connection has been established, any more recent configuration stored in the codec will be copied to the CE Console utility.

You may not want live synchronization to take place. You should then disconnect from the codec before you start working with the CE Console. Your configuration project can be stored on your computer like any other file. When all settings and alterations have been made, just connect to the codec and the synchronization process will take care of the uploading.

However, sometimes you may want to create settings whose effects you would like to monitor immediately. This could typically be “cut and try” adjustments using the Audio console equalizer settings (see page 10). In such cases you should stay connected with the codec.

Establishing connection with the codec:

1. Click on **Codec** > **Connect**.
2. You will now be prompted to key in the IP address (or the Host name) and the password to the codec.
3. As soon as connection has been established, the configuration stored in the codec will be copied to the CE Console utility.



 | © 2016 Cisco Systems, Inc. All rights reserved.

The Audio Console

AN OVERVIEW

About the Audio Console

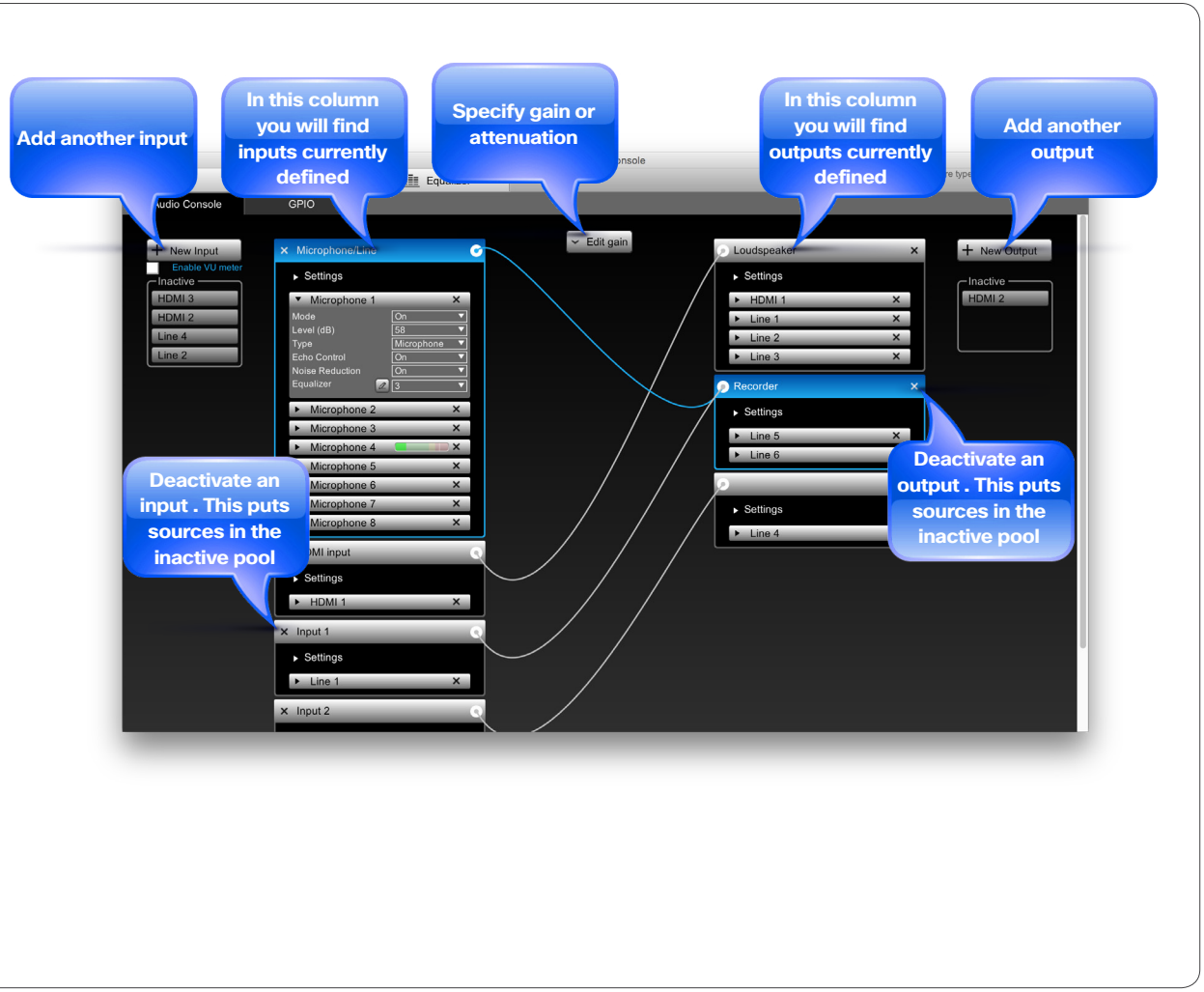
The Audio Console lets you configure local audio connections without the need for programming the codec.

As can be seen from the Fig., the Audio Console is a visual representation of audio chains showing how audio input sources are connected to audio output sources.

An audio chain consists of a physical input socket assigned to a virtual input module (see the next page for more on virtuality) in turn connected to a virtual output module to which a physical output has been assigned.

The virtual modules are used to impose such things as gain, equalization and channel mixing to the signal.

Note that this is a local setup only–the Audio Console has no influence whatsoever on the audio sent to the other side. That will be taken care of by the codec itself in the usual manner.



Invoking the Audio Console

Do as follows:

1. Click File > New or File > Open to either start a new project or open an existing project.
2. Select which codec this shall apply to.
3. Click Audio Console.

Assigning physical sockets to virtual modules

About virtual and physical modules

An input module, such as the **PC input** module, is a virtual input module in the sense that is a non-physical module that is used to specify what shall to be done to the signal passing through.

The idea behind the Audio Console is to give you the possibility to change the default routings and behavior of the local audio part of the codec. When you modify an input module, you will specify where the processed signal is going when it leaves the input module, as well as specify from where the signal physically is coming. In other words, from which physical input socket of the codec.

Connecting to a physical source

The codec SX80 comes with a fixed number of physical input sockets. Any of these can be assigned to any virtual input module.

Note that a given physical source can be assigned to a single input module only.

However, you may assign more than one physical input source to an input module and mix them there. In addition, any input module can be connected to any number of output modules. The first time you start a new Audio Console project all physical inputs have been assigned to input modules leaving none available for reassignment or succesful inclusion of new modules, so what to do?

The answer is that you should then borrow a physical input from one of the other modules.

Just click on the physical input source you want to assign and drag it into the other module, as outlined in the Fig.

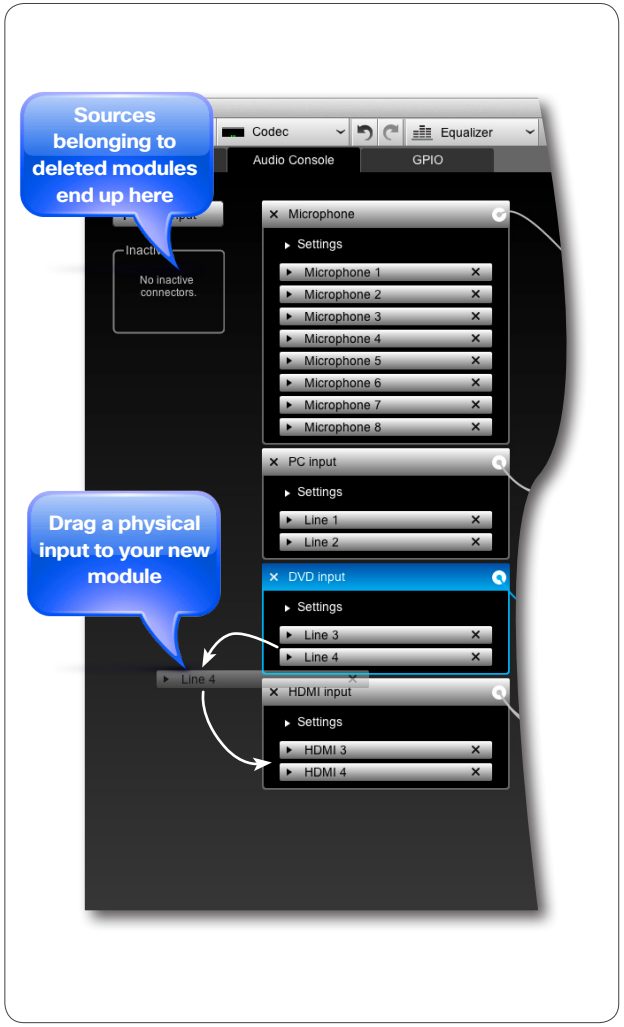
Housekeeping—removing unused sources

When you have added your new modules and all is set up, a little housekeeping may be advisable to retain a clean Audio Console desktop. Consequently, we recommend that you remove an unused input module by clicking on the ×.

Physical input sources assigned to this module will then be put in the **Inactive** pool automatically, making them available for future use.

This applies to output modules as well

The above discussion applies to the output modules as well, of course. The difference lies merely in that the virtual output modules receive the signal from one or more virtual input modules and after processing convey the signal to one or more of the codec's physical outputs.



 | © 2016 Cisco Systems, Inc. All rights reserved.

Modifying existing modules

USING MICROPHONE INPUT MODULE AS AN EXAMPLE

Introduction

We are going to use the Microphone settings as an example, but the procedure is valid for all inputs.

The Euroblock/Phoenix equipped microphone inputs are all combined line and microphone inputs and the Audio Console offers you the option to choose whether an input shall be used as Mic or Line.

There are two sets of settings applying here; Global settings for the group and Individual settings for each individual input.

The Codec SX80 has 8 Mic/Line inputs, 4 line inputs, and 3 HDMI inputs.

Global settings

Mixer mode, which is either:

- Fixed, which means that the mixed signals all have the same weight and consequently remain unchanged.
- Auto, which means that if one sound source is dominating, the other sources are attenuated by 20 dB each.
- GainShared, which means that if one or some of the sound sources dominate, the rest of the sources are attenuated by a few dB so that the total gain of the mixed signal remains the same all the time. This can be regarded as a soft version of the Auto setting.

Mute, which is either On or Off.

AGC (Automatic Gain Control) which is either On or Off. The AGC will adjust the gain continuously to maintain a constant sound level.

Channels, which is either 1 (mono) or 2 (stereo).

Individual settings

- Mode, which is either On or Off.
- Level (dB), which lets you specify the amount of input gain. Reference level (0 dB) depends on whether Type is set to Microphone or to Line.
- If Type is set to Microphone, 0 dB input level corresponds to –36 dBu.
- If Type is set to Line, 0 dB input level corresponds to +6 dBu.
- Echo Control, which can be set to either On or Off.
- Noise Reduction, which can be set to either On or Off.
- Equalizer, which can be set to either Off or to one of a total of 8 presets. To define these presets see the following pages.

Note that if you apply the above thinking to a module connected to HDMI, the individual settings available will be Mode and Level only.

 | © 2016 Cisco Systems, Inc. All rights reserved.

About the equalizer functionality

The system has 8 user-definable equalizers, each made up of 6 second order IIR filter sections.

These 6 filter sections can be modified independently of each other by setting the four parameters:

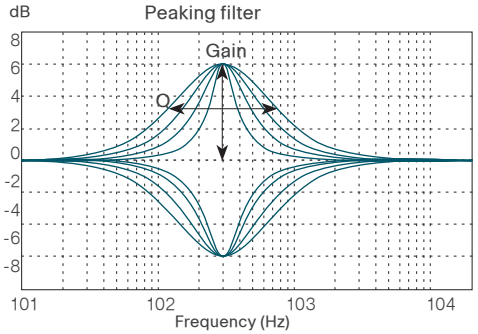
- **Filter types**. The filter types available are peaking (band pass / band stop), low shelf, high shelf, low pass and high pass.
- **Gain**. Available range is –20.00 to +20.00 dB in steps of 0.01 dB.
- **Frequency**. Available range is 20–20 000 Hz in steps of 1 dB.
- **Q-value**. Available range is 0.13–100.00 in steps of 0.01.

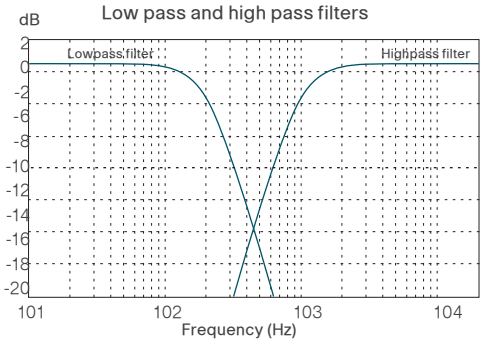
You may set these parameters by dragging and dropping, by specifying them numerically, or combine the two to the extent you want.

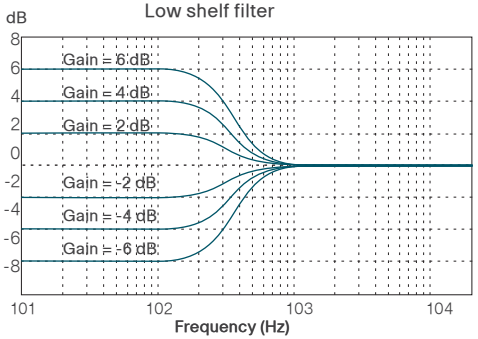
Each of these 8 equalizers can be applied to one or more of the audio inputs of the codec, but only one per input–they cannot be cascaded.

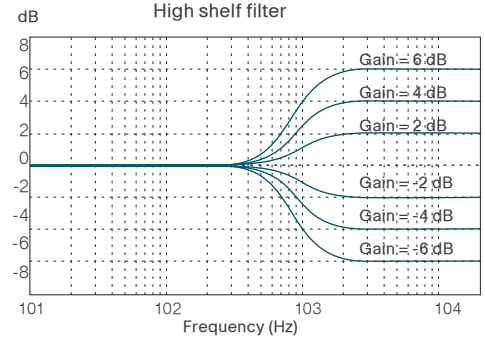
For maximum flat response, the Q-value should be set to $1/\sqrt{2}$. for the high- and low-pass filters, as well as for the high- and low-shelf filters.

The Q-value of a filter is defined as the center frequency divided by the –3 dB filter bandwidth.









 | © 2016 Cisco Systems, Inc. All rights reserved.

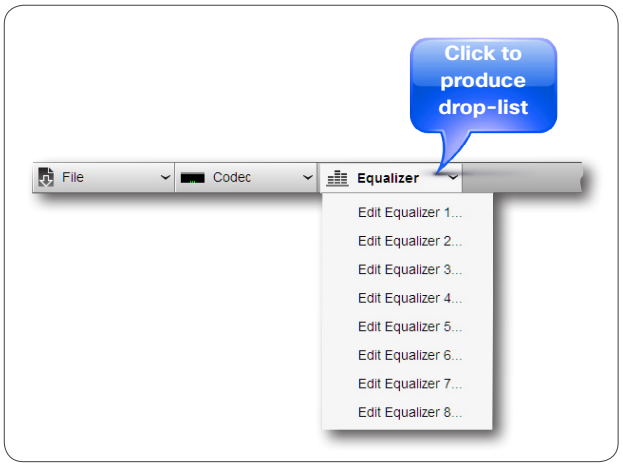
A tour of the equalizer editor

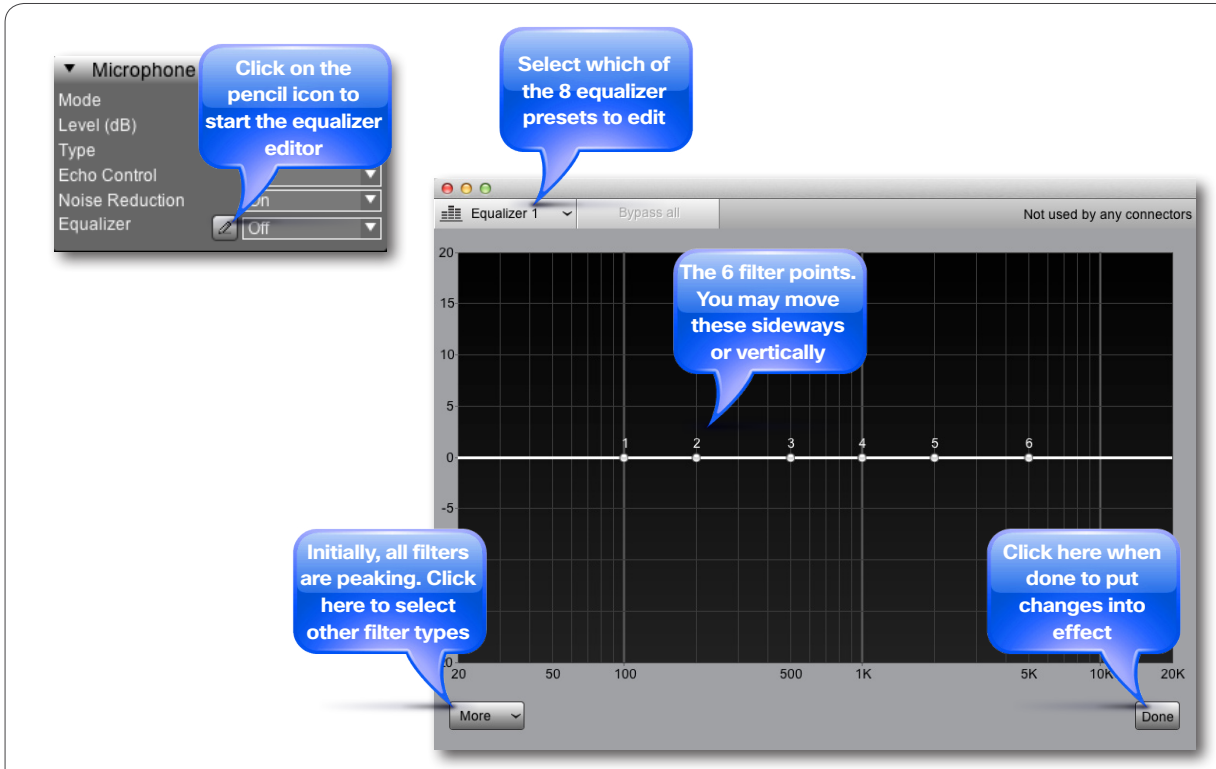
To begin editing a frequency response, either click on Equalizer along the top menu bar (see below), or click on the icon (see right). This will launch the Equalizer editor.

You have 8 independent equalizers to choose between. Select which to edit.

The editor displays a line with 6 uniformly distributed points. Click on a point to select it.

Initially, all filters are of the peaking type (band pass / band stop). If you need to employ high pass, low pass, high shelf or low shelf filters instead you will have to enter the More menu.





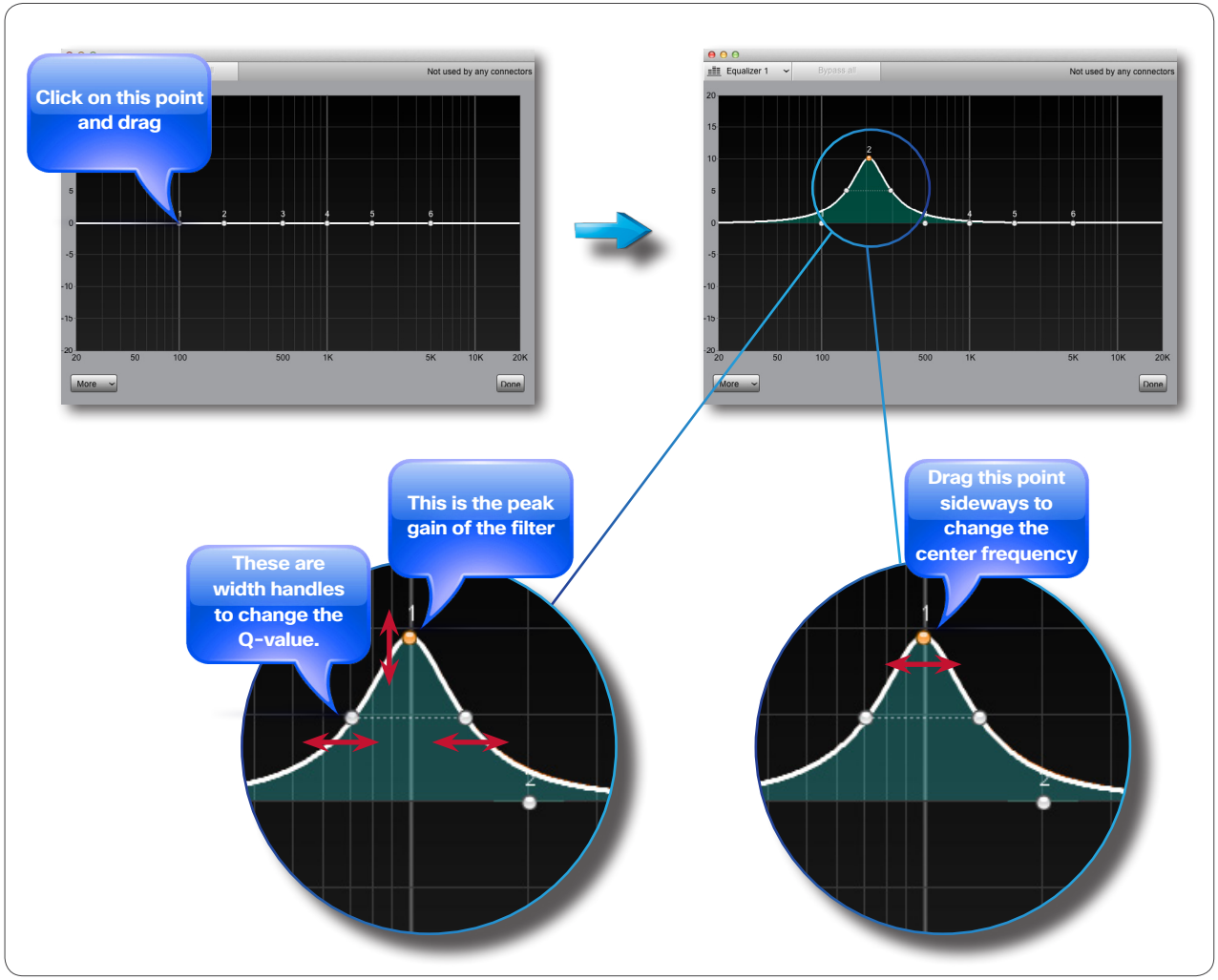
Creating a simple peaking filter

As an introductory example we shall run through the process of creating a simple peaking (band pass) filter.

Such a filter is characterized by the center frequency f_c, the gain and the Q-value.

Do as follows:

1. Click and drag on any of the six points along the frequency axis. The one selected turns yellow and will move as you drag.
2. Move vertically to set the gain and sideways to change the center frequency.
3. Use the handles to modify the Q-value.



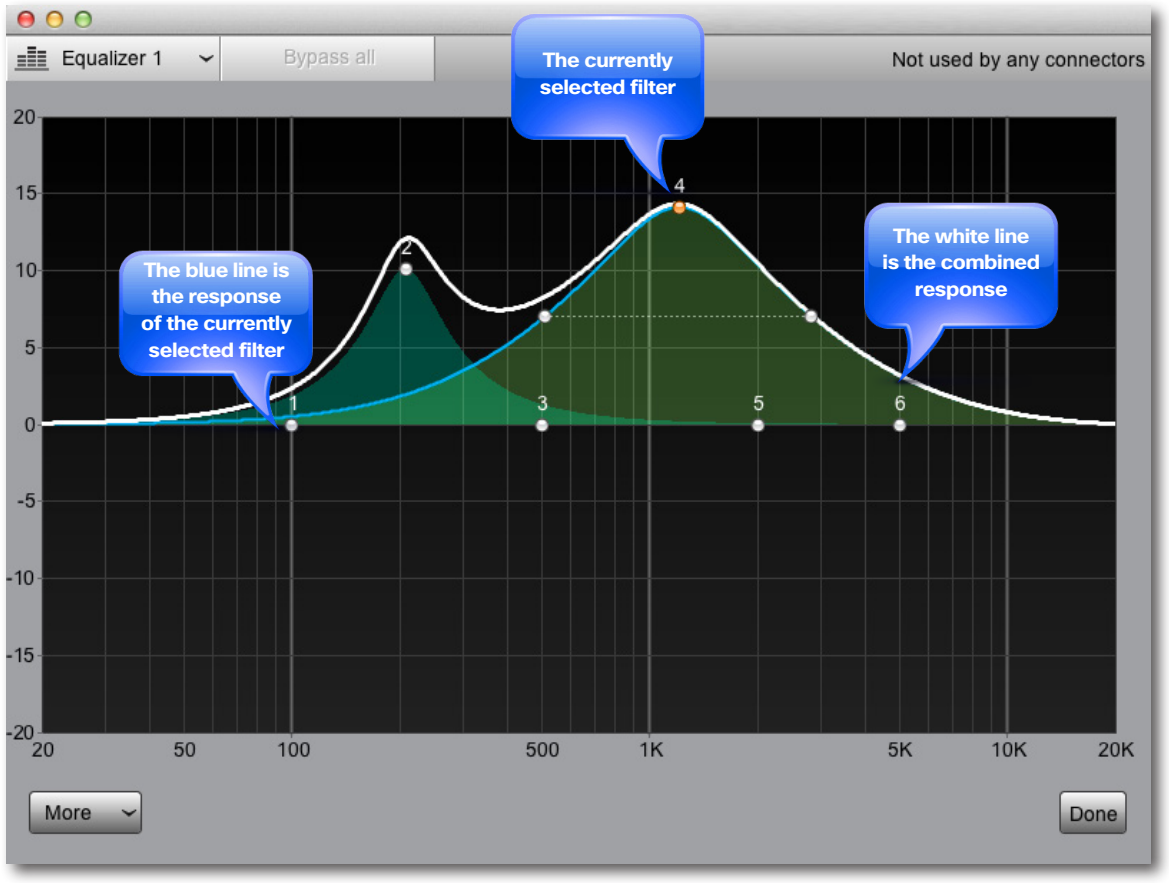
 | © 2016 Cisco Systems, Inc. All rights reserved.

Adding another peaking filter

By clicking on any of the five remaining unused points along the frequency axis, you may add another peaking filter. The procedure is identical to the one described on the previous page.

Note how the new filters is shown along with the combined response.

Any of these may be dragged below the 0 dB line to create a band stop type filter as well.



Using the numerical settings

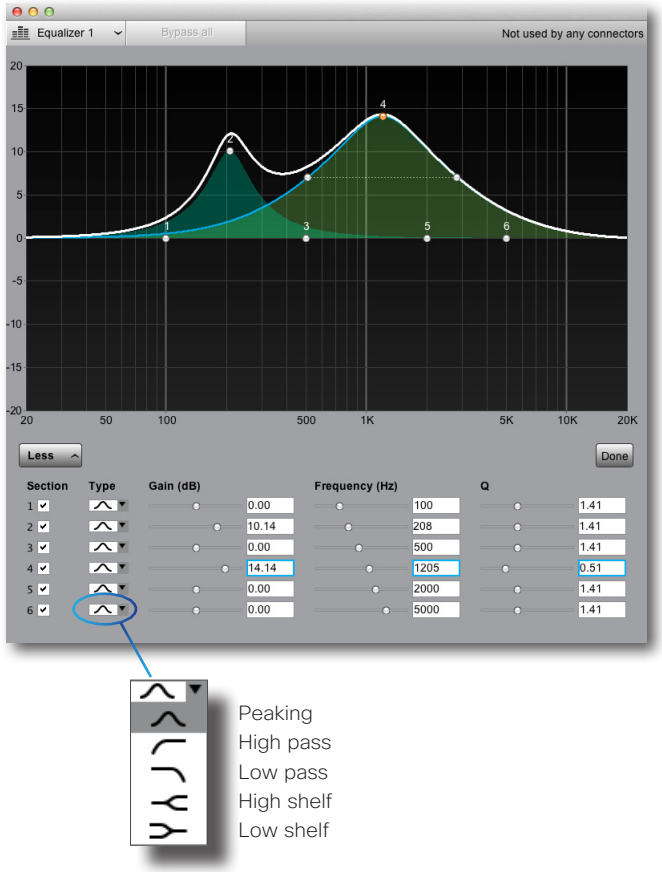
Initially, all 6 filters are set to peaking (band pass / band stop). You may, however, transform any filter to a high pass, low pass, high shelf or low shelf filter instead. In addition you may control each filter's settings numerically.

To do this access the numerical settings by clicking on the More button. The editor will now look as shown.

There is a line for each of the six filter sections, allowing you to:

- Active / deactive that section
- Define filter type (see Fig.)
- Set a gain for the filter
- Set the center frequency / cut-off frequency for the filter
- Set the Q-value for the filter

You may combine the graphical tools with the numerical setting options as much as you want. Changing one updates the other accordingly.



 | © 2016 Cisco Systems, Inc. All rights reserved.

About outputs

Remember that the Audio Console lets you set up local audio connections only, it has no influence on what is sent to the other side.

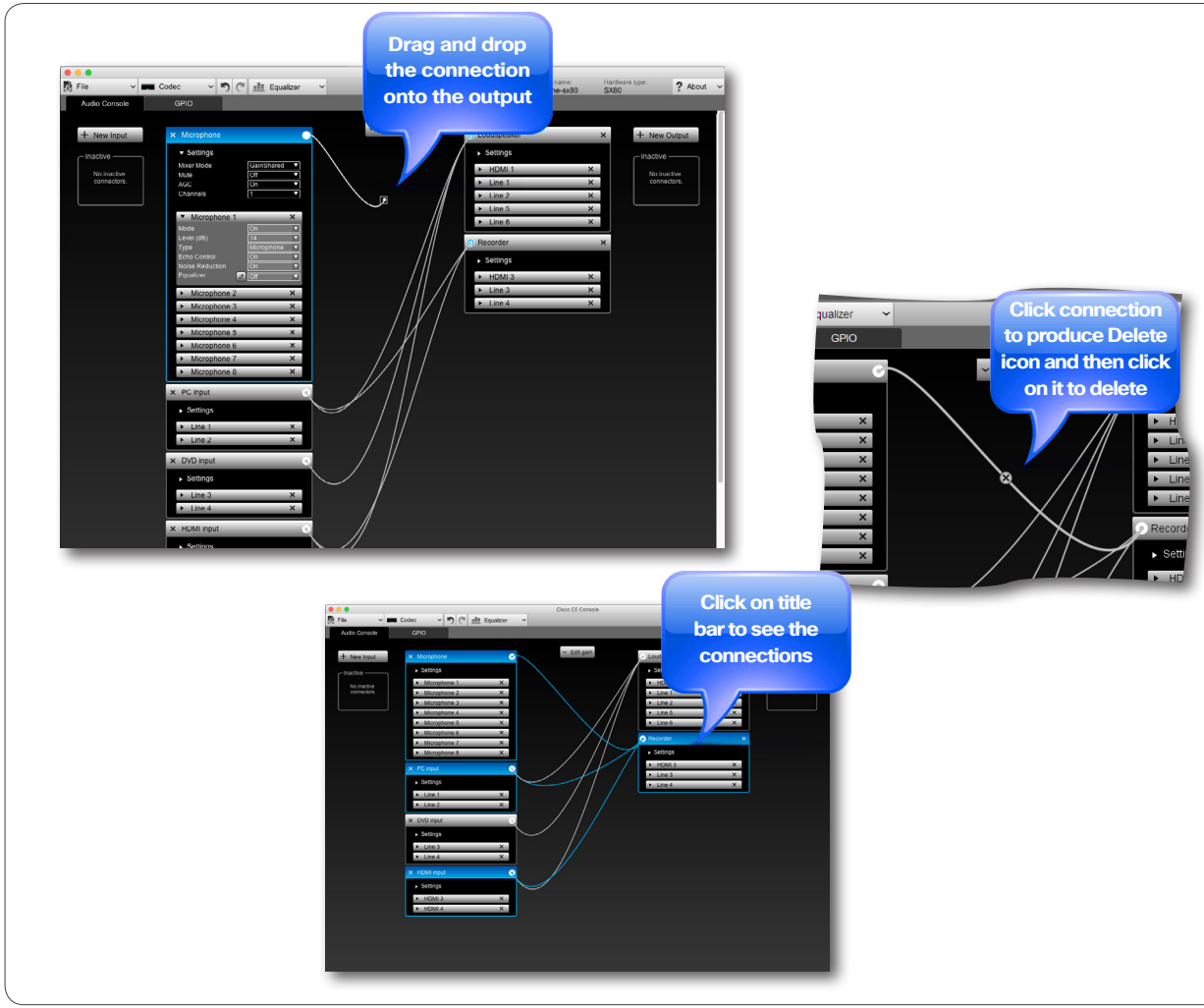
Initially, the Audio Console has two default output sections; Loudspeaker and Recorder. If you want to use others, you may add them any time. Adding outputs is described on page 18.

Add and delete connections

As an example we will connect the Microphones to the Recorder. To connect an input to an output, click on the outbox of the input and drag the connection to the inbox of the output you want to connect to.

If you want to remove a connection, click on the “wire” to produce the icon and the click this icon to delete the connection.

To see which input section(s) any given output is connected to (or vice versa, of course), just click on the corresponding title bar, as shown.



 | © 2016 Cisco Systems, Inc. All rights reserved.

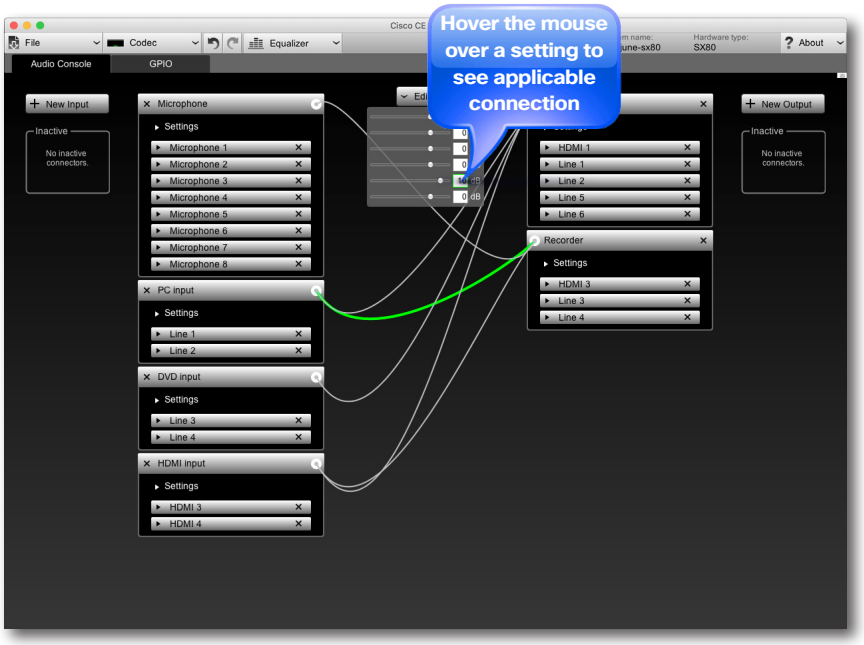
About the gain setting

You may add gain to any connection between input and output. Note that gain can also be negative, i.e. attenuation. The gain range is –54 dB to +15 dB.

To expand the **Gain setting** menu, click on it. To contract it, click on the header again.

There is one gain setting for each connection. To see which gain setting corresponds to which connection, just hover over a gain setting with the mouse. The corresponding connection will be shown in green.

Set the gain using the slider, or by specifying it directly in the numerical input field.



 | © 2016 Cisco Systems, Inc. All rights reserved.

Adding or reassigning input modules

Add new modules or rename existing?

If you are going to reconfigure the current system rather than expanding it to add functionality, there is no need to add new input modules. You may then just as well rename an existing one and use that as your new module.

To change the name of an existing input module, just double-click on its name (for nameless modules double-click where the name will appear when modules have names).

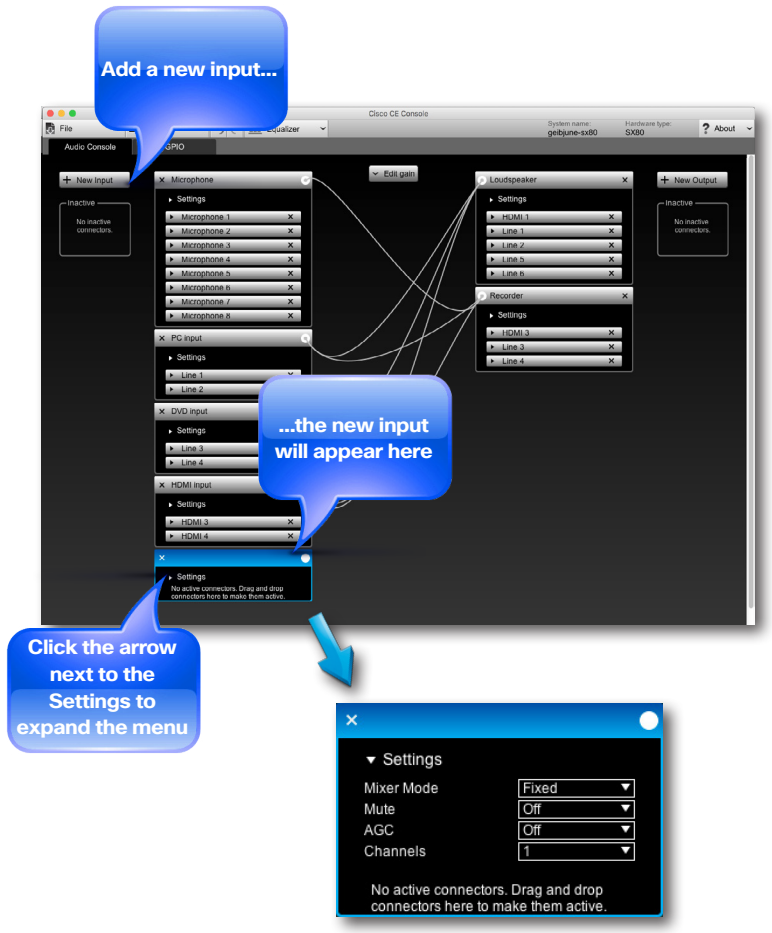
Adding an input module

To create a new input module click New Input. A new input will appear at the bottom, as shown. Give the new local input a name and click the arrow to expand the Settings field. To name a module later, double-click the name field.

The settings are similar to those outlined in Global Settings on page page 9. Set as required.

The next thing to do will be to specify from which physical input this input shall get its signal. This is described on page 8.

To delete this input module, click on the ×.



 | © 2016 Cisco Systems, Inc. All rights reserved.

Adding or reassigning output modules

Add new modules or rename existing?

If you are going to reconfigure the current system rather than expanding it to add functionality, there is no need to add new output modules. You may then just as well rename an existing one and use that as your new module.

To change the name of an existing output module, just double-click on its name (for nameless modules double-click where the name will appear when modules have names).

Adding an output module

To create a new output click New Output. A new output will appear at the bottom, as shown. Give the new local output a name and click the arrow to expand the Settings field. To name a module later, double-click the name field.

As for the input modules, there are two sets of settings available; Global Settings for the group and Individual Settings for this particular module. All these options are treated on the next page.

To delete this output module, click on the ×.

Assign physical outputs to the module

The individual settings available for an output module depends on which physical output socket(s) you assign to it. Read more about this on page 9.

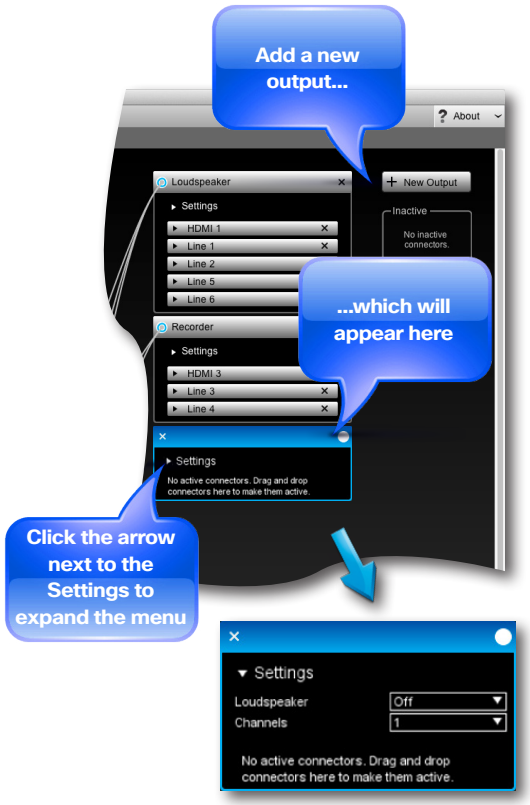
Consequently, we recommend that you assign (or reassign) physical output sockets to the module before you alter the settings. For a description on how to do this, see page 8.

The Settings

The Global settings for an output module are:

- Loudspeaker, which can be set to either On or Off.
- Channels, which can be either 1 (mono) or 2 (stereo).

The Individual settings depend on the physical output sockets you assign to the module. See page 9 for more on this.



 | © 2016 Cisco Systems, Inc. All rights reserved.

General purpose input/output

CONFIGURING THE GPIO FUNCTIONS

About GPIO

The GPIO Console allows you to configure and view the state of the GPIO connectors found in the Codec SX80.

To access this setup, click GPIO on the menu bar.

The GPIO socket is designed for 6-pin Phoenix plugs with +12 V, GND and 4 ports for On/Off control.

Initially, a GPIO port is high (2–12 VDC) until a certain condition occurs, on which it will go low (0–1 VDC). Observe that maximum output current from the +12 V is 500 mA @ 12 VDC, while a port is capable of sinking 500 mA @ 48 VDC. This means that in most cases the use of an external power supply to run external devices will be preferable, compared to driving an external unit from the codec's +12VDC power.

The following options exist:

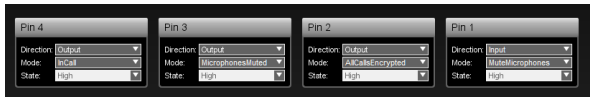
Direction can be Output or Input

Output Direction:

- Manual State can be used to force the output State to go high or low. Since the codec will operate on its own, without the CE Console program controlling it, Manual State will not apply unless you use API to set up conditions.
- InCall. The port will go low when the system is in a call.
- MicrophonesMuted. The port will go low when the microphones are muted.
- PresentationOn. The port will go low when presentation (second video stream) is active.
- AllCallsEncrypted. The port will go low when the all calls are encrypted.
- StandbyActive. The port will go low when the system is active (not in standby).

Input Direction

- MuteMicrophones. When the port goes low, the microphones will be muted.



 | © 2016 Cisco Systems, Inc. All rights reserved.

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS.

THE SOFTWARE LICENSE AND LIMITED WARRANTY FOR THE ACCOMPANYING PRODUCT ARE SET FORTH IN THE INFORMATION PACKET THAT SHIPPED WITH THE PRODUCT AND ARE INCORPORATED HEREIN BY THIS REFERENCE. IF YOU ARE UNABLE TO LOCATE THE SOFTWARE LICENSE OR LIMITED WARRANTY, CONTACT YOUR CISCO REPRESENTATIVE FOR A COPY.

The Cisco implementation of CEP header compression is an adaptation of a program developed by the University of California, Berkeley (UCB) as part of UCB's public domain version of the UNIX operating system. All rights reserved. Copyright © 1981, Regents of the University of California.

NOTWITHSTANDING ANY OTHER WARRANTY HEREIN, ALL DOCUMENT FILES AND SOFTWARE OF THESE SUPPLIERS ARE PROVIDED "AS IS" WITH ALL FAULTS. CISCO AND THE ABOVE-NAMED SUPPLIERS DISCLAIM ALL WARRANTIES, EXPRESSED OR IMPLIED, INCLUDING, WITHOUT LIMITATION, THOSE OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE AND NONINFRINGEMENT OR ARISING FROM A COURSE OF DEALING, USAGE, OR TRADE PRACTICE.

IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

Cisco and the Cisco Logo are trademarks of Cisco Systems, Inc. and/or its affiliates in the U.S. and other countries. A listing of Cisco's trademarks can be found at www.cisco.com/go/trademarks. Third party trademarks mentioned are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (1005R)

Any Internet Protocol (IP) addresses and phone numbers used in this document are not intended to be actual addresses and phone numbers. Any examples, command display output, network topology diagrams, and other figures included in the document are shown for illustrative purposes only. Any use of actual IP addresses or phone numbers in illustrative content is unintentional and coincidental.

Cisco and the Cisco Logo are trademarks of Cisco Systems, Inc. and/or its affiliates in the U.S. and other countries. A listing of Cisco's trademarks can be found at www.cisco.com/go/trademarks. Third-party trademarks mentioned are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (1007R)