

CHANNEL D Pure Music®

User's Guide - NetSend

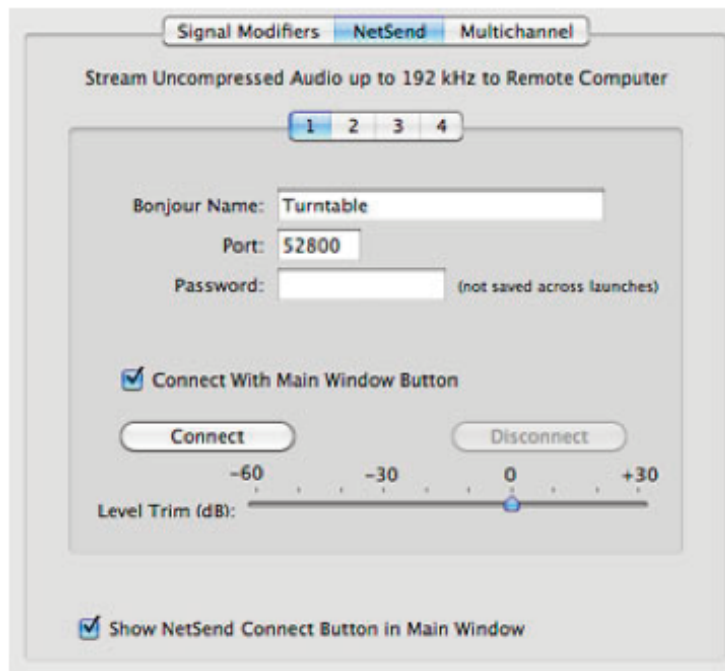
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<http://www.channel-d.com>

CHANNEL D

NetSend

NetSend streams high resolution audio (up to 192 kHz / 24 bit) at very low latency to a remote Mac computer with an audio interface. A multichannel audio interface at the receiving end can be used to drive different sound playback zones, each with their own EQ, signal delay (very useful), and volume settings. If using Pure Music's crossover feature, the full-frequency-range signal is transmitted. (Note: a wired 100 or 1000 BT Ethernet connection is recommended).



The Mac computer receiving audio need not have the same type of audio interface, and need not even be running at the same sample rate as the sending Mac computer.

The CPU requirements also are minimal at the receiving end (500 MHz or faster G4, G5, Intel processor; Mac OS X 10.4 or later). An old Mac is perfect for this purpose.

Up to four separate named streams can be sent to four separate computers. Each computer, in turn, can output multiple audio channels to different zones, limited only by the number of output channels in the audio interface.

The example below illustrates streaming high resolution audio to another computer on your LAN, and using the other computer for playing the audio through its audio output.

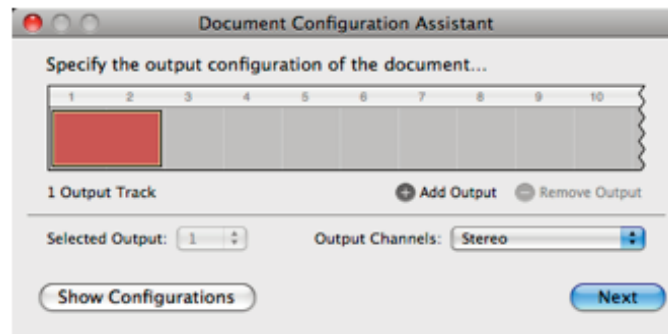
You will first need to obtain and install a certain free software application, produced by Apple, called AU Lab. (Apple does not authorize third parties to distribute this application, so it must be obtained as described below.)

The AU Lab application is part of Apple's XCode Developer Tools package.

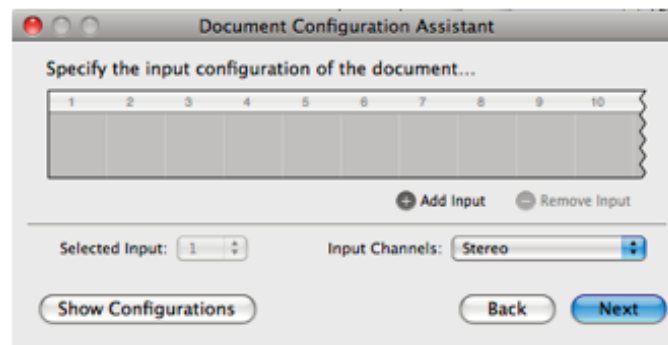
Apple's XCode Developer Tools are available on the Mac OS X System Software Installation DVD that came with your computer. They also can be downloaded (but it's quite a large download) after registering for a free developer account at Apple's website (<http://developer.apple.com>).

After installing XCode, the AU Lab application can be found in /Developer/Applications/Audio. The AU Lab application needs to be located on / copied to the receiving computer.

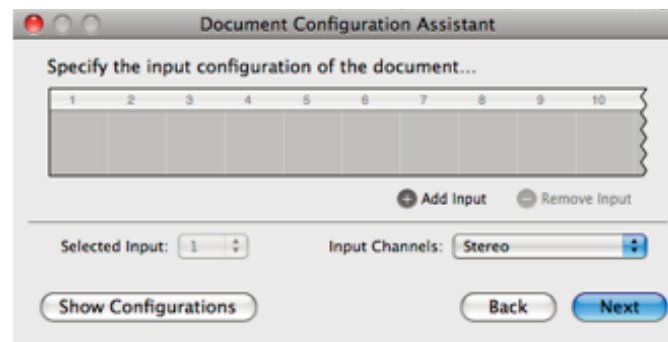
1. With audio playing in Pure Music on the sending computer, click the NetSend button.
2. Launch AU Lab on the receiving computer. (In this example, the Macintosh built-in audio only has output capability, but it doesn't matter, because only the output will be used).



3. Click Next in the "Specify the output configuration" dialog (note: this procedure will be slightly different for AU Lab versions prior to 2.0).



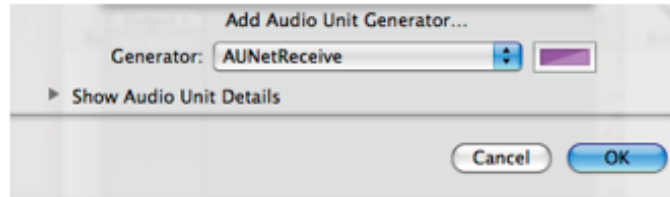
4. Click Next in the following dialog (Specify the input configuration) without adding an input (the audio input won't be used on the receiving computer).



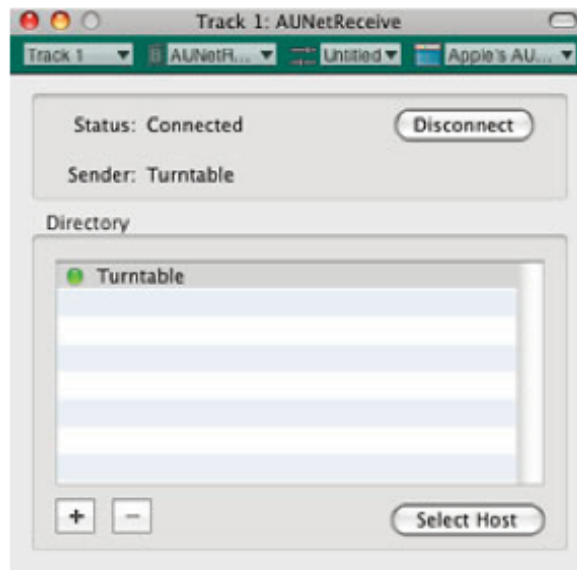
5. In the subsequent dialog box, select the audio interface you want to use for playback.



The sample rate of the audio interface on the receiving computer doesn't have to be the same as the sending computer. However, the CPU usage will be lowest if it's set to the same sample rate or an integer multiple of the sample rate of the sending computer.



6. Select Add Audio Unit Generator... from the Edit menu. Select AUNetReceive from the pop-up menu. Click Ok.



7. The AUNetReceive panel should show Connected as status, as shown. If not, click on Turntable (the default stream name in Pure Music's NetSend setup panel) in the list, then Select Host, and Connect.



If nothing appears in the AUNetReceive list, and you're sure that your LAN is functioning properly, check the firewall settings on your computer (Apple System Preferences, Sharing control panel). **In the Firewall tab, AUNetSend (or the port used in the NetSend panel of Pure Music) should be checked in the list on both the sending and receiving computers.**

The meters in the AU Generator "channel strip" panel (as shown at right) should be responding to the audio, and streaming audio playing from the audio interface of the receiving computer.

Save the settings of AU Lab on the receiving computer, so that all of the above configuration steps can be performed automatically simply by opening the saved settings document.

Notes on using Network Audio:

1. Audio can be routed to multiple output channels at the receiving computer, if a multichannel (> 2 output channels) interface is connected. Just select additional outputs in Step 5 above.

These other stereo channels can be individually delayed (by adding an Apple AUDelay Audio Unit

Effect, etc. to the channel strip on the receiving computer) for acoustic synchronization and time alignment (removing “echo” due to audio travel time in air).

2. To reset the network latency (delay), click on the Audio Engine Running icon twice to stop and restart the network stream. The network latency will increase to keep the audio stream connected if something uses a lot of CPU, so it's best to minimize activities (including use of the network) on the sending and receiving computers. *Restarting the audio engine, as described here, may be necessary if the latency increases, which will cause a noticeable delay in the output from the remote computer.*

The minimum latency (delay) for the audio output on the receiving computer depends on the sending computer's sample rate (shorter latency for faster sample rate). Latency for a hard-wired Gigabit or 100 BT connection should be on the order of only 10 to 20 milliseconds.

AU Lab, and the receiving computer's AU Lab configuration document, may be left open and running continuously on the receiving computer. The audio output on the receiving computer will be silent when the sending computer isn't streaming audio (except for audio being played locally on the receiving computer, of course). Then, when you want to send streaming audio to the receiving computer, just click the NetSend button in the main window of Pure Music on the sending computer, to reconnect and resume streaming audio (if things are configured as explained above, the connection will occur automatically without further intervention).

With a wired and fast Ethernet connection, the network latency will be repeatable, so that any audio delay compensation used to eliminate echoes (as explained below) will work without further intervention, after initially being configured. This is very useful if you have a computer on your LAN, used as a continuously running file server, which now can double as a high-definition streaming audio receiver and player.

Time Alignment of Streamed Audio

Add AudioUnit Delay plug-ins to the channel strip on the receiving computer for time alignment when audio is being played from two separate audio systems to curtail the effect of acoustical “crosstalk” between the systems. This works best when the primary system is audible from the location of the remote system, and the systems are separated by more than about 30 feet.

For example, if the main system is playing at a loud volume, and the quieter, remote system is used for background audio, the audio propagation delay (sound travels at about 1100 feet per second in air) causes a distinct echoing of the audio when music playing from the primary loudspeakers and the remote loudspeakers is heard together (and assuming the remote loudspeakers aren't playing loud enough to be heard very far away, such as if they are located in a separate room). This echoing can be distracting to normal conversation at the remote location.

Introducing a compensating signal delay using the Delay control in Pure Music's Net tab snaps the sound into focus. The bass from the in the primary system (especially if using a subwoofer) also will seem to emanate from smaller remote speakers, which may lack deep bass capability. *The overall improvement in sound quality is quite dramatic.*



As shown above, AU Delay plug-ins were added as Effects to the signal buses in AuLab on the receiving computer. One signal output bus (stereo output) was delayed by 24.5 milliseconds, and the other 33.1 milliseconds.

The correct delay can be quickly determined (while playing music) by carefully listening in or near the remote location, and increasing or decreasing the delay to eliminate the echo. A rough rule of thumb is to use one millisecond of delay per foot of separation. The correct delay used will be somewhat less, because the latency (about 10-20 ms) of a hard-wired 100 BT or Gigabit connection will add to the overall delay.

The stability (and predictability) of the Ethernet latency, and successfully using the delay feature, depends on having a hard-wired connection. The latency of a WiFi connection may not permit precise delay compensation (because the delay latency of the streaming connection can be variable).

Multiple sets of speakers connected to multiple output channels on the remote computer also may each have different delays (and EQ) incorporated into their respective Effect buses in AU Lab, by adding Delay (or EQ) AudioUnits to the Effects pop-up menus in the channel strip for each bus.