

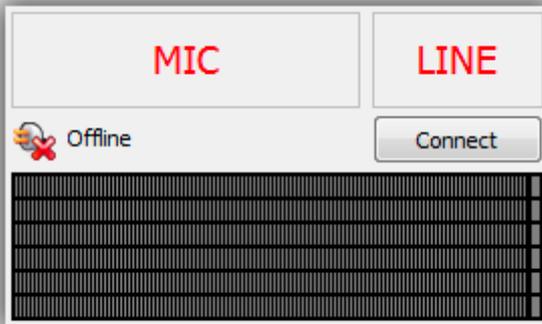
MP3 Format, LAME encoder

If you intend to stream in MP3 format, you need a copy of the LAME encoder (lame.exe). As the MP3 format is patented, we are not allowed to ship any MP3 enabled technology, and we cannot give advise about the legal requirements to use LAME or the MP3 format in your country.

To enable MP3 stream/file create in Aircast, find a copy of LAME on the Internet, and copy lame.exe (only that file, no other DLLs or so) to the Aircast program folder alongside Aircast.exe.

Encoder Status Screen Object

To manage the encoder connections, see audio levels etc. during runtime, you must add the *Encoder Status* screen object to the Aircast playout window:



To add the window, open the configuration app, go to *GUI → Screen Objects*, click *Add* and then *Encoder Status*.

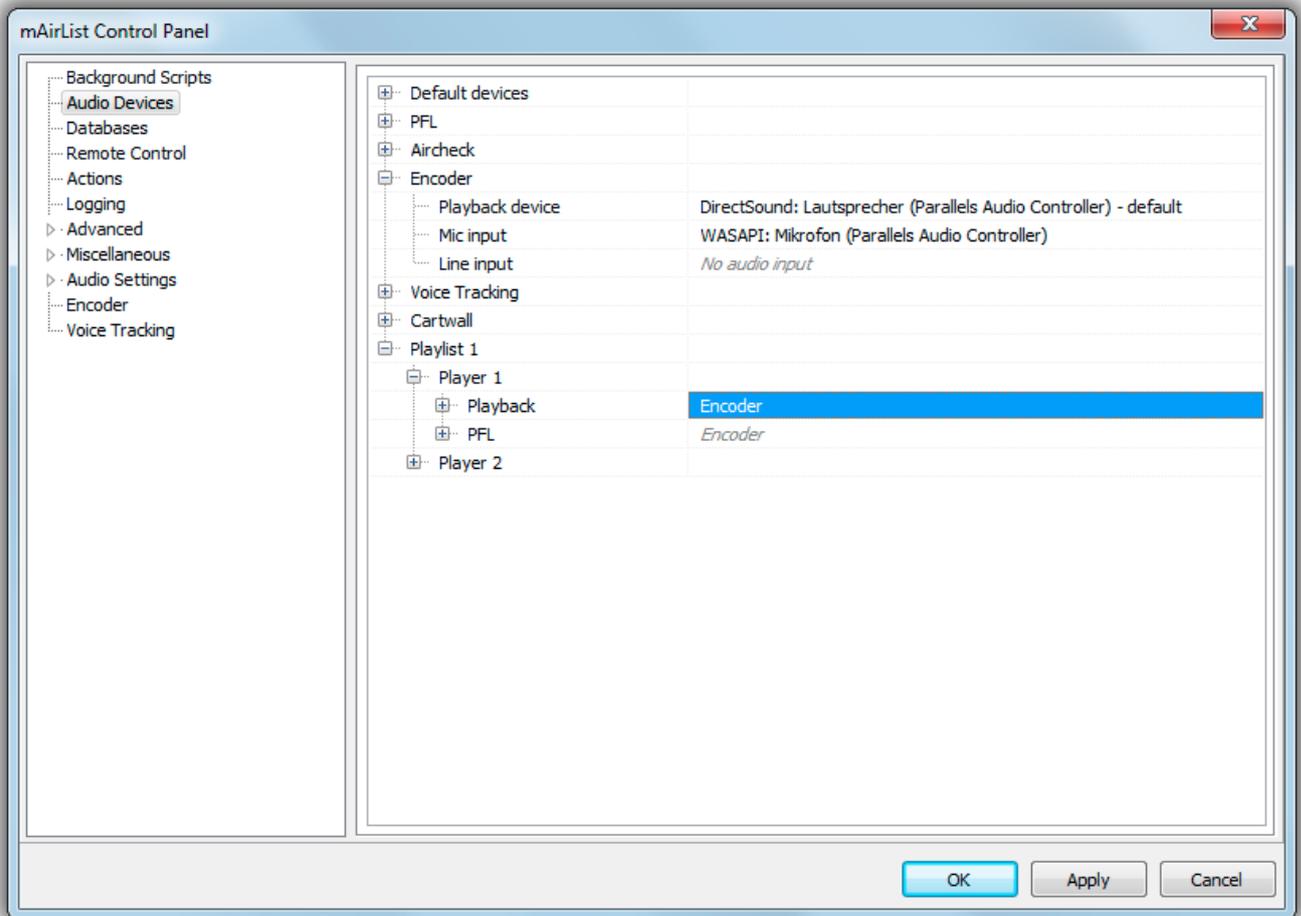
The status window contains the following elements:

- A MIC button to enable/disable the Mic input
- A LINE button to enable/disable the Line input
- The status indicator (offline/connecting/connected/number of listeners) - double-clicking this label will also bring you directly to the encoder configuration in the Aircast Control Panel
- The Connect/Disconnect button
- Audio levels (usually two bars for a stereo signal - in the above screenshot, regionalization with two extra regions is active, so it is six bars in total)

Audio Routing

Audio can be routed freely to and from the encoder, but there are two common setups described here: Internal audio routing (no mixing console used) and external audio routing (recording master signal from external mixer).

All audio settings described here refer to the *Audio Devices* page in the configuration app or Aircast Control Panel:



Internal Audio Routing

In this setup, the audio of all players is routed directly into the encoder, so it stays “inside” Aircast rather than being sent to a soundcard. Optionally, you can listen to the encoder master signal on any soundcard. You can also use the MIC button to toggle the microphone input when you want to make an announcement.

Internal audio routing can also be used on server hardware with no physical soundcard. All audio stays inside the application, no virtual soundcard etc. is required.

Steps to follow:

- Set the Playback device for all players (*Playlist 1* → *Player 1* etc.) and the cartwall to “Encoder”
- Optionally, as a precaution: Set the PFL device for all players and the cartwall to any other soundcard or “no audio output” so that it won't be broadcasted on your stream.

- Optionally: Set *Encoder* → *Mic input* to the soundcard input where your microphone is connected
- Optionally: Set *Encoder* → *Playback device* to the soundcard on which you would like to listen to your broadcast

During your broadcast, use the MIC button to turn your microphone on and off. Note that, due to the nature of digital audio on computers, there will most likely be a delay between what you speak and what you hear on the playback soundcard. With good audio cards and ASIO or WASAPI, the delay can be minimized to a level that you don't really notice. Alternatively, turn off *Filter microphone signal from local playback* in the encoder options so that you don't hear yourself at all.

The LINE button is not used in this setup, you can disable it in the encoder status screen object settings in the configuration app.

External Audio Routing

This setup is for people who own a physical mixing console and use that console to mix the on air signal rather than using the encoder built-in mixer. In essence, you connect all audio sources (Aircast players and cartwall, microphones, CD players, etc.) to your mixing console, and then send the master signal back to your PC via Line In.

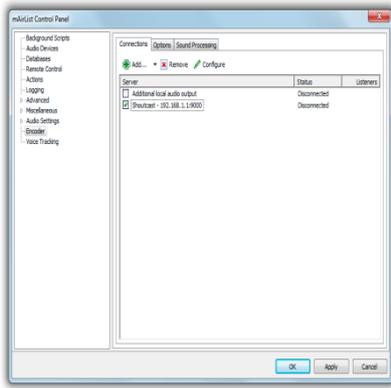
Steps to follow:

- Set the Playback device for the players (*Playlist 1* → *Player 1* etc.) and the cartwall to the soundcard output that is connected to the respective channels on your mixing console
- Set *Encoder* → *Line input* to the soundcard input where your the master signal from your console is connected
- Important: Set *Encoder* → *Playback device* to “No audio output”. If you don't, the encoder master signal is sent back to the soundcard, and there is a good chance that it will loop through your console, causing feedbacks and echoes.

Now when you start Aircast, click the LINE button in the encoder status window to enable recording from the Line input. (Note: To automate this step, go to *Actions* in the configuration app, select *After startup* in the list at the top, and add a new action of type *Encoder* → *Enable line input*.)

The MIC button is not used in this setup, you can disable it in the encoder status screen object settings in the configuration app.

Server Connections



Server connections can be set up in the *Encoder* page in the configuration app or Aircast Control Panel. From the main window, double click the status label (“connected”, “disconnected”) and you will be taken directly to the right place.

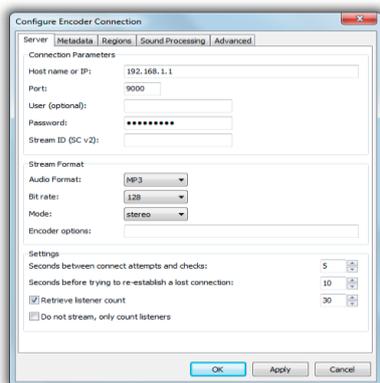
Aircast supports multiple encoder connections at the same time, in different formats. To add a new connection, click *Add* and select the desired type. You can enable/disable connections at runtime using the small checkboxes in the connection list.

In the following section we'll look at the different connection types and their parameters.

Shoutcast, Icecast

These two connection types are for streaming connections to a Shoutcast (v1 or v2) or Icecast (v2) server, the most commonly used type of streaming servers on the Internet. The setup for Icecast and Shoutcast is almost the same.

Server



On this tab, enter the **Host name or IP** and **port** of the server and your **password** - your streaming server provider will give you the required information. The **user name** is only used for Icecast in some special cases, most of the time it can be left blank. For Icecast servers, you must also specify the **mount point**. For Shoutcast v2 servers, enter the **Stream ID**.

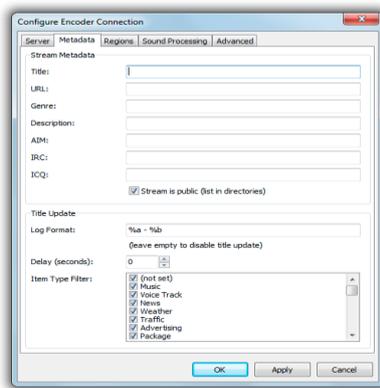
In the **stream format** section, specify the **audio format**, **bitrate** and **stereo/mono mode** of your stream. The **encoder options** you enter here will be passed to the command line

encoder (lame.exe, oggenc.exe, opusenc.exe) as additional parameters, they will allow you to fine-tune the audio settings according to the documentation of the respective encoder.

The supported **audio formats** depend on the type of server. For Shoutcast, only MP3 is supported. For Icecast, MP3, Ogg Vorbis and Opus are supported. If you want to stream in a different format not listed here, you can set the format to *Custom* and enter the MIME type in the text field that appears. Also enter the full path and parameters of the command line encoder to use in the *Encoder options* box. The command line encoder must be set up to receive raw PCM data via STDIN, and return encoded data via STDOUT.

In the **Settings** section, you can fine-tune the interval between connection attempts, reconnection attempts and listener count queries. It's also possible to skip the entire streaming and only count listeners - useful when you have a bunch of transcoding servers and want to query their listeners count to be displayed in the Encoder Status window.

Metadata



The **Stream Metadata** is the basic metadata of the stream like title, genre, URL, etc. - this is what's displayed in the public stream directories if you choose to appear there.

In the **Title Update** section, you can choose how Aircast displays the title of the current song in the stream. Use any of the usual logging variables. You can also specify a delay before the song is updated (useful to account for delays in the audio), and a filter so that e.g. only music is displayed but no jingles.

Regions

Here you can select which region signals should be sent to this particular connection. Please see the notes on Regionalization below.

Sound Processing

Connection-specific sound processing can be set up here. Please see the section on sound processing below.

Advanced

If you want to give the connection a different name in the connection list (default is the host and port), enter the custom caption here.

Integrated Streaming Server

The integrated streaming server is a Shoutcast compatible server which is built right into Aircast, so the listeners (audio players) can connect directly to Aircast rather than using a Shoutcast/Icecast server as a relay.

Note that the integrated server doesn't have as many options as a full-featured Shoutcast or Icecast, and is also not able to handle a lot of listeners. So this feature is rather meant for applications with a small number of clients, e.g. inhouse streaming.

Record to File

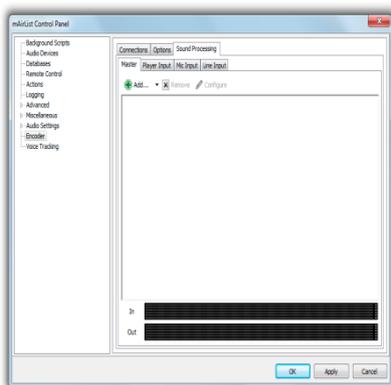
This is a virtual connection type that allows you to save the streamed audio into a (compressed) audio file rather than sending it to a server.

The output folder and file format can be adjust freely, using logging variables for the current date/time. You can also create a new file every x minutes, which is useful if you want to build a continous archive of your streamed broadcasts.

Additional Local Audio Output

Using this virtual connection type, you can route the master signal (and any regional submix thereof) to an available soundcard. This is useful if you want to listen to a particular region rather than the entire master signal.

Sound Processing (DSP)



Aircast allows the integration of internal or external (VST, Winamp) DSP plugins into the encoder signal. The DSPs can be applied at various stages of the signal flow (see the block diagram above):

- Master sum
- Player input

- Mic input
- Line input
- Signal of any particular server connection

The connection-based DSP chains are managed in the connection properties dialog, the other chains on the *Encoder* configuration page, *Sound Processing* tab.

Use the *Add* button to add a new effect to the DSP chain and set up its parameters. When you have multiple effects in the list, you can reposition them by dragging them with the mouse.

Regionalization

(work in progress)