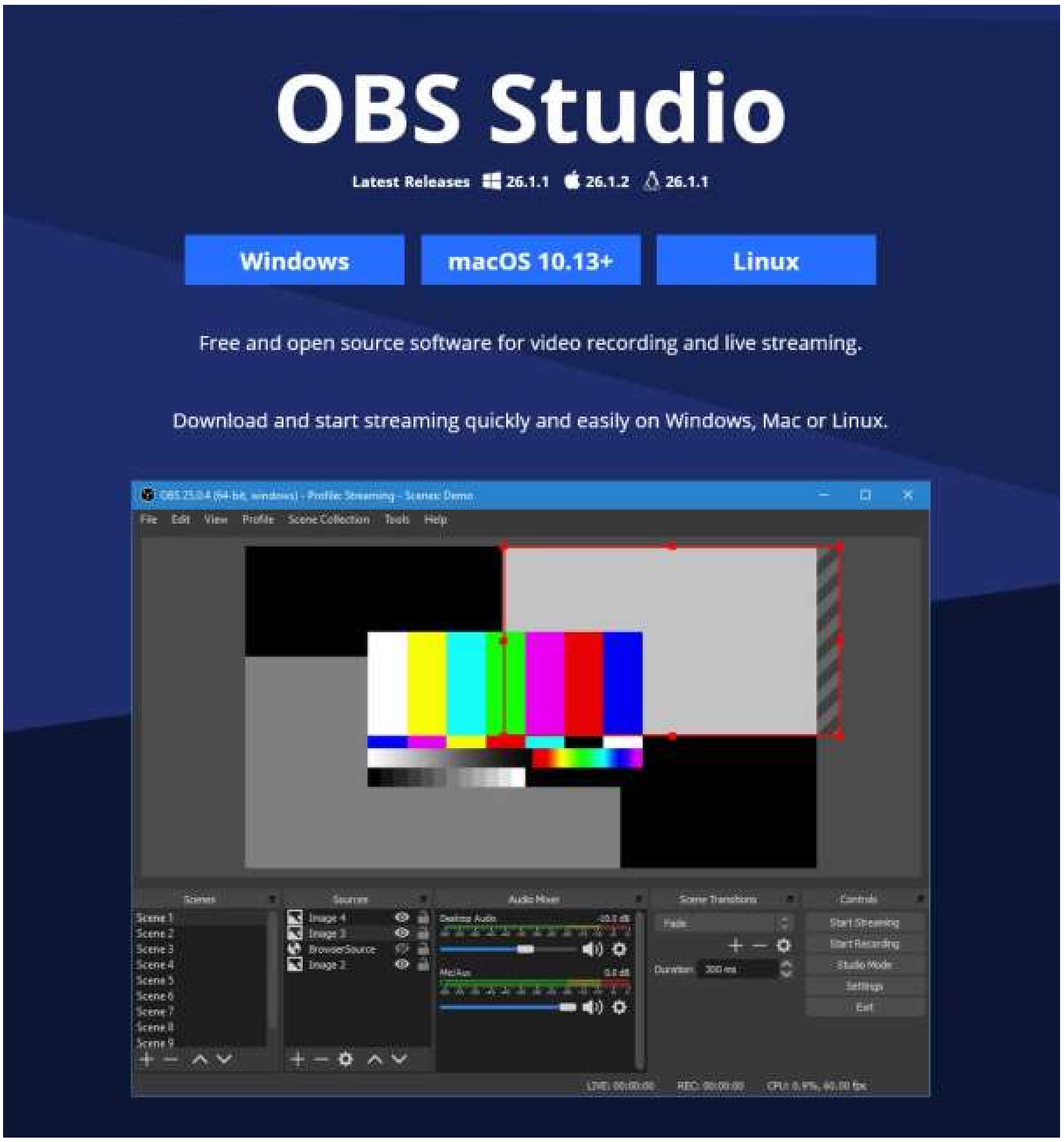
OBS Studio Reference Manual


OBS Studio
Latest Releases 26.1.1 26.1.2 26.1.1
Windows
macOS 10.13+
Linux
Free and open source software for video recording and live streaming.
Download and start streaming quickly and easily on Windows, Mac or Linux.

PDF version 1.0-0 Feb 1st, 2021.

Table of Contents

Foreword

This PDF document is an exact copy of some official OBS documentation pages located at https://obsproject.com/wiki/, gathered to ease off line reading.

The pages copy/pasted in this document are those of the chapters **System Requirements** to **Streaming with SRT Protocol**.

Other pages will be included in another PDF file to come.

A translation to French too… 😊

System Requirements

OBS Studio requires the following hardware and operating systems to function. Having a compatible system does not guarantee that it is capable of streaming or recording using OBS. The CPU requirements vary considerably depending on the chosen encoder, resolution, FPS and your scene complexity. Try the Tools -> Auto Configuration Wizard in OBS itself to find appropriate settings for your specs.

Basic Requirements

Windows:

- DirectX 10.1 compatible GPU
- Windows 8 / 8.1 / 10

macOS:

- Intel CPU (PPC is not supported)
- OpenGL 3.3 compatible GPU
- macOS High Sierra (10.13) or later

Linux/Unix

- OpenGL 3.3 compatible GPU
- X window system

Hardware Encoders

Hardware encoders are generally recommended for local recordings, but not streaming. They can be a last resort if software encoding is not possible. They offer minimal performance impact in exchange for a reduction in quality at the same bitrates as software (x264) encoding using the default preset of veryfast. Currently, all consumer-level hardware encoders are provided by certain GPUs, as listed below.

Please make sure to use the latest drivers for your GPU:

- NVIDIA
- AMD
- Intel GPU drivers are usually available from your mainboard manufacturer, but can also be found here

NVIDIA NVENC

NVIDIA GeForce GTX-class GPU with Kepler architecture (GKxx) or newer (starting with some cards of GTX 600 series).
Support for mobile GPUs can exist, but most will not support NVENC. If you get an error

when trying to use the NVENC encoder and your drivers are already up to date, your GPU very likely does not support NVENC.

Intel Quick Sync Video (QSV)

Intel HD Graphics GPU on Intel Core-i-CPU 2xxx (Sandy Bridge) or newer.
Due to low quality of early iterations of QSV, Intel Core-i-CPU 4xxx (Haswell) or newer is recommended.

QSV can be unstable on Windows 7 and may require workarounds to enable it.

AMD Advanced Media Framework (AMF)

Please read the AMF plugin documentation for compatible GPUs as well as the troubleshooting guide for issues.

OBS Studio Quickstart

Just follow these **4 steps** to start streaming or recording!

1. Run the auto-configuration wizard

When you load OBS Studio for the first time you should see the Auto-Configuration Wizard. If you're new to OBS or just want to get started as quickly as possible, follow the steps to get good starting settings for your setup. If you only see the main OBS Studio window, you can access the Auto-Config Wizard in the Tools menu at the top. This wizard will automatically test your system and attempt to find settings that your PC can handle. This includes streaming or recording, resolution, bitrate, encoder, streaming provider and more. You can always modify the settings manually later.

2. Set up your audio devices

By default, OBS Studio is set to capture your system default desktop audio device and microphone. You can verify this by looking at the volume meters in the mixer section of the main OBS Studio window, and see if they are active. If they aren't moving, or you suspect the wrong device is being captured, click on Settings -> Audio and select the devices manually.

macOS users: If you're on macOS, you'll need an extra app to capture desktop audio. This is due to limitations in macOS that provide no direct capture methods for desktop audio devices.

3. Add your sources for video

Next, you'll see that the preview is a black screen. OBS does not capture any video by default. To get started capturing, **you need to add a Source**. At the bottom of the window is a box called 'Sources'. Click on the + (or right click inside the box) and pick the source you want. As a few examples, select Game Capture if you're capturing a game, Window for non-game applications, or Video Capture Device for a webcam or capture card.

Laptop users: Here's our troubleshooting guide if your game/window/display capture sources still show a black screen.

Sources and Scenes are the bread and butter of OBS Studio, and can be super powerful. Click here to read more about them.

4. Test your stream and record settings

Double check that all your settings are how you want them in Settings -> Output. **Then, just hit Start Recording or Start Streaming.** We strongly encourage running a test for a few minutes to make sure that there are no issues, rather than just jumping in to your first stream or recording. If you run into any issues, or need further help, take a peek at our help portal.

Once you're satisfied, you can go on to creating great content. That's all there is to it!

If you want to read a more in-depth guide about the power of OBS Studio, jump into the Overview wiki.

FAQ

Where are my recordings saved?

Once your recording is done, you can find it using File -> Show Recordings. You can change this in File -> Settings -> Output -> Recording.

I need my recordings in MP4!

File -> Remux Recordings will quickly and easily convert your video files into MP4.

Help, my video is laggy!

Depending on the kind of lag, this could be related to a slow internet connection, your game using too many resources, or incorrect settings. Read the troubleshooting guides linked below.

I need help with other issues!

- Troubleshooting Guides
- Community Chat & more

OBS Studio Overview

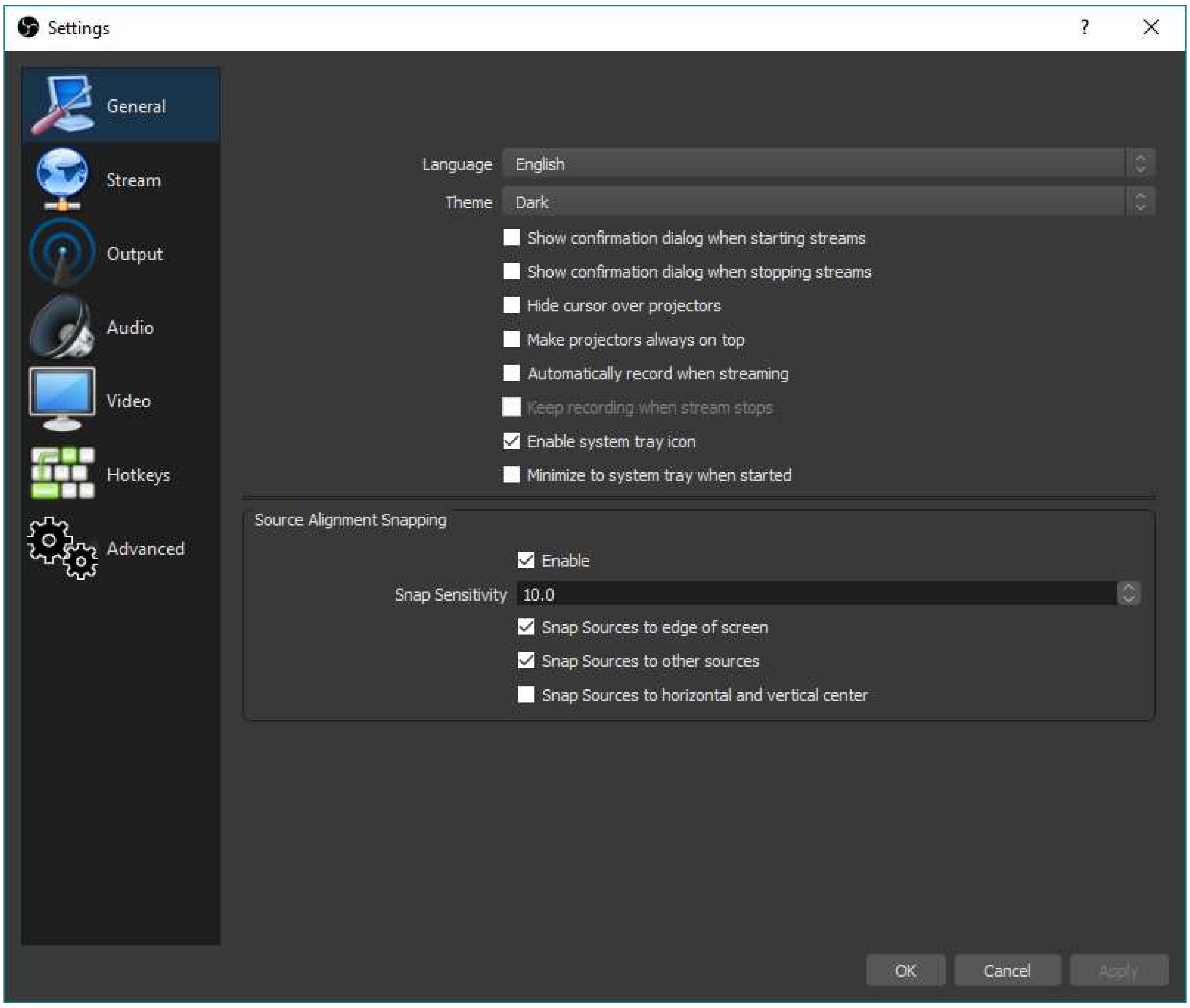
This guide assumes you already have a basic knowledge of streaming services and streaming terminology, and aims to give you a quick overview of the most important options and features in OBS Studio. However, even if you don't have any previous experience, don't be discouraged! This guide can still be very useful. It may seem like a lot to take in all at once; just take your time and read carefully.

This guide is broken into these main sections:

Quickstart

The 4-step Quickstart guide has been moved to OBS Studio Quickstart. Use it if you're brand new to OBS Studio and just want to get started as soon as possible!

Settings



Language: English
Theme: Dark
☐ Show confirmation dialog when starting streams
☐ Show confirmation dialog when stopping streams
☐ Hide cursor over projectors
☐ Make projectors always on top
☐ Automatically record when streaming
☐ Keep recording when stream stops
☑ Enable system tray icon
☐ Minimize to system tray when started

Source Alignment Snapping
☑ Enable
Snap Sensitivity: 10.0
☑ Snap Sources to edge of screen
☑ Snap Sources to other sources
☐ Snap Sources to horizontal and vertical center

Overall, most settings will be fairly self-explanatory. This will not cover everything, but will be an overview of the basics and what you can do in each section.

General

- Change theme (light/dark)
- Enable/disable system tray icon (required for hiding to tray)
- Show confirmation for starting/stopping streams
- Enable/Disable Source snapping
- Enable/Disable automatic recording when streaming

Stream

- Select one of the included Services (Check "Show all services" to see the full list in the dropdown) or set Custom Streaming Server
- Select the server (or enter its URL for Custom Streaming Servers)
- Enter your Streamkey

Output

Note: This section covers Simple output mode. Advanced output mode gives you further options to configure your Stream and Recording independently if you want to. We will not be covering advanced options in this guide (see links at the end). Generally, unless you know why you need to change it, the default options are recommended.

For Streaming:

- Set your Video Bitrate. This changes according to your upload speed and the limitations of the service you use (for example, 6000kbps max for Twitch.tv)
- Set your Audio Bitrate. Around 160kbps for streaming, or lower if you have a low upload speed

For Recording:

- Set your Recording Path
- Record using a preset (Same as stream, High Quality, Indistinguishable Quality, and Lossless) and a different Encoder (if available)
- Select a Hardware Encoder if available (only if you use a different preset than "same as stream" and if a Hardware Encoder is available)[QuickSync, NVENC or AMD VCE]
    - Encoder option will not show up until you select a recording quality other than Same as stream
- Enable the Replay Buffer (hotkey to save the buffer must be set under the Hotkeys section)

Audio

- Change the Sampling Rate
- Select up to two Desktop Audio Devices
- Select up to three Microphone/Auxiliary Devices
- Enable Push-to-mute or Push-to-talk for each device (keys configured in the Hotkeys section)

Video

- Base (Canvas) Resolution
    - This is the amount of space you have to fit your sources. You usually want this to match your monitor display resolution, or if you are playing games, the game resolution. Default is your primary monitor.
- Output (Scaled) Resolution should be the Stream Output Resolution (720p, 480p, etc)
    - This is the resolution that the stream/recording will output at, using the selected downscale filter. If you have 1080p sources, but want to stream at 720p, this is where you would set that.
- Common FPS Value should match your desired output FPS (30/60 for example)
    - Note that 60-fps streaming can be very taxing on your system compared to 30-fps. Test ahead of time and ensure your system has enough resources available.

Hotkeys

Here you can set hotkeys that do a variety of things. Some examples:

- Start/Stop Streaming/Recording
- Hide/Show Sources
- Switch to a specific Scene
- Push-to-talk/Push-to-mute
- Capture active window for Game Capture
- Start/Stop/Save Replay Buffer

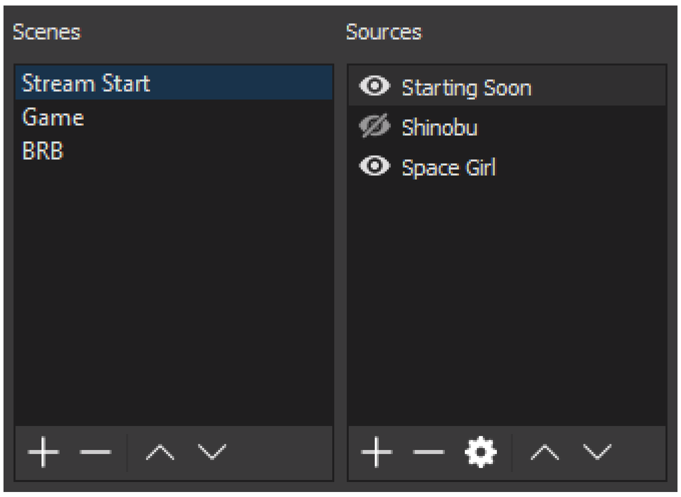
Keyboard keys are supported. To use joysticks as hotkeys on Linux and Windows, you can use antimicro to map the joystick keys to common keyboard keys and then use them in OBS.

Advanced

Most of these settings should not be changed unless you understand exactly why you need to change them. However, some settings can be useful for new streamers, such as:

- Change the Filename Formatting (or use folders: %CCYY\%MM\%DD\%hh-%mm-%ss = 2016\07\10\12-35-25.flv)
    - Hover over the field for a popup that explains each available variable for naming
- Activate Stream Delay
- Configure Automatic Reconnect
- Do not touch anything else in Advanced unless you absolutely know what you're doing. Really, we mean it. If you have questions, search around or hop in the support chat and ask!

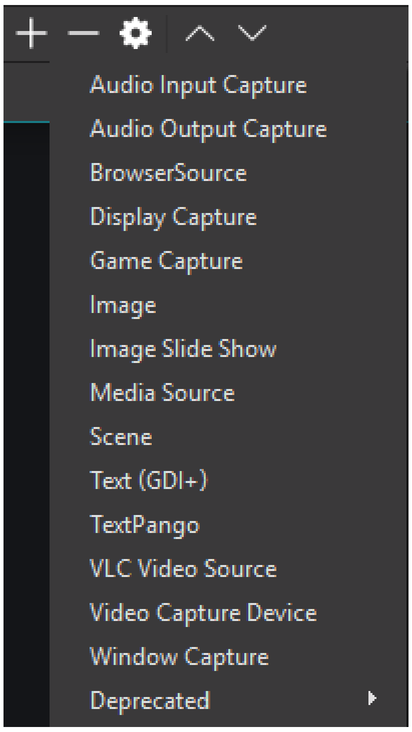
Scenes and Sources



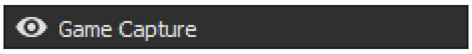
Scenes and Sources are the meat of OBS Studio. These are where you set up your stream layout, add your games, webcams, and any other devices or media that you want in the output.

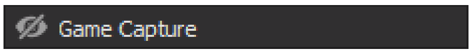
Right click in the box under Scenes (or use the plus at the bottom) to add a scene if there are none listed yet. You can create as many Scenes as you want, and name them to easily distinguish between them. For example: Welcome, Desktop, Game, Break, End. The arrow buttons can be used to change the order. As an important note, all Scenes and Sources are global in OBS Studio, so they can not share a name. This means if you name a source Game, you can't have a Scene with the name Game.

Once you have created a Scene, right click in the Sources box (or use the plus at the bottom) to add what ever you want to capture. Whether it's a specific window, a capture card or game, image, text or your entire display that you want to capture, there are several different sources available in OBS Studio for you to choose from. Try them out!

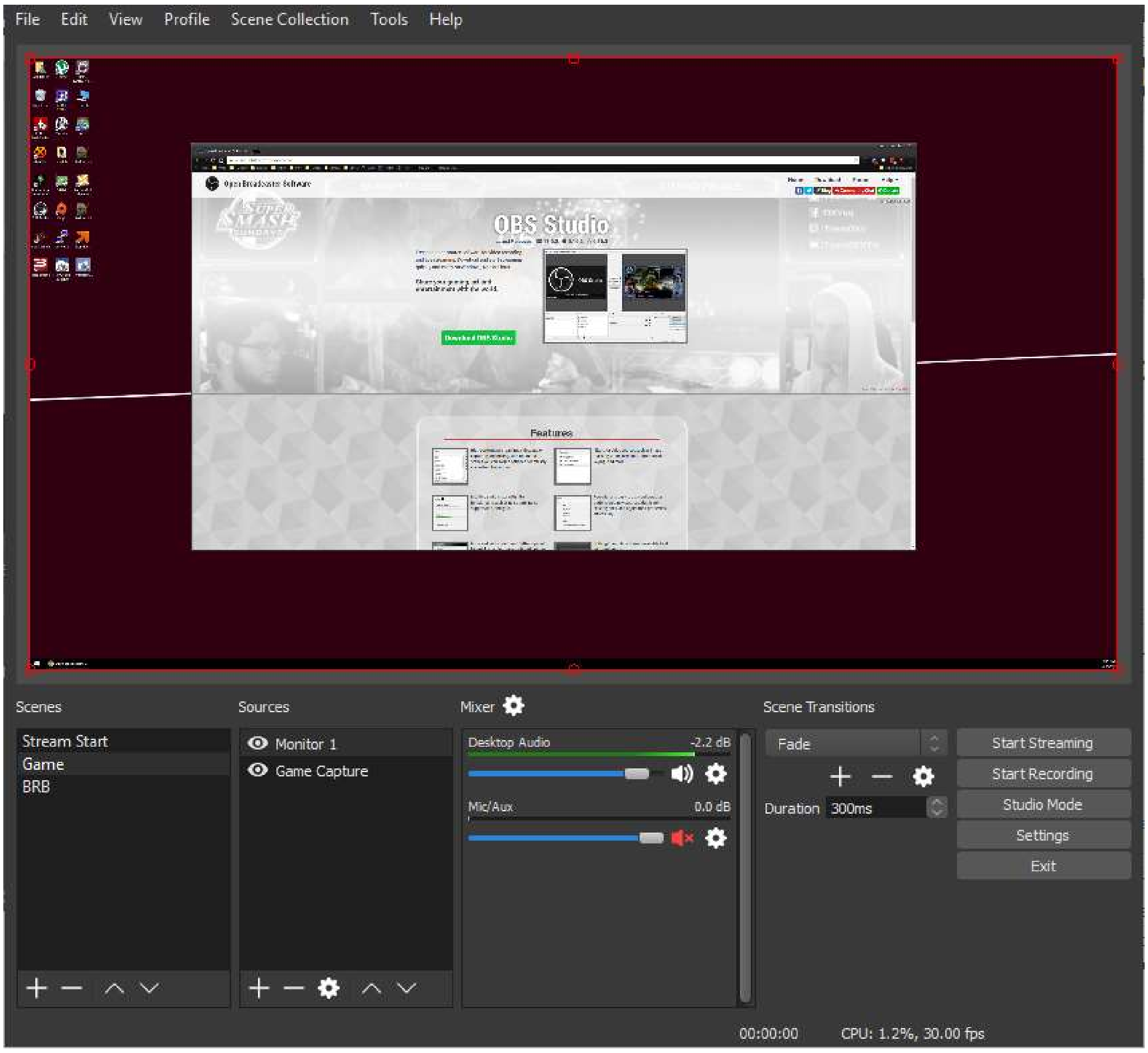


You can re-align sources in the preview and change their order by using drag and drop in the list, or using the up and down arrow buttons. A Source that is listed above another Source in the list will be on top and might hide what's beneath it. This can also be useful for situations where you want something on top of another source, like a webcam to show over your game play. Any time you see an eye icon, you can click it to show or hide the associated item with it (this applies to filters as well)

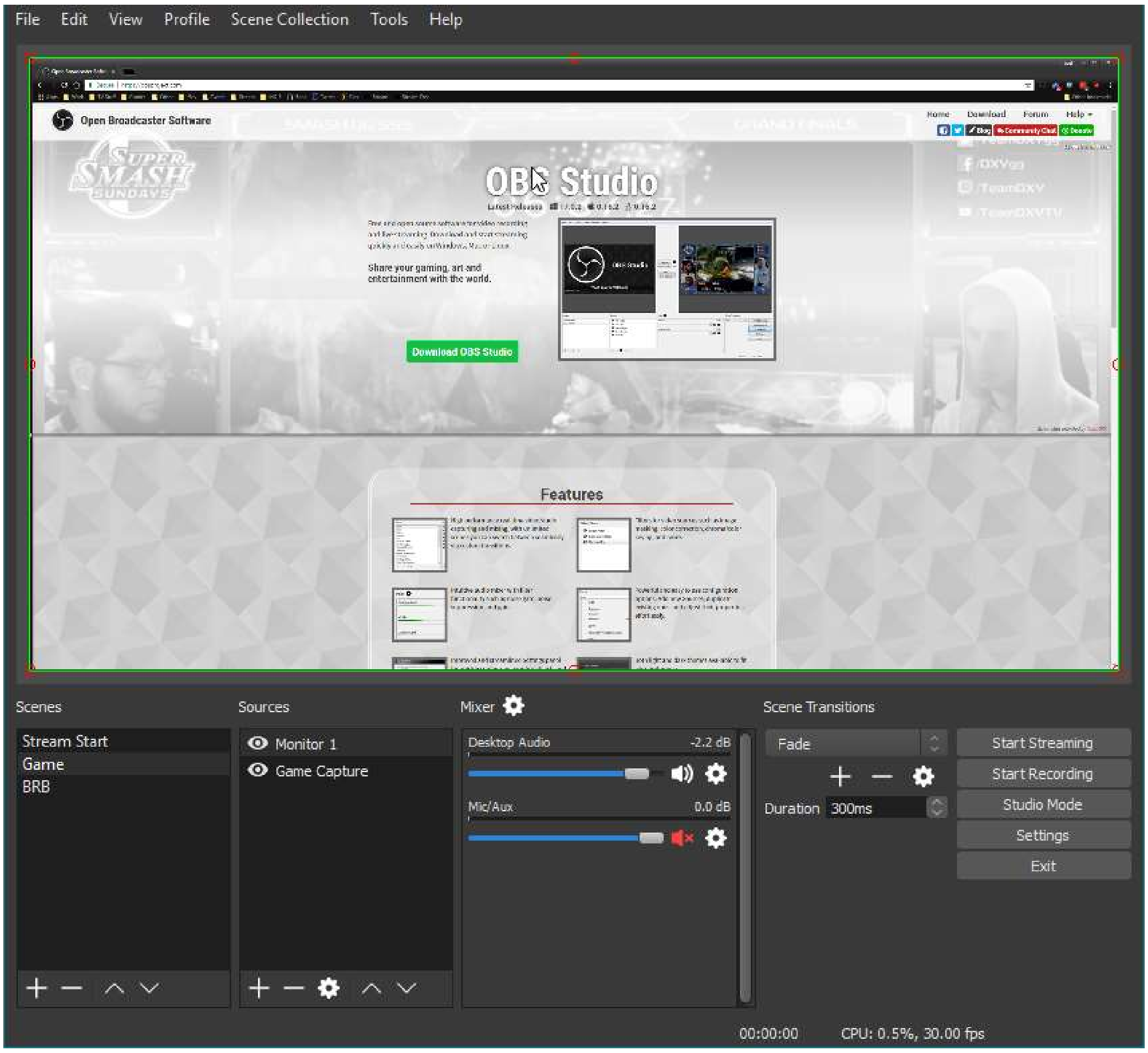
Visible: 

Hidden: 

When a Source is selected in the Sources list, you will see a red box that shows up around it. This is the bounding box, and can be used to position sources within the preview as well as make the source larger or smaller.



If you need to crop a source, hold the Alt key and drag the bounding box. The edges will change to green to show it's being cropped. You can see both techniques being used here to crop and enlarge only the part of the screen we want to show:

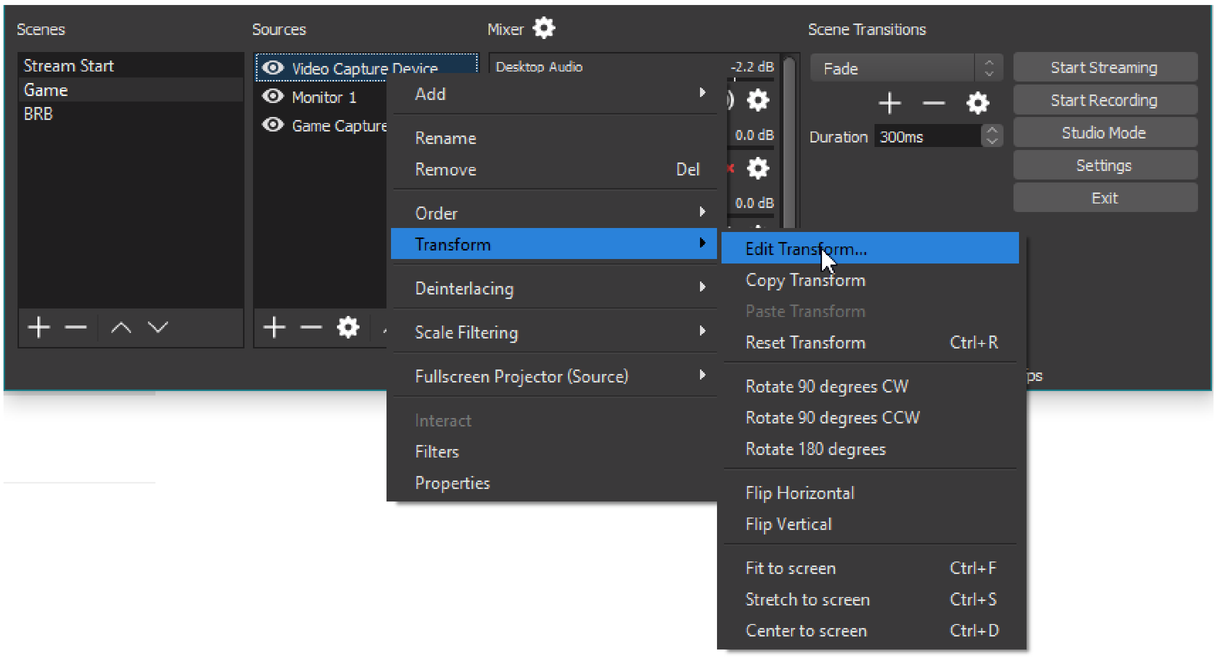


If you later on change the *Base (Canvas) Resolution* of OBS Studio, you will have to re-align or re-size the sources. Changing the *Output (Scaled) Resolution* does not have this effect.

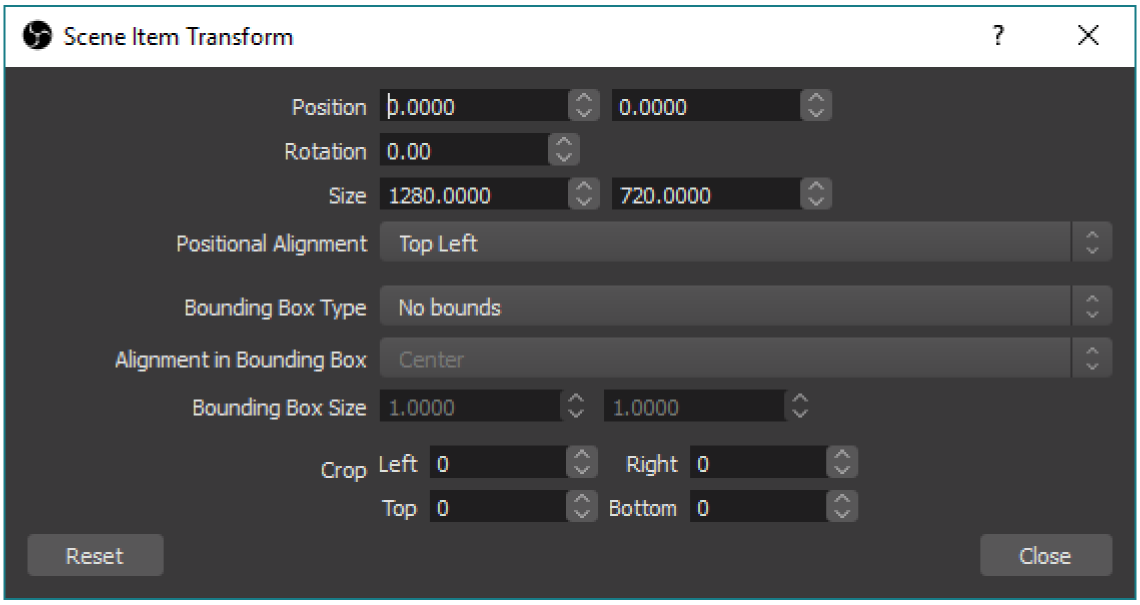
The following Hotkeys are available in the preview to tweak the source position and size:

- Hold CTRL to disable Source/Edge snapping
- Hold ALT to enable cropping
- CTRL+F for fit to Screen
- CTRL+S for stretch to Screen
- CTRL+D for center to Screen
- CTRL+R to reset a source size/position

You can also right-click each source in the list to access further options. This is also how you access filters, which is discussed in the very next section!



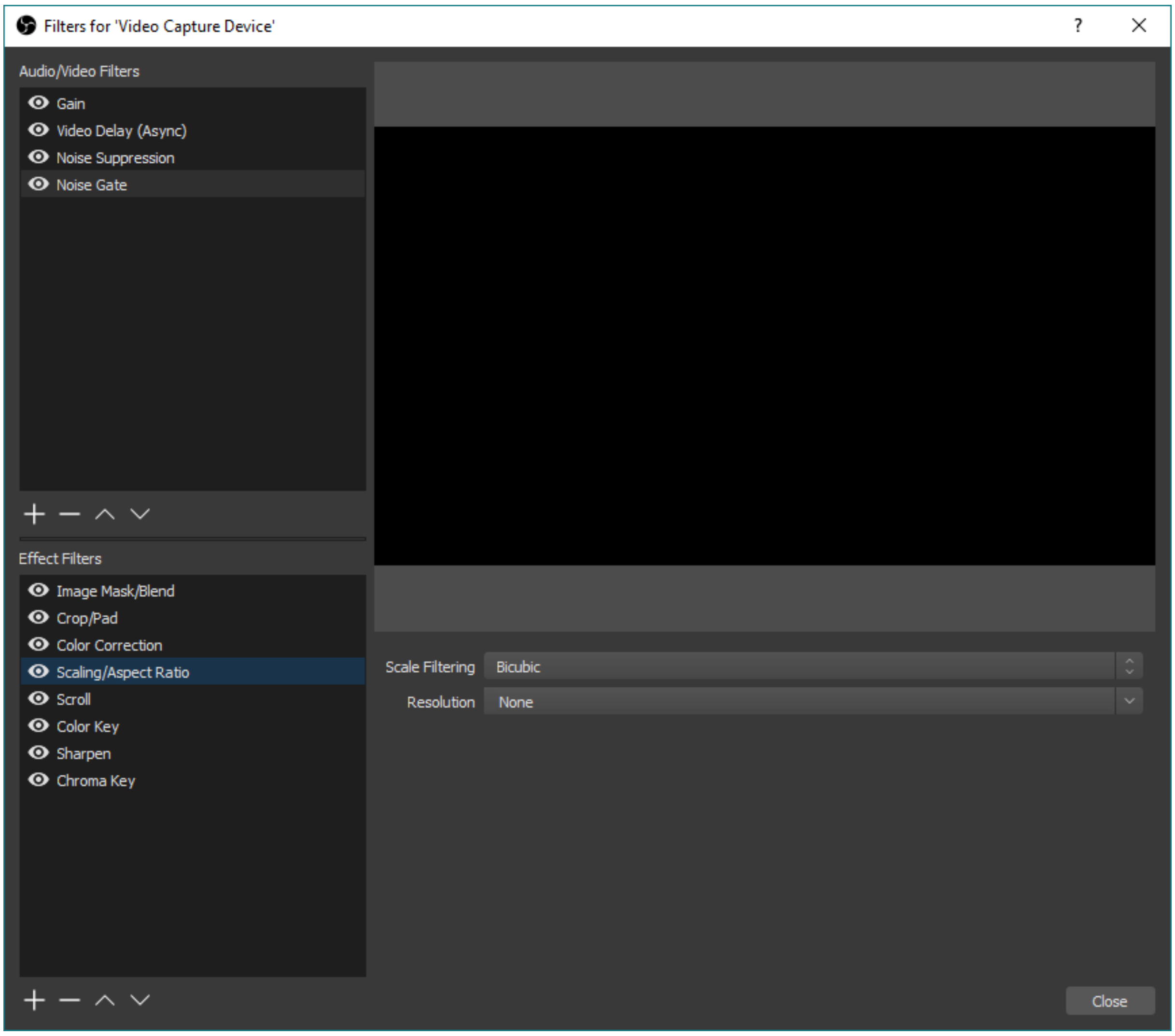
Edit Transform menu:


Scene Item Transform

Position	0.0000	0.0000
Rotation	0.00	
Size	1280.0000	720.0000
Positional Alignment	Top Left	
Bounding Box Type	No bounds	
Alignment in Bounding Box	Center	
Bounding Box Size	1.0000	1.0000

Crop: Left 0, Right 0, Top 0, Bottom 0

Filters



Scale Filtering Bicubic
Resolution None

Filters can be added to each Source/Audio Device, and even to a Scene. You add a filter by right-clicking onto the desired Source or Scene in the list, and then selecting Filters. For Audio devices, click the little cogwheel next to the volume bar in the Audio Mixer and select Filters. Clicking the eye icon next to an added filter will enable/disable that filter, similar to showing/hiding a source. See the Filters Guide link for a more in-depth guide on Filters and their specific function.

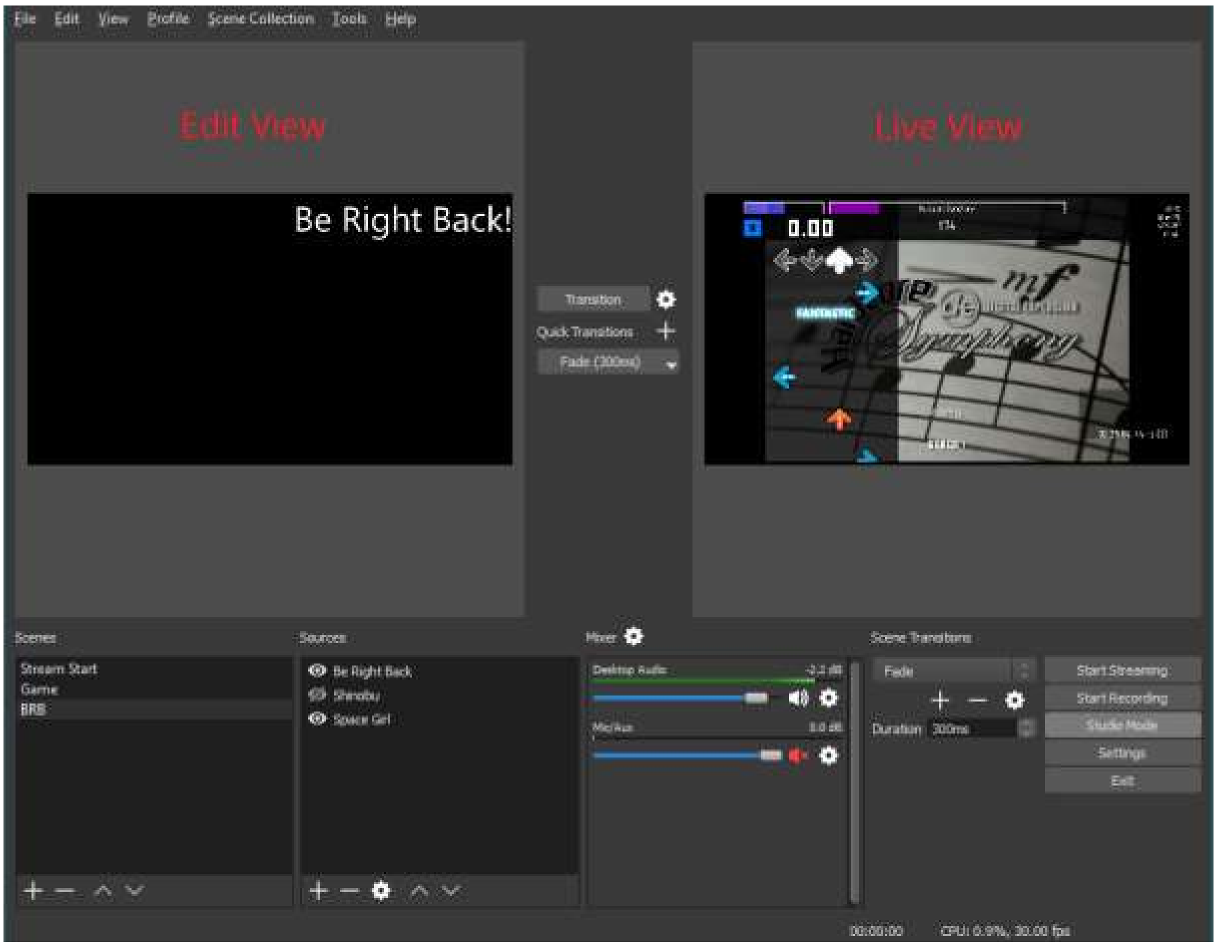
List of Effect Filters:

- Image Mask/Blend
- Crop/Pad
- Color Correction
- Scaling/Aspect Ratio
- Scroll
- Color Key
- Sharpen
- Chroma Key

List of Audio/Video Filters:

- Gain
- Video Delay (not available for Window, Display, or Game Capture)
- Noise Suppression
- Noise Gate

Studio Mode



Studio mode can be a bit confusing. First, let's understand what the purpose of Studio mode is.

Activating Studio Mode allows you to change your Scenes in the background without your viewers being able to see you making those changes. After you click on the Studio Mode button, you will see the current Live Scene (what your viewers see) on the right while your edit Scene on the left.

After you are done editing the Scene you can click on "Transition" (or use a Quick Transition/Hotkey if you added one) to swap the left and right, making the Scene you were editing the live Scene. If you are changing Scenes, the last active Scene will be shown in the edit area on the left. After you are done with everything and transitioned to the changed Scene, you can deactivate Studio Mode until you need to edit again. Viewers cannot see when Studio Mode is enabled or not.

Scene Transitions

Some Transitions in OBS Studio are available from the first launch: Fade and Cut Transition. You can add more using the plus button in the Scene Transitions section. Currently Swipe, Slide, Fade to color, and Luma Wipe (with several wipe options) are available. More may come in the future.

You can switch the currently active Transition with the drop-down menu and change its duration.

After you have configured a Transition, you can also add it as a Quick Transition in Studio Mode. For Quick Transitions, you can configure the duration (length) of the transition as well. For example, you can add the same Fade to your Quick Transitions Menu with 500ms and 1000ms if you wanted. Also, you could configure a Swipe left and a Swipe right transition, then add them both as Quick Transitions and use them to first swipe left in your Scene A to change Scene B, and then swipe right back to your newly configured Scene B when you are done.

Testing

There are many more options and functions inside OBS Studio, but I hope you got a good idea of the different things that are possible with this great tool.

Test, test, test!

Now it's time to start testing! Are the settings working? Is the stream running smooth? Are all your Scenes configured how you want them? It's strongly recommended that you test everything as best as you can before starting your first live stream. This can help work out any kinks or performance issues you might be having ahead of time, and save the embarrassment of any issues on the stream itself.

If you have problems, start a thread in the Support section of the forums. Be sure to include a log file. Alternately, you can join our community chat to get live help from community volunteers.

If you want more control over your Recording settings or want to record multiple audio tracks, switch the Output Mode to Advanced and check out this guide: High quality recording and multiple Audio Tracks

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Resource Links

- Advanced local recording guide
- Video guide for OBS Studio
- Buffering issues
- CPU overload/High Encoding issues

- Dropped Frame issues
- Laptop troubleshooting
- In-depth filters guide
- Enabled CoreAudio encoder for better audio quality

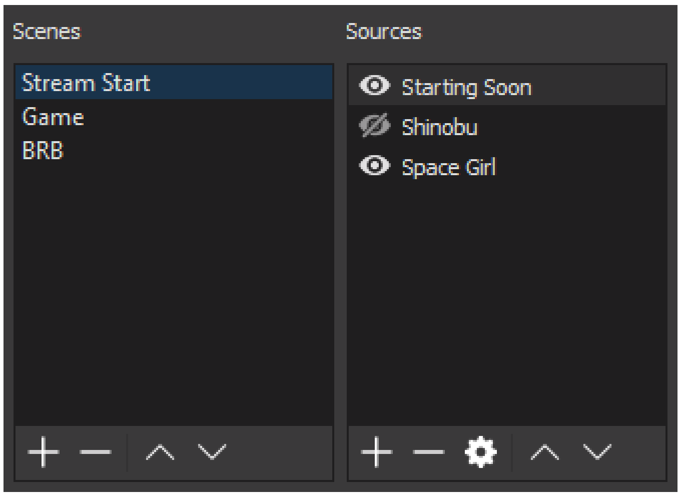
`Original guide created by Jack0r, edits/updates by Fenrir`

Note: This guide is a work in progress, and any updates/contributions are very welcome!

Sources Guide

Table of Contents

Scenes and Sources Overview



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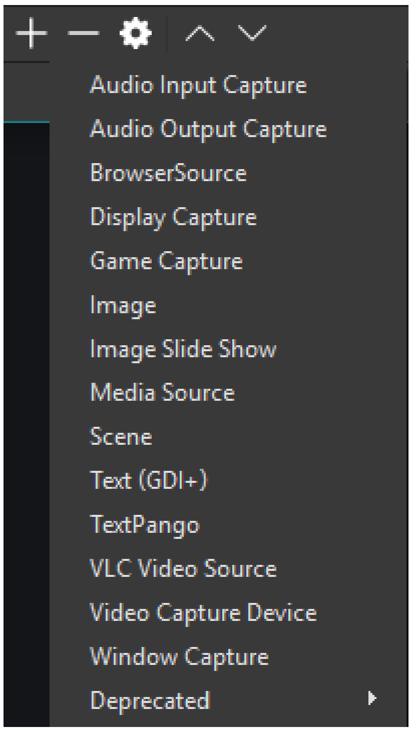

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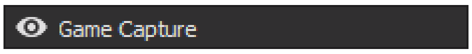
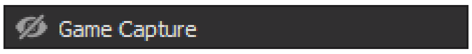
Visible: 

Hidden: 

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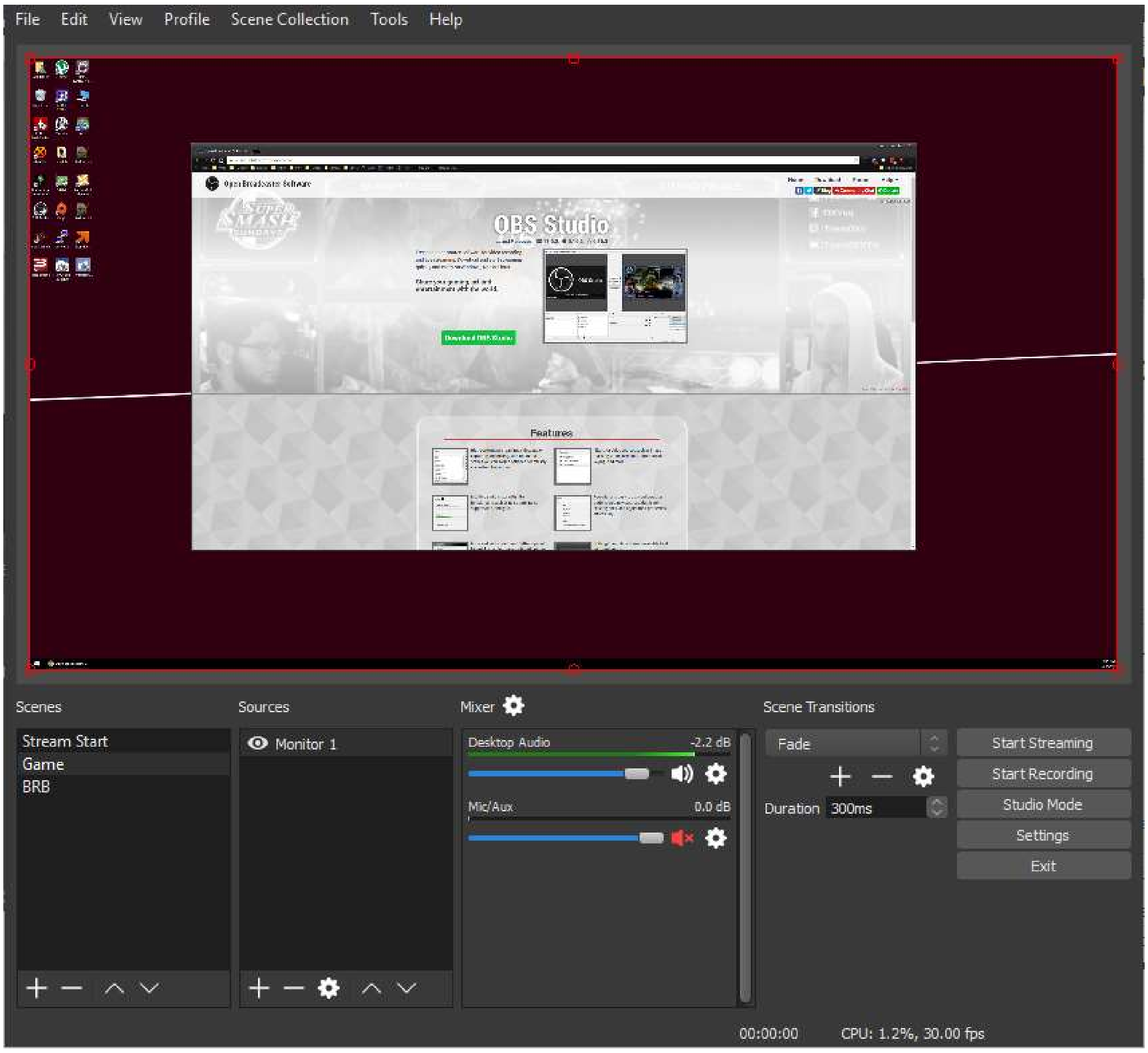

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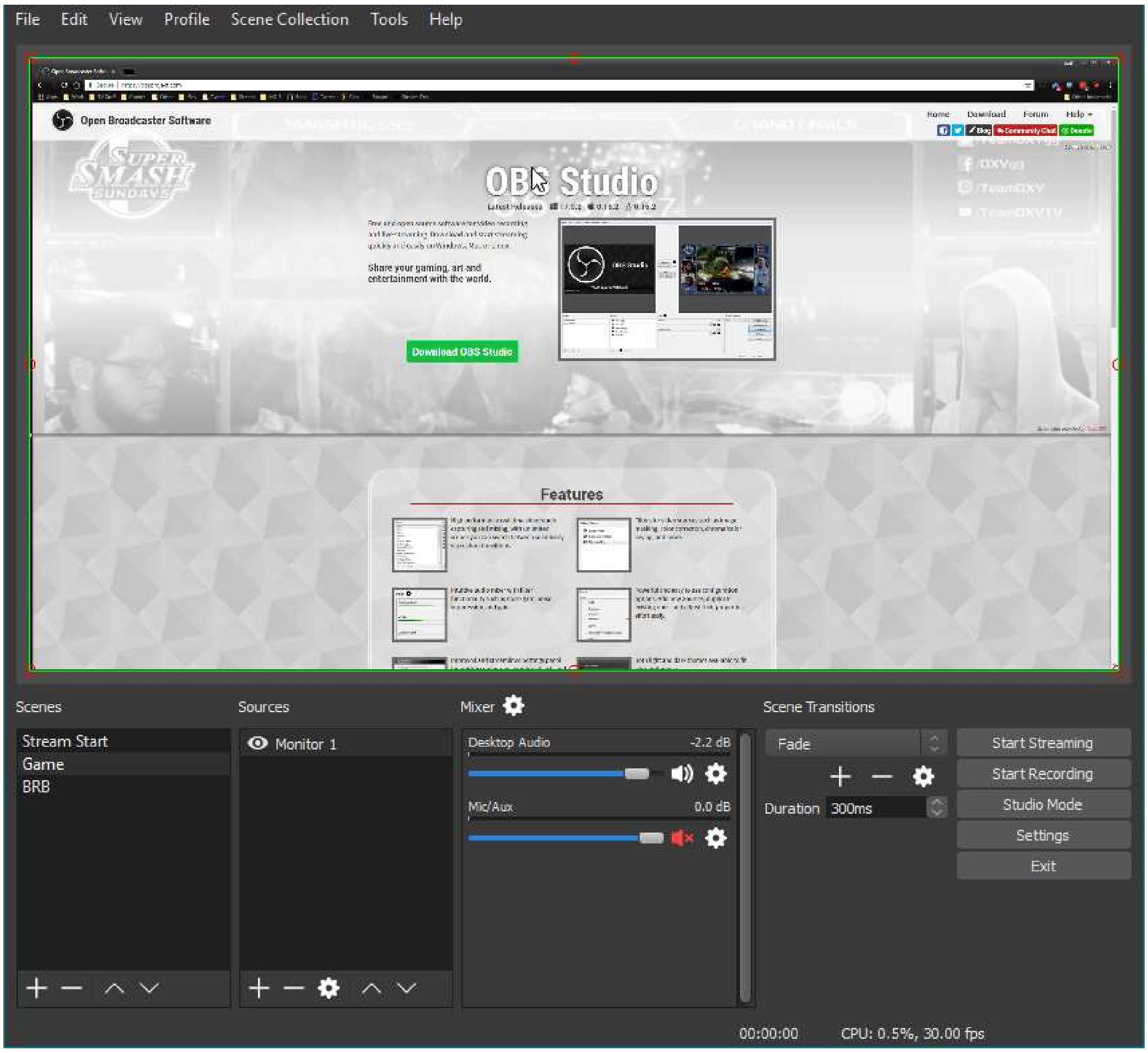

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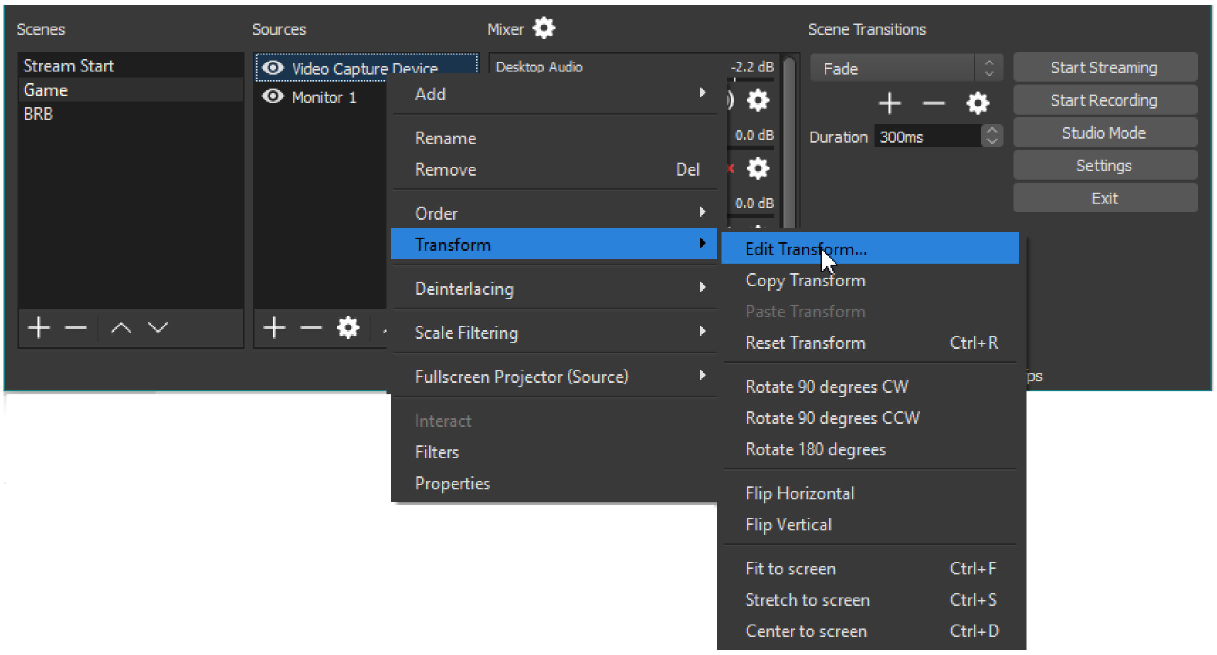
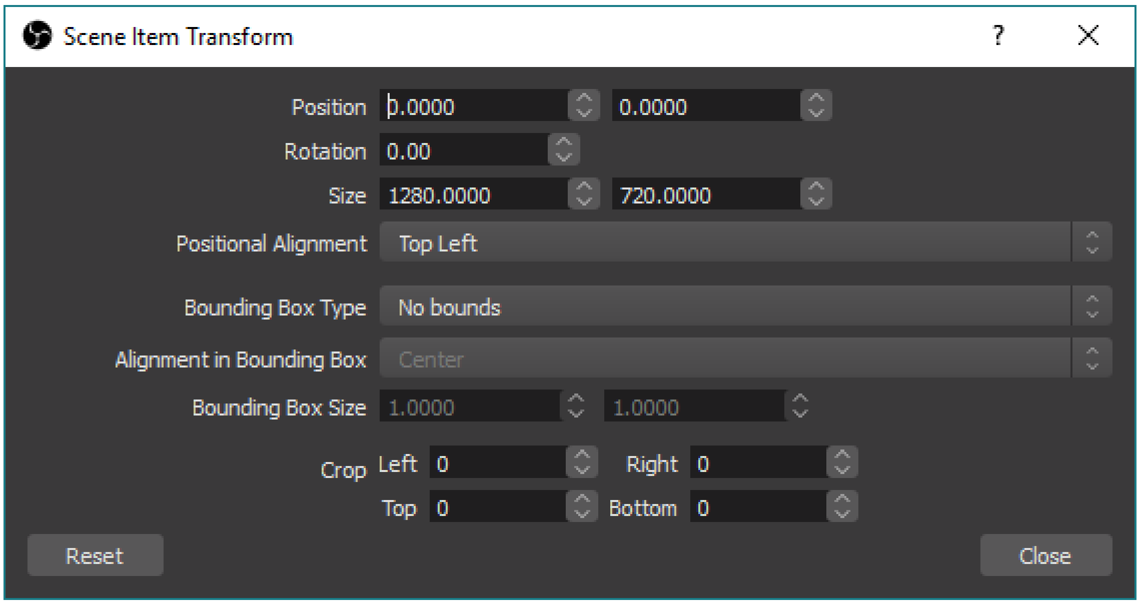
- Hold ALT + drag the bounding box to crop
- CTRL+F for fit to Screen
- CTRL+S for stretch to Screen
- CTRL+D for center to Screen
- CTRL+R to reset a source size/position

You can also right-click each source in the list to access further options. This is where you access the Filters sub-menu, which is discussed in detail in our Filters Guide.



Edit Transform menu:


Scene Item Transform

Position	0.0000	0.0000
Rotation	0.00	
Size	1280.0000	720.0000
Positional Alignment	Top Left	
Bounding Box Type	No bounds	
Alignment in Bounding Box	Center	
Bounding Box Size	1.0000	1.0000
Crop Left	0	Right 0
Top	0	Bottom 0

Source Descriptions

Audio Input/Output Capture

WARNING: Audio Input/Output Capture source can cause an echo effect if you have the same device selected in Settings -> Audio. If you plan on adding audio devices directly to your scenes, make sure they are disabled globally first.

This source allows you to add an audio input or output device (i.e. microphone or headset repsectively) to a specific scene. Simply pick the device you wish to capture, and the audio from that device will be captured when the source is active. These sources can be useful if you only want specific audio devices active in specific scenes, rather than globally through all of OBS.

Blackmagic Device

The Blackmagic Device source allows you to add a variety of Blackmagic Design capture cards, using their provided developer SDK to ensure the best possible compatibility.

Blackmagic Device source has the following options:

- Device: Dropdown list that allows you to select the Blackmagic Design device you wish to use.
- Mode: Sets the video mode of the device. This should match your media (i.e. camera or game console) output resolution and framerate.
- Format: Selects the video format the device runs in. This should also match your media output you wish to capture.
- Use Buffering (Checkbox): Enables or disables the use of buffering on the video/audio for the device. This can help with issues on systems with low resources available or a device with bad drivers/other hardware issues.

BrowserSource

Browser source is one of the most versatile sources available in OBS. It is, quite literally, a web browser that you can add directly to OBS. This allows you to perform all sorts of custom layout, image, video, and even audio tasks. Anything that you can program to run in a normal browser (within reason, of course), can be added directly to OBS.

Browser Source is fairly straight forward with its options.

- Local File (Checkbox): Tells the source if you are loading a web page from your local machine, or remotely.
- Width: Sets the viewport width of the browser page.
- Height: Sets the viewport height of the browser page.
- FPS: Sets the FPS the browser page will render at.
- CSS: By default, sets the background to be transparent, removes margins on the body, and hides the scroll bar (if the page renders larger than your viewport width/height)
    - **Default CSS**: body { background-color: rgba(0, 0, 0, 0); margin: 0px auto; overflow: hidden; }
- Shutdown source when not visible (Checkbox): Unloads the page when the source is no longer visible (by clicking the eye icon to hide, or not in the active scene).
- Refresh browser source when scene becomes active (Checkbox): Refresh the page when it becomes active (scene is switched to)
- Refresh cache of current page (Button): CLicking this will immediately refresh the page and reload any content.

As Browser Source is based on CEF, any CEF flags (--enable-gpu for example) can be passed from the OBS Studio shortcut. A fairly comprehensive list can be found here

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Color Source

As the name implies, this source creates a solid color for you to add to your scene. This can be used for things like background colors or even a global color tint by using the alpha channel.

Color Source has the following options:

- Color: Allows you to set the color the source will display. This is also where you set any desired alpha.
- Width/Height: This allows you to set the resolution of the source. The primary point of changing this would be to set an aspect ratio that matches your canvas display.

Display Capture

Display Capture is used to capture your entire monitor. The options here are few, with a selection for the display you wish to capture, and a checkbox to enable or disable showing of your cursor.

You can only add one display capture source per display. If you need your display in multiple scenes, make sure you use Add Existing!

Game Capture

Game Capture is one of the primary sources that many users will be looking to use. This Source lets you directly capture the game you are playing, so long as it's DirectX or OpenGL. Game Capture is the most efficient way you can add your games to OBS, and should always be tried first. There are only a very small number of games that do not work with Game Capture. If you have issues with it and cannot solve them on your own, stop by the support chat!

Once you add your Game Capture Source, you will need to select a few options that suit your current needs. First is the Mode.

- Capture any fullscreen application
- Capture specific window
- Capture foreground window with hotkey

The first option, **capture any fullscreen application**, will automatically detect any game running fullscreen on your primary monitor, and add the video output to the Game Capture source in OBS. If you play your games fullscreen, this is the option you should choose. Note that if you only have a single monitor, alt+tabbing out of the game to check OBS will cause the game to stop rendering, so you won't see it show up in the OBS preview.

The second option, **capture specific window**, allows you to simply select the active game you want to capture. Window Match Priority lets you select which parameters will be used to separate the available windows. For example, if you have two clients of the same game running with the same executable name, but different window titles, changing the Window Match Priority to Window Title can be very helpful in selecting the proper game to be captured. Experiment if you have issues selecting the right game and see which works best for you.

The final option, **capture foreground window with hotkey**, lets you set a specific key to tell the Game Capture source which game you want to show up. This is very useful if you change games often during a stream, and you don't want to have to go back into the Game Capture properties every time to select your new game. The hotkey can be set under Settings -> Hotkeys once the Game Capture source is added to your scene.

There are several other options, and we'll give a short description of them. In general, the default options are sufficient for most applications and you should not change them unless you know why you need to, and understand what they mean. As always, if you have any questions, please stop by the forums or chat.

- Multi-adapter Compatibility: Used for systems that have multiple GPUs (such as a laptop). It changes the capture method from shared texture capture to memory capture. Memory capture is ***far less efficient*** than shared texture capture, and this option should only be enabled if you have no other options.
- Force Scaling: Allows you to force a scale on the capture source.
- Limit capture framerate: Limits the Game Capture source from capturing at a frame rate higher than OBS it set to use.
- Capture Cursor: As implied, this will either show or hide the mouse cursor in your game. Does not apply to mouse cursors rendered in the game itself, those will always be captured.
- Use anti-cheat compatibility hook:
- Capture third-party overlays (such as steam): If your game has an overlay that doesn't conflict with Game Capture, this will allow Game Capture to capture it as well.

Image

This source allows you to various image types to your scene. Most image formats are supported. Alpha channel support is also available where applicable.

The only options in Image Source are the path to the image, and unload image when not showing. This checkbox will unload the image from memory while it is not active, which can be useful if you have a large amount of images and few system resources available.

Image Source supports the following image formats: .bmp, .tga, .png, .jpeg, .jpg, and .gif.

Image Slide Show

The Image Slide Show Source allows you to add multiple images that will rotate through as a slide show. To use this source, click the + sign to add either individual files or directories to be loaded. Once you have all the images you want, you can configure how you want them to display.

- Transition: This drop down box allows you to select the animation type that will play during a transition between files. The default is a simple fade, but it can be changed to a cut, slide, or swipe
- Time Between Slides (milliseconds): As the name implies, the time each image will be displayed before initiating a transition to the next.
- Transition Speed (milliseconds): This is how fast the transition animation will take to go from starting to completely changed to the next image.
    - **NOTE:** This value ***does not*** increase the time between slides. For example, if you have the Time Between Slides set to 10000ms, and the Transition Speed set to 2000ms, the transition will BEGIN at 8000ms from the end of the last transition, with the next slide being fully visible at 10000ms. If this value is HIGHER than the Time Between Slides, it will automatically reduce the time to match. It also cannot be lower than 50ms.
- Randomize Playback (checkbox): This option will let you choose if you want to randomize the images being played. If this is enabled, the next image will be randomized at each transition. If disabled, they will be shown in the order of the file list.

Image Slide Show Source supports the following image formats: .bmp, .tga, .png, .jpeg, .jpg, and .gif.

Intel® RealSense™ 3D Camera GreenScreen

This source type will allow you to use the digital green screen features of an Intel RealSense camera. RealSense cameras have a combination of a normal camera, infrared lasers, and an infrared camera to map the space in front of your PC and detect which parts of the video can be removed in a green screen effect. There is no configuration necessary for this source, simply add the source, pick your camera, and watch the magic!

Media Source

Media source is a great option to add all sorts of different media types to your stream. The currently supported file types are:

- Video: .mp4, .ts, .mov, .flv, .mkv, .avi, .gif, .webm
- Audio: .mp3, .aac, .ogg, .wav

Simple click the Browse button to select your file, or uncheck the "Local file" box to allow a URL or other remote location to be added. For remote files, the URL/path goes in Input, and generally Input Format can be left blank.

Once the file has been selected, there are only a few options that would need to be looked at.

- Loop: Sets if the file will loop back to the beginning once playback has completed. Useful for .gifs
- Restart playback when source becomes active: This will allow you to set the file to restart once the source is active. Active means in the current visible scene
- Use hardware decoding when available: Fairly self explanatory!
- Hide source when playback ends: If enabled, the source will automatically hide itself when the file has completed playback. Useful for video files so they do not show the last frame indefinitely.
- Advanced: These options should only be access by users who understand what they are, and why they need them, so they will not be covered in this guide.

Scene

This Source is often overlooked, and while one of the simplest in function, it can allow for some of the most powerful functionality in OBS Studio. Since all Scenes are considered Sources, you can add an entire Scene as a Source anywhere you want.

For example, this can allow you to create a static overlay that you want to use in every Scene, called Overlay. In several other scenes, let's say Main and Game, you can add the Overlay Scene. If any changes are necessary to any of the Sources in Overlay, you can just update them it will be updated everywhere else. This is just one of many examples on how this Source type can be used!

Text (GDI+)

Text source can allow you to add simple text renders to your stream or recording layout. To get going, most of the default settings will be fine (except maybe the colors), and you can just type what you want your text to say in the field labeled "Text."

If you want to load the text from a file, simple check the "Read from file" checkbox, and select the file that has the text you want to read in it. The file must be UTF-8 (most default text files will be), and the file will be reloaded on save. This means you can edit the file and it will automatically update.

Once the text is present, there are quite a few options for styling it. You can:

- Change the foreground color
- Change the background color
- Add a gradient
- Adjust the opacity (transparency/alpha)

- Set the horizontal and vertical align (relative to the red source bounding box)
- Add an outline (with options for size, color, and opacity)
- Use custom text extents for the size of the source, as well as if the text should wrap if the width is exceeded.

VLC Video Source

Similar to Media Source, you can add video and other media files to this source to be played in your scenes. This source type will use the VLC libraries for extended media support over the built-in Media Source. It requires that VLC be installed on your system to show up as an available source in OBS. If you are using 64bit OBS, you must install 64bit VLC and if you are using 32bit OBS, you must install 32bit VLC.

To add files, click the + sign to browse to your file, directory, or URL that you wish to add. If you add multiple, they will be played in the order they were added. If Loop playlist is checked, the playlist will start over once the end is reached.

The Visibility behavior drop down menu allows you to choose how visibility affects playback of the files. They should be self-explanatory.

Video Capture Device

The Video Capture Device Source allows you to add a variety of video devices, including but not limited to webcams and capture cards. On Windows, for a device to work with OBS the drivers needs to support DirectShow output. Since DirectShow is the standard output format for Windows, there's a good chance that this is what your device outputs. All major webcams and capture cards support DirectShow and will work with OBS.

To add your device, simply add the Source, open the properties and select your device from the Device drop down list.

You then have several options to configure it:

- Deactivate/Active (Button): Clicking this will either turn your device off (when it reads Deactivate) or on (when it reads Activate).
- Configure Video (Button): This button will open any driver configuration utilities. For example, with a Logitech webcam it will open the Logitech camera configuration software for enabling custom options like facial tracking. With a capture card, it might open the options to configure the input resolution and FPS.
- Configure Crossbar (Button): Opens the device's Crossbar Configuration, if available. Consult your device's manual for more information.
- Resolution/FPS Type: Most of the time this can be left on Device Default. If you are having issues with your device showing up, change it to custom and the following options become available to set manually:
    - Resolution: This sets the base resolution for the video device. Make sure it's a mode that your device supports!

    - FPS: Sets the FPS of the device.
    - Video Format: If your device supports multiple video output formats (Such as MJPEG or XRGB), you can select the preferred format here.
- YUV Color Space: Sets the color space the device will output in.
- YUV Color range: Sets the color range the device will output in.
- Buffering: This has three different modes, Enable, Disable, and Auto-Detect. Enable will turn buffering on, which can help if you are getting stuttering during playback. Setting to Disable will turn buffering off, which can help if you are experiencing a delay on the device. Auto-Detect is recommended, as it will attempt to ask the device which method is preferred.
- Flip Vertically (Checkbox): Flips the video image vertically. Some devices will (rarely) send the video input upside down.
- Audio Output Mode: You can set to Capture Only (meaning you will not hear, without Audio Monitoring), or Output desktop audio (DirectSound/WaveOut). Enabling desktop output will send the device's audio out through your system default device.
- Use custom audio device (Checkbox): When enabled, the Audio Device selection becomes available, and allows you to use a separate audio device that will be linked to your video device. This can be useful if you want to use an external microphone on a webcam, for instance, and want to tie the audio directly to the Source itself.

Window Capture

Window Capture allows you to capture a specific window and its contents. The advantages to using this source over Display Capture is that only the selected window will be shown, even if there are other windows in front of it (**WIN7 NOTE**: Having Aero disabled can interfere with this functionality.).

Deprecated Sources

List of currently deprecated sources. Sources listed here should not be used unless aboslutely necessary, and are only left in for backwards compatibility reasons.

- Text (FreeType2)

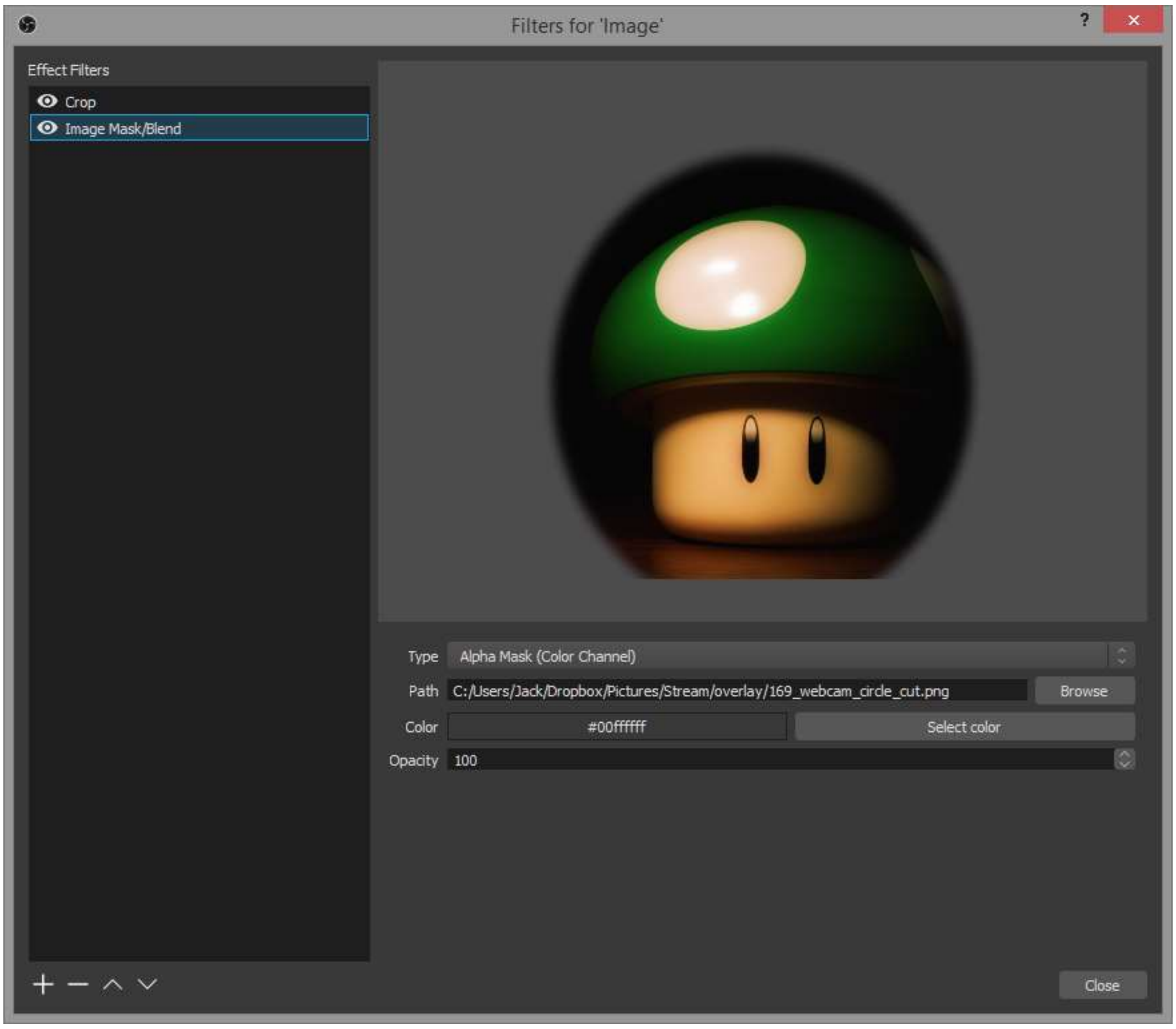
Filters Guide

In OBS Studio we have the ability to add filters to our Sources, Scenes and even our Audio Devices. The following filters are available in OBS Studio 23.0.0:

You can add them by right-click your desired Scene, Source or Device and selecting "Filters" (for Audio Devices, click on the gear icon next to your device). But let me explain what the different filters allow you to do.

Scene and Source Filters

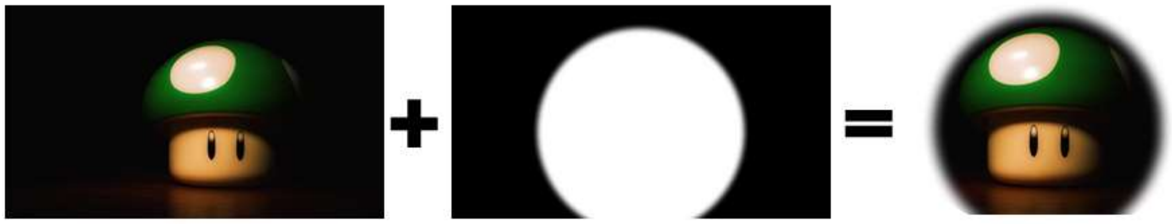
Image Mask/Blend



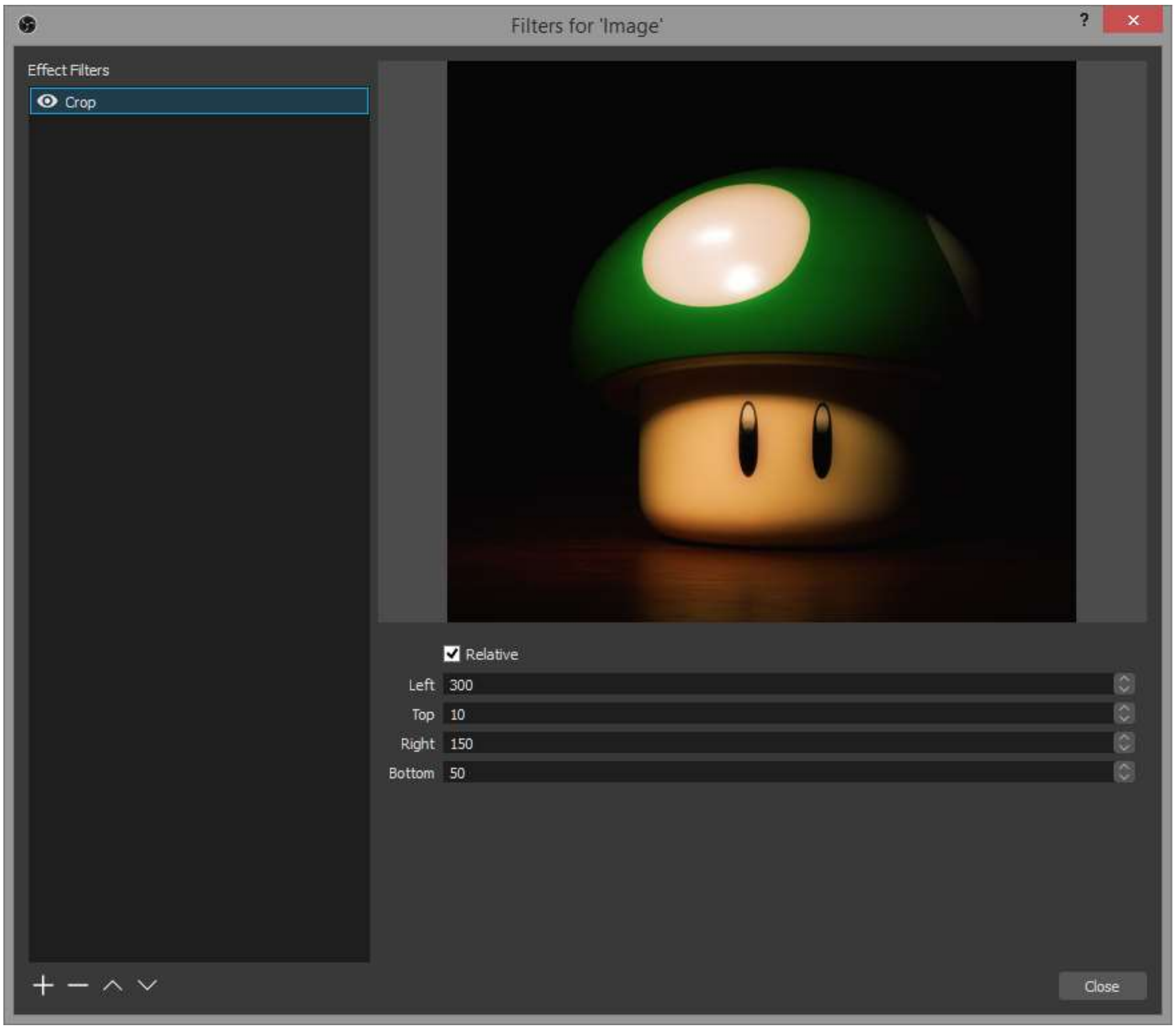
Crop
Image Mask/Blend

Type Alpha Mask (Color Channel)
Path C:/Users/Jack/Dropbox/Pictures/Stream/overlay/169_webcam_circle_cut.png
Color #00ffffff
Opacity 100

The Image Mask/Blend filter gives us the option to use the Color or Alpha Channel of an Image as a Mask or to Blend an Image (multiply, addition, subtraction) over your Scene or Source. This can be used to give your webcam a round border for example:



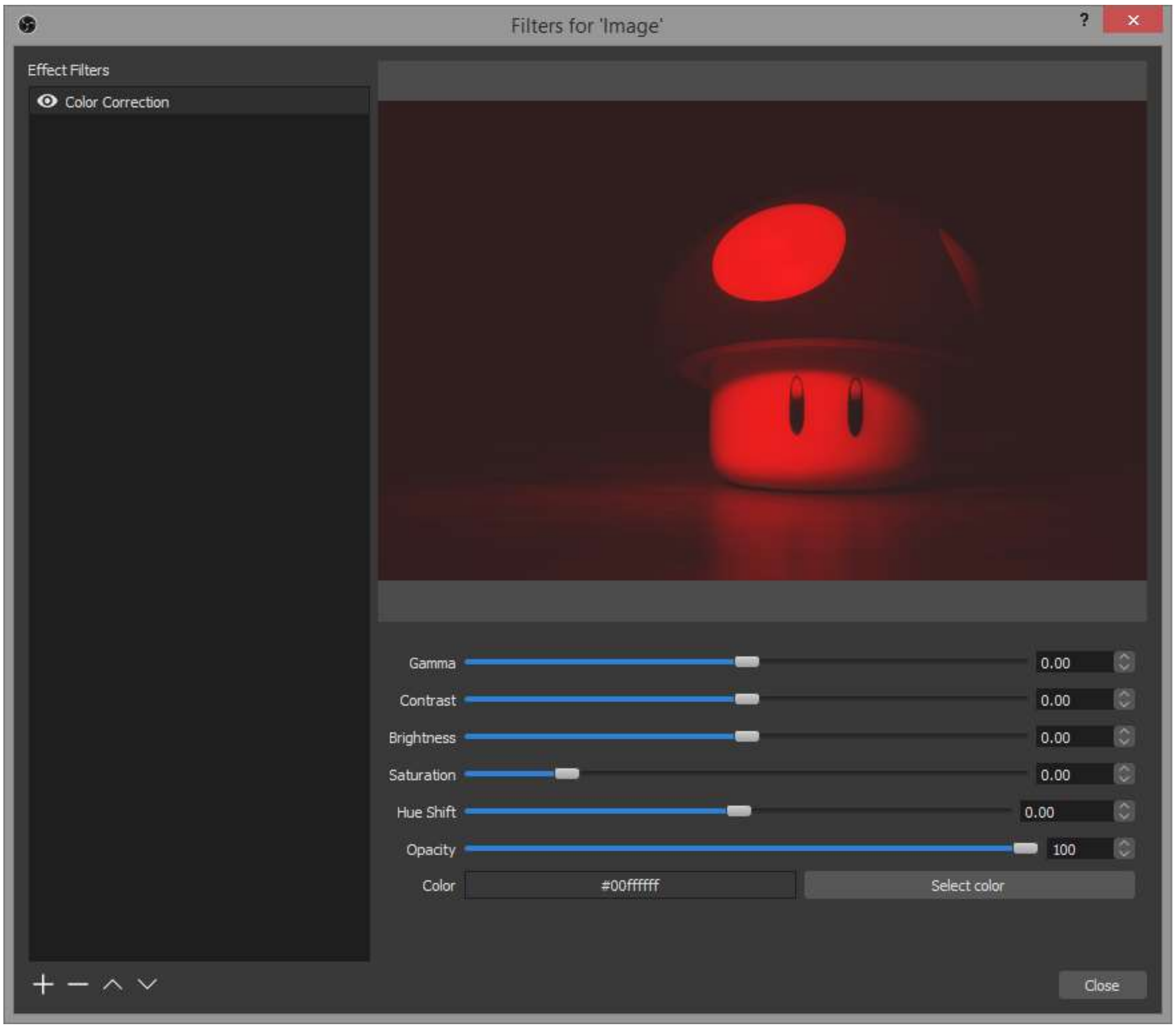
Crop



☑ Relative
Left 300
Top 10
Right 150
Bottom 50

The crop filter should explain itself but in short it lets you cut off the top/left/right/bottom of your source/scene to only show the parts you want.

Color Correction


Filters for 'Image'

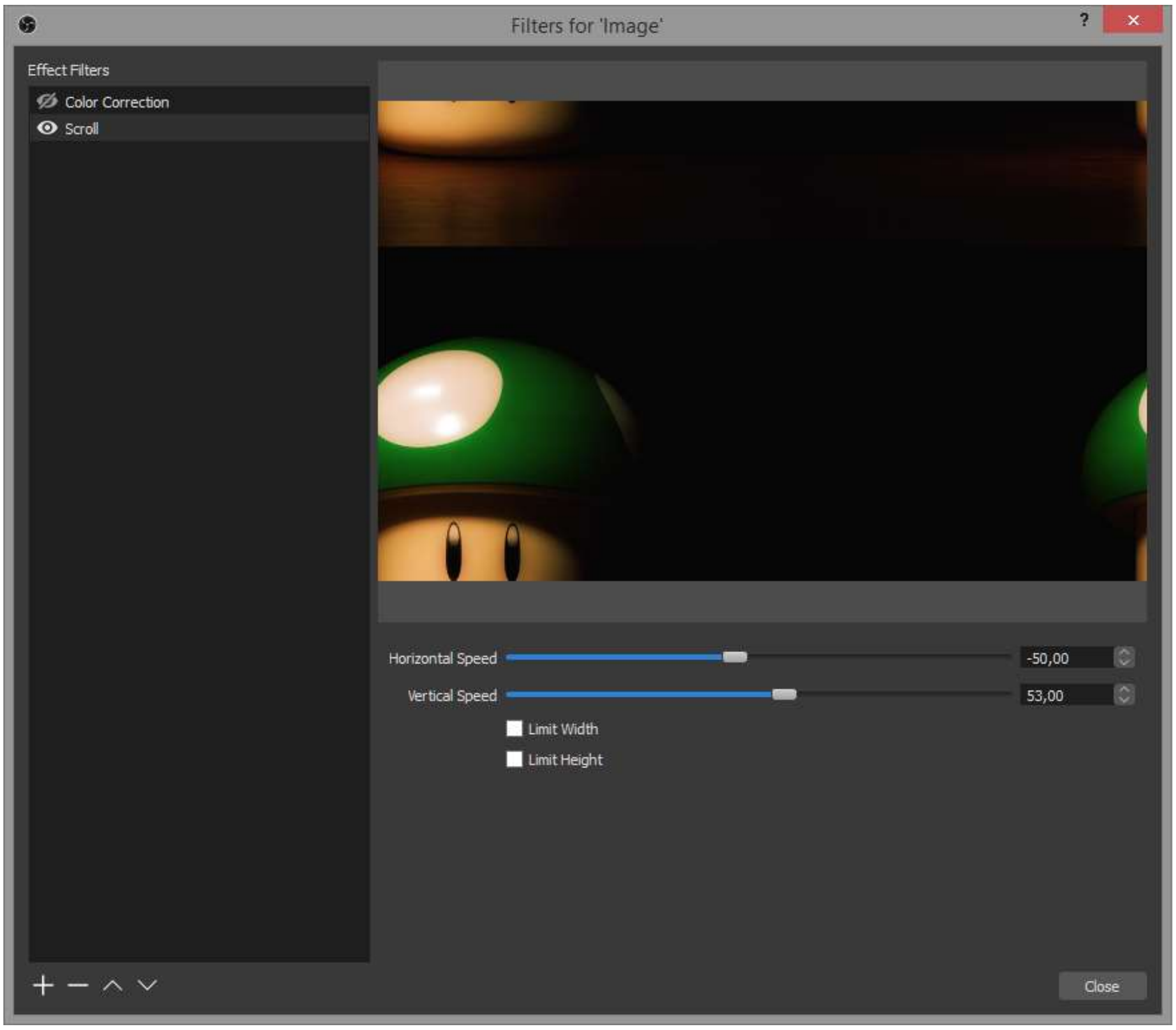
Effect Filters

Color Correction

Gamma	0.00
Contrast	0.00
Brightness	0.00
Saturation	0.00
Hue Shift	0.00
Opacity	100
Color	#00ffffff

Again, the name says it all. You can change the contrast, brightness and gamma of your source and even provide a color overlay.

Scroll

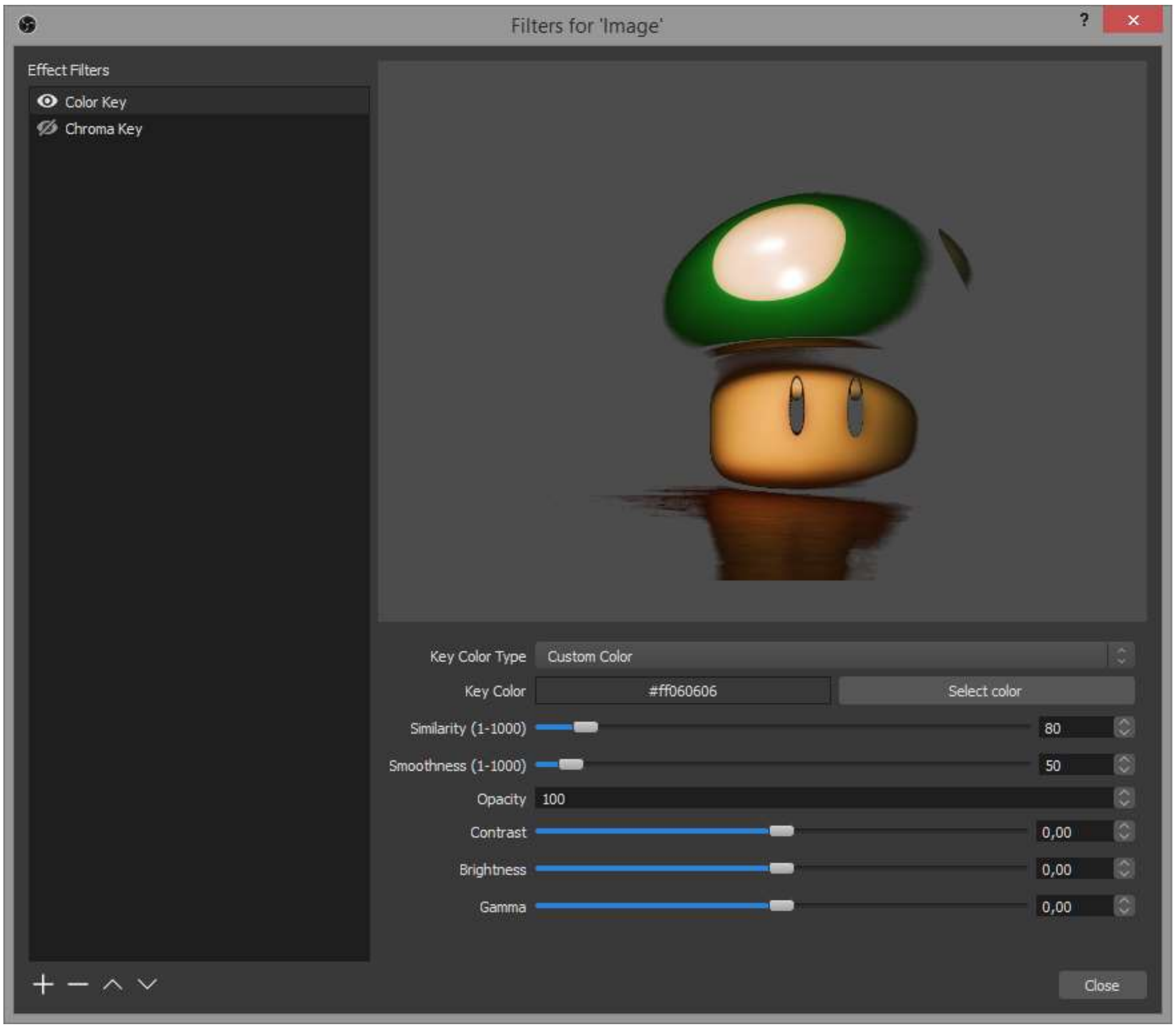


Horizontal Speed -50,00
Vertical Speed 53,00
Limit Width
Limit Height

The scroll filter gives us the ability to give our text for example a scrolling effect, left-to-right and top-to-bottom. Negative and positive values will change the direction in which your source will scroll and you can limit the height and width if necessary.

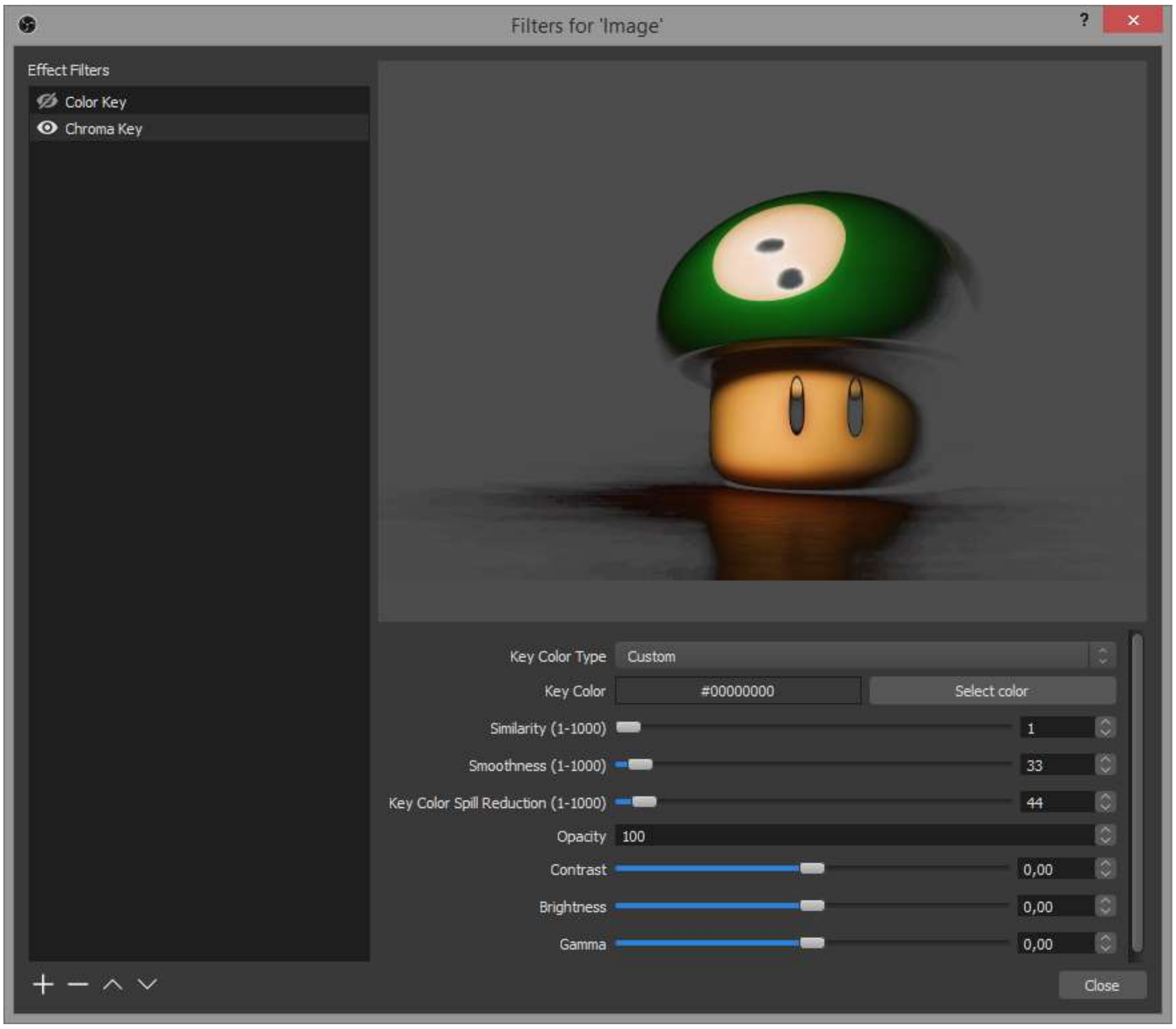
Color Key and Chroma Key

Color Key:



Key Color Type: Custom Color
Key Color: #ff060606
Similarity (1-1000): 80
Smoothness (1-1000): 50
Opacity: 100
Contrast: 0,00
Brightness: 0,00
Gamma: 0,00

Chroma Key:



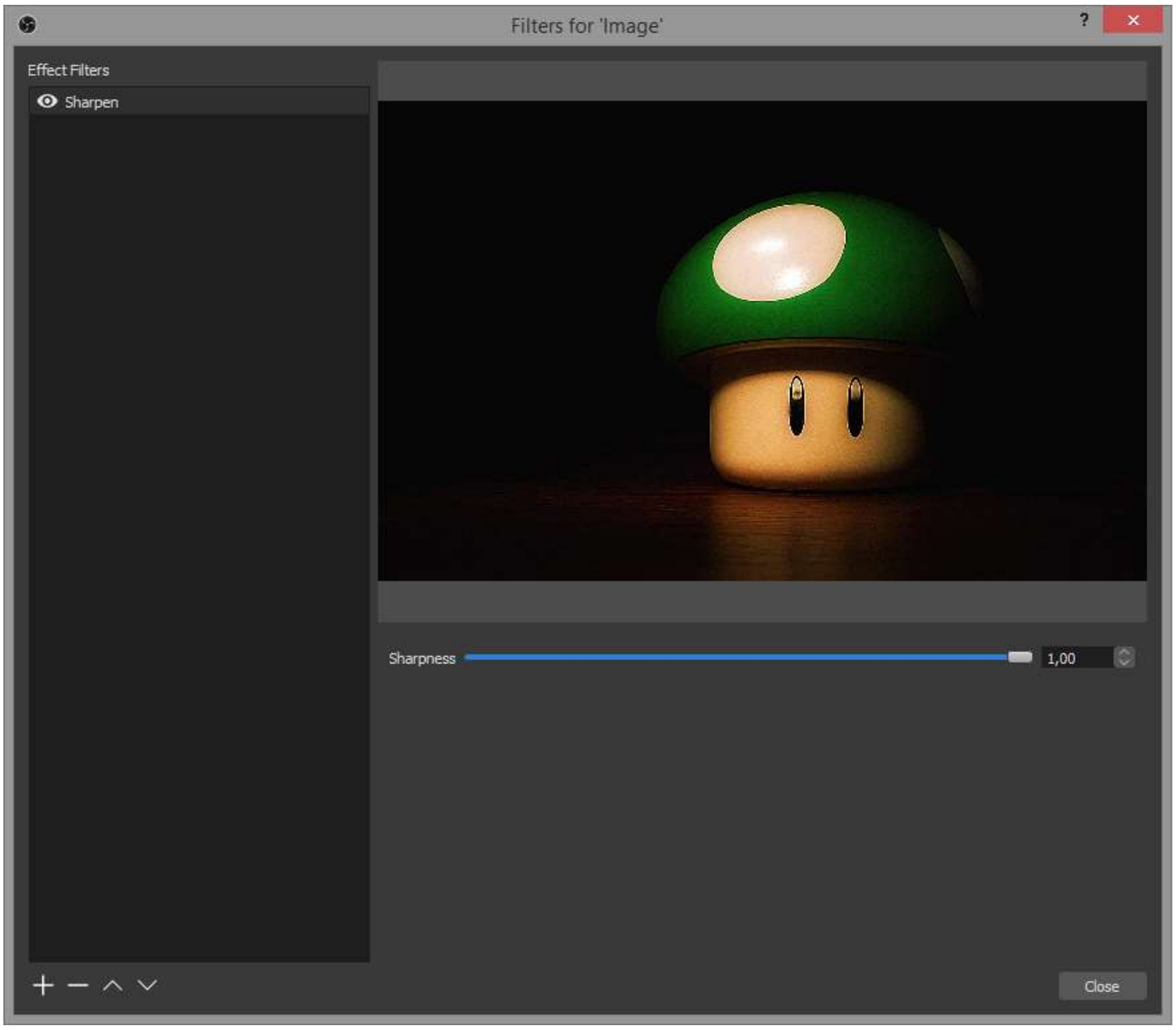
Key Color Type: Custom
Key Color: #00000000 — Select color
Similarity (1-1000): 1
Smoothness (1-1000): 33
Key Color Spill Reduction (1-1000): 44
Opacity: 100
Contrast: 0,00
Brightness: 0,00
Gamma: 0,00

Both the Color Key and Chroma Key filter can be used to remove a certain color of your source and make it transparent. This can be used for green screens and similar stuff. They behave slightly differently, so you will need to experiment and see which works best for your personal use case.

LUT Filter

This filter allows you to apply a LUT to your video sources.

Sharpen



Sharpness 1,00

The sharpen filter should explain itself as well, if you feel your webcam input for example is a bit blurred and you want to improve the overall sharpness a bit, add the filter and test with different values.

Audio Device Filters

Compressor

A compressor is very useful if your source (typically a microphone) is set for a normal level but can sometimes spike much louder, such as impromptu shouting or getting into a heated discussion. It will automatically lower the source's volume to reduce the likelihood of it peaking above 0dB, which can cause clipping and distortion, and then turn it back up once the volume is back to normal.

In short, a compressor makes loud sounds quieter and typically would be placed at or near the beginning of your filter chain.

- **Ratio**: The amount of compression or gain reduction to apply to a signal that is above the threshold. For example, 2:1 will be a weak compression (this translates to an audio

level 6dB above the threshold will be 3dB above after the compression), while 6:1 will be a much stronger compression.
- **Threshold**: Once the signal reaches this level the compressor will begin to apply compression at the set ratio. When levels are below the threshold the ratio is 1:1 which translates to no gain reduction.
- **Attack**: How quickly, in milliseconds, you want the compressor to reach full gain reduction when levels exceed the threshold.
- **Release**: How quickly, in milliseconds, you want the compressor to return to zero gain reduction after levels drop below the threshold.
- **Output Gain**: When you compress a signal it typically ends up quieter which reduces your average level. Applying Output gain brings the average level of the source back up which can help improve its presence over-top of other audio sources.
- **Sidechain/Ducking Source**: When a compressor is placed on an output audio source such as your Desktop, Sidechain can use the input from a microphone/aux source to reduce the volume of the output source. See Sidechain Compression/Ducking.

Note. A source can still exceed 0dB with a compressor if your input level is too loud, your ratio/threshold are set incorrectly or you apply too much output gain. To ensure you do not exceed 0dB you can use a Limiter at the end of your filter chain.

Sidechain Compression/Ducking

Sidechain compression, also known as *Ducking*, can be used to make room for your voice when speaking over-top of music and games by lowering your desktop audio when you speak.

The following sidechain compression settings are recommended as a place to start. Adjust the threshold to control the strength of the ducking, attack/release control how quickly the volume changes.

- **Ratio**: 32:1
- **Threshold**: -36dB
- **Attack**: 100ms (How fast the audio will duck out)
- **Release**: 600ms (How fast the audio will return to full volume)
- **Output Gain**: 0dB (You do not apply output gain when using Sidechain compression)
- **Sidechain/Ducking Source**: Microphone

Expander

An expander can be used to reduce background noise such as computer fans, mouse/keyboard clicks, breathing and unwanted mouth noises. An expander reduces the level of an audio signal by applying gain reduction, similar to a compressor but below the threshold instead of above it.

An expander can be used in place of a gate for noise reduction, they can produce a smoother open and close due to having an adjustable ratio whereas a gate is a fixed.

In short, an expander makes quiet sounds quieter and typically would be placed near the end of your filter chain, after any compression/other effects but before a Limiter.

- **Presets**: Defines some default values to use for the Ratio and Release time.
    - **Expander**: Low ratio and release time, good for light noise reduction.
    - **Gate**: High ratio and release time, will gain reduce a signal similar to a gate.
- **Ratio**: The amount of expansion or gain reduction to apply to a signal that is below the threshold. A lower ratio such as 2:1 is good for light noise reduction, a higher ratio such as 10:1 will completely gain reduce a signal. A nice balance is 4:1, this should provide an adequate amount of gain reduction without fully gating the signal.
- **Threshold**: Once the input reaches this level the expander will stop gain reducing the signal. Adjust the threshold until the noise you are trying to reduce is gone, but don't go too far or your voice will begin to get cut off.
- **Attack**: How quickly, in milliseconds, you want the expander to stop gain reducing or *open* once the threshold is exceeded. **An attack between 5-10ms is recommended.**
- **Release**: How quickly, in milliseconds, you want the expander to reach full gain reduction or *close* once the input drops below the threshold. **A release between 50-120ms is recommended.**
- **Output Gain**: Increases the output level of the expander by applying gain, generally not needed but can be used to increase your microphone level before outputting it.
- **Detection**: Changes how the input level is measured, affecting the sensitivity of the threshold detection. **RMS is recommended.**
    - **RMS**: Averages the input level measurement over the last 10ms to reduce the sensitivity of the threshold detection, helps smooth out and prevent the expander from opening due to quick little peaks of noise.
    - **Peak**: Input level measurement is not averaged over time, expander is quicker to react to peak level changes.

Gain

Gain should generally be applied at the source before it reaches OBS, but if needed the gain filter can help with very quiet audio sources to increase the output volume.

Invert Polarity

Used to correct phase cancellation issues.

Limiter

Limiters are used to prevent an audio signal from peaking above 0dB which can cause clipping and distortion. A limiter is a special type of compressor with a very fast attack and a very high ratio.

- **Threshold**: The maximum output level an audio signal can hit, no signal can exceed this level.
- **Release**: Because a limiter is a compressor it applies gain reduction to brick-wall the output level. If and when a signal tries to exceed the threshold, the release is how

quickly the limiter will stop gain reducing after the level drops below the set threshold.

When using a Limiter it should be the last filter in your chain.

Noise Gate

The Noise Gate allows you to cut off all background noise while you are not talking. Select a close threshold above the noise volume and an open threshold slightly below your voice input to get good results.

Noise Suppression

The Noise Suppression filter can be used to remove mild background noise or white noise that may be in any of your audio sources. While this is generally not effective at large amounts of background noise (i.e. in a loud room) it can be quite effective at reducing things like PC fan noise or other environmental noises.

0 is off. The further you move the slider to the left, the 'stronger' the filter will be, and the more sounds it will filter out. Keep in mind that this can distort other sounds (like your voice).

VST Plugin

OBS Studio supports many VST2.x plugins. Adding a VST plugin is as simple as adding any other audio filter, but there are some limitations. VST1.x, VST3.x, MIDI control/input in VST plugins, and shell VST plugins are not supported at this time. We have not tested all plugins, and some VST plugins may cause crashes. Make sure you save and back up any settings to avoid loss of data when experimenting with VST.

Lastly, always keep an eye on CPU usage, some VST plugins can be very CPU hungry!

OBS Studio will search for plugins in the following locations:

- Windows (*.dll)
  - C:/Program Files/Steinberg/VstPlugins/
  - C:/Program Files/Common Files/Steinberg/Shared Components/
  - C:/Program Files/Common Files/VST2
  - C:/Program Files/Common Files/VSTPlugins/
  - C:/Program Files/VSTPlugins/
- macOS (*.vst)
  - /Library/Audio/Plug-Ins/VST/
  - ~/Library/Audio/Plug-ins/VST/
- Linux - NOT YET IMPLEMENTED (*.so and* .o)
  - /usr/lib/vst/

  - /usr/lib/lxvst/
  - /usr/lib/linux_vst/
  - /usr/lib64/vst/
  - /usr/lib64/lxvst/
  - /usr/lib64/linux_vst/
  - /usr/local/lib/vst/
  - /usr/local/lib/lxvst/
  - /usr/local/lib/linux_vst/
  - /usr/local/lib64/vst/
  - /usr/local/lib64/lxvst/
  - /usr/local/lib64/linux_vst/
  - ~/.vst/
  - ~/.lxvst/
  - NOTE: If the user has set the VST_PATH environmental variable, OBS will ignore the other search locations and just use the locations listed in VST_PATH.

A short list of free plugins that were used to develop and test the VST support in OBS Studio can be found below. Your experiences may differ, but these are the ones we know have been tested to work in our environments:

- https://www.reaper.fm/reaplugs/
- https://www.meldaproduction.com/download/plugins

Untested, but highly reviewed:

- https://varietyofsound.wordpress.com/
- http://www.vst4free.com/index.php?dev=Kjaerhus_Audio
- http://www.lesliesanford.com/vst/plugins/

In the future more filters will be added to OBS Studio, so always keep an eye out for the next update of the software.

`Original guide by Jack0r, updates/edits by Fenrir and the #obs-dev team`

Keyboard Shortcuts

Sources List Shortcuts

Only active when a source is selected

General

- Copy Source: Ctrl+C
- Paste Source: Ctrl+V
- Remove Source: Del

Order

- Move source up in sources list: Ctrl+Up
- Move source down in sources list: Ctrl+Down
- Move source to top of sources list: Ctrl+Home
- Move source to bottom of sources list: Ctrl+End

Transform

- Edit transform: Ctrl+E
- Reset transform: Ctrl+R
- Fit to screen: Ctrl+F
- Stretch to screen: Ctrl+S
- Center to screen: Ctrl+D

Preview Window Shortcuts

Only active when a bounding box is selected in the preview window

- Nudge source up: Up arrow
- Nudge source down: Down arrow
- Nudge source left: Left arrow
- Nudge source right: Right arrow
- Crop a source: Hold alt and drag the bounding box (edges should turn green)
- Stretch a source: Hold shift and drag the bounding box
- Temporarily disable source snapping (if enabled): Hold ctrl while moving/adjusting bounding box

Only active when the Preview is set to Fixed Scaling (canvas or output)

- Move preview: Hold space and drag it with your mouse
- Zoom preview: Hold space + mouse scroll wheel

Surround Sound Streaming And Recording

These features require OBS-Studio 21.0 or newer.

Table of Contents:

Overview

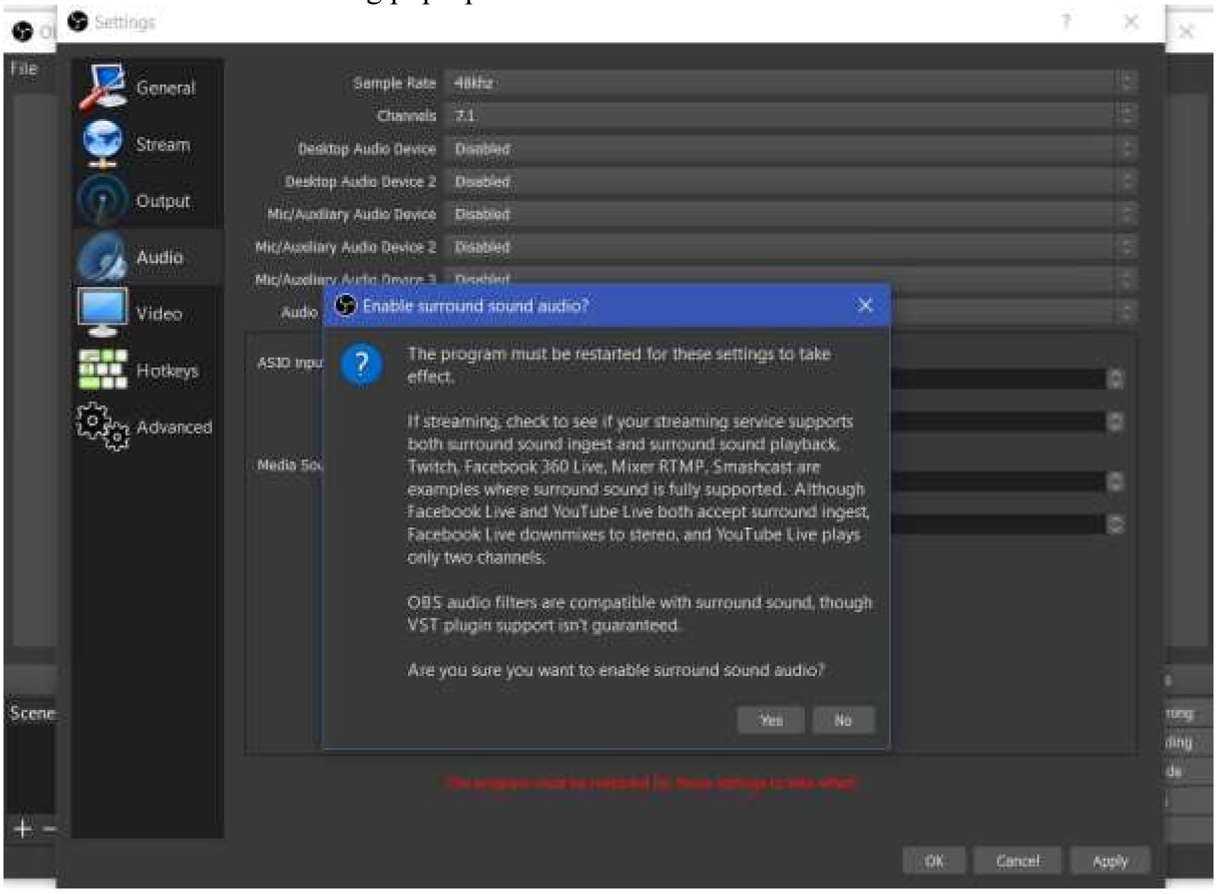
OBS-Studio is the first mainstream streaming software to support **surround sound** streaming and recording.
Traditionally this is a feature reserved to professional broadcast appliances.
OBS-Studio can stream and record up to 8 audio channels.
The audio channels can be surround sound channels or more general multichannel ones.

Use cases:

- **gaming**
- **music**
- **ambisonics**
- **multilingual streaming**

Setup

- **Settings > Audio** : just select a channel different from Mono or Stereo in the Channels list. Click OK to the Warning pop-up and restart OBS.



Sample Rate: 48khz
Channels: 7.1
Desktop Audio Device: Disabled
Desktop Audio Device 2: Disabled
Mic/Auxiliary Audio Device: Disabled
Mic/Auxiliary Audio Device 2: Disabled
Mic/Auxiliary Audio Device 3: Disabled

Enable surround sound audio?

The program must be restarted for these settings to take effect.

If streaming, check to see if your streaming service supports both surround sound ingest and surround sound playback. Twitch, Facebook 360 Live, Mixer RTMP, Smashcast are examples where surround sound is fully supported. Although Facebook Live and YouTube Live both accept surround ingest, Facebook Live downmixes to stereo, and YouTube Live plays only two channels.

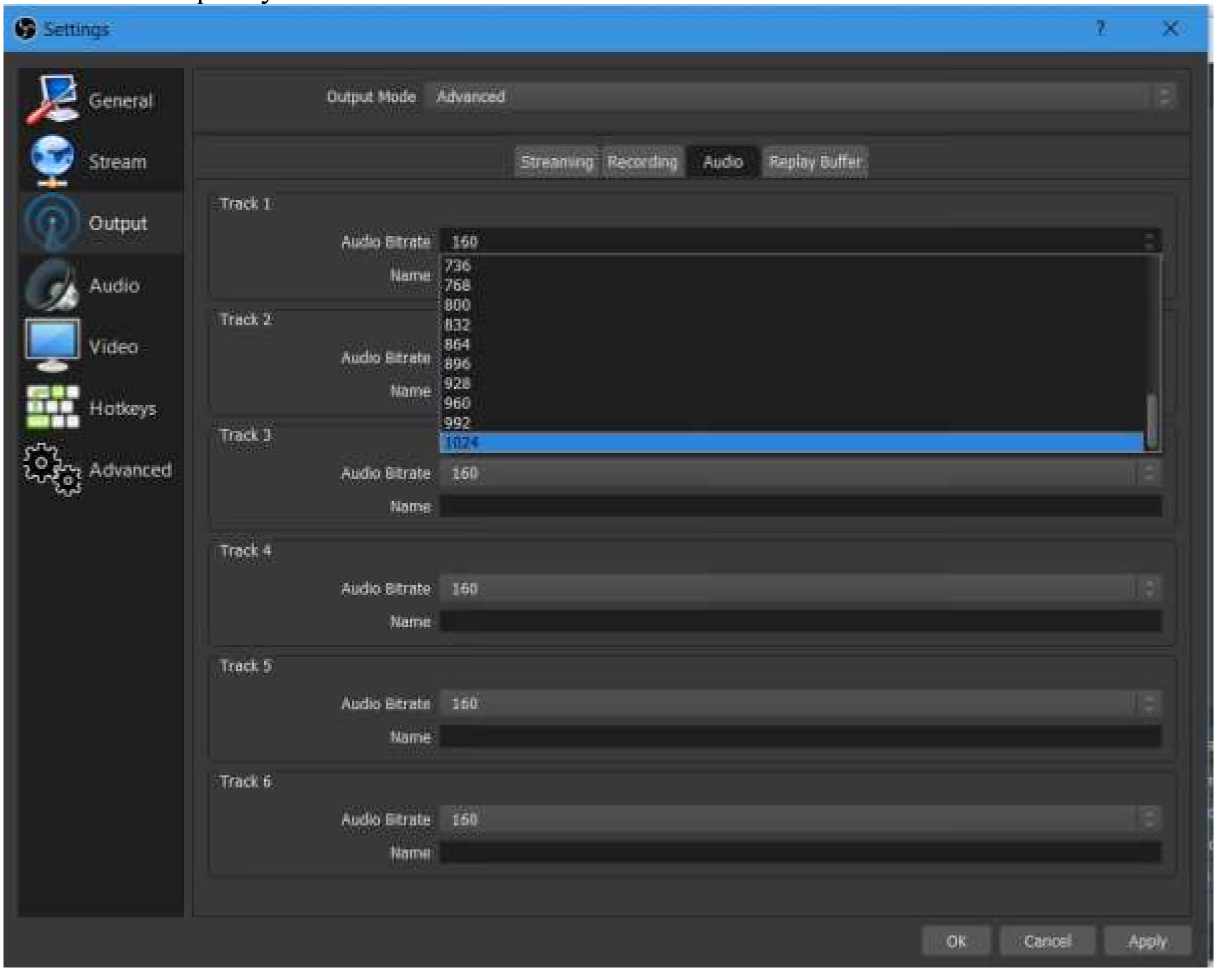
OBS audio filters are compatible with surround sound, though VST plugin support isn't guaranteed.

Are you sure you want to enable surround sound audio?

[illegible]

- **Settings > Output** :
- **Output Mode** : select Advanced
- **Advanced > Audio Tab** : larger bitrates are unlocked for surround sound (up to 1024 kbs). Select a large bitrate. By default OBS-Studio selects 160 kbs for stereo. This is insufficient for surround sound. As a rule of thumb 64 kbs x number of channels

should be CD quality.



Output Mode: Advanced

Track 1
Audio Bitrate: 160
Name
736
768
800
832
864
896
928
960
992
1024

Track 2
Audio Bitrate
Name

Track 3
Audio Bitrate: 160
Name

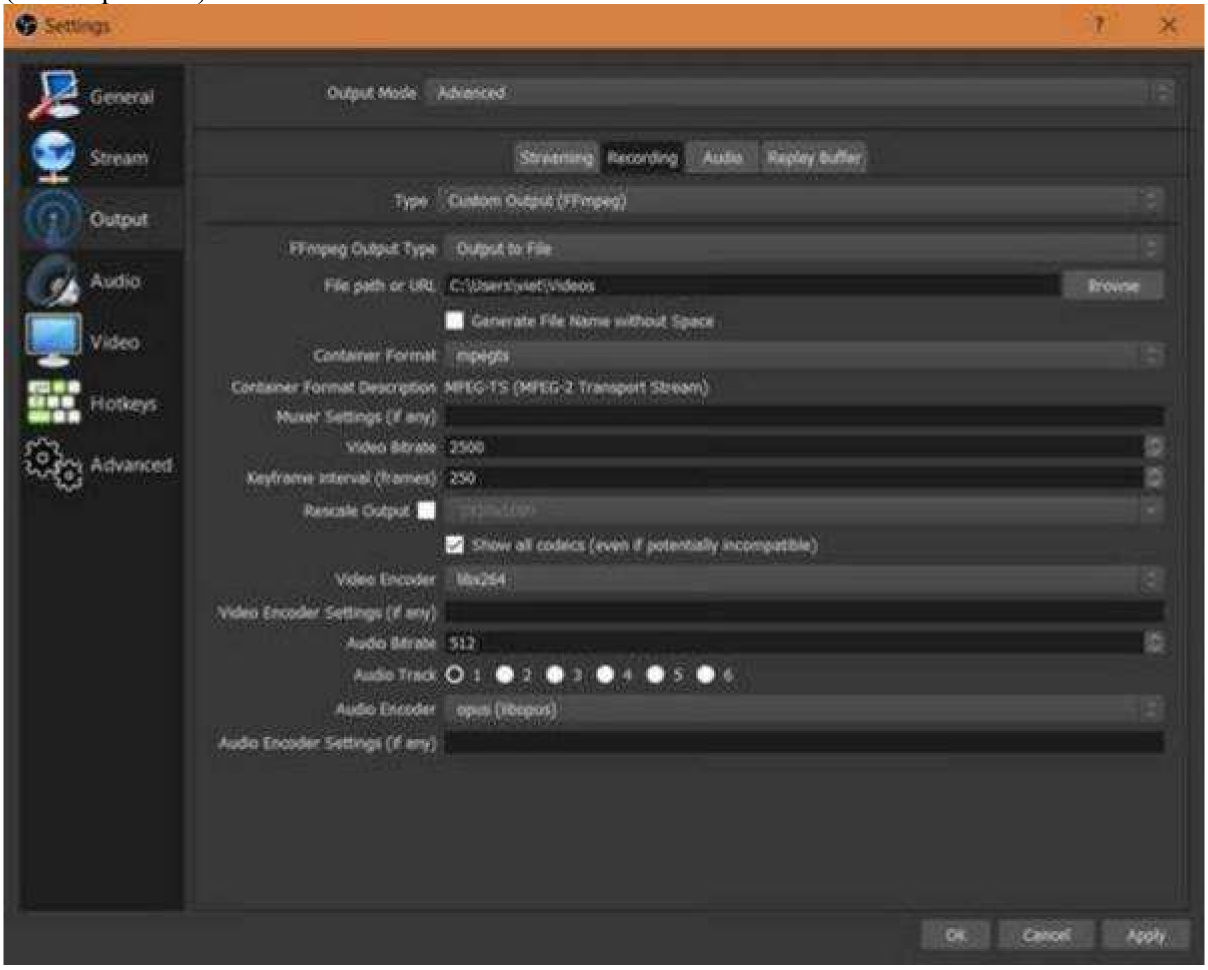
Track 4
Audio Bitrate: 160
Name

Track 5
Audio Bitrate: 160
Name

Track 6
Audio Bitrate: 160
Name

- **Recording** :
  either standard or Custom FFmpeg recording can be selected.
  In the prior case the codec used will be aac ; in the latter case, all (free) codecs supported by FFmpeg are available.
  (check the box 'Show all codecs').
  For surround sound, will be of interest: libopus, aac (native), pcm formats

(uncompressed).



Output Mode: Advanced

Type: Custom Output (FFmpeg)

FFmpeg Output Type: Output to File

File path or URL: C:\Users\viet\Videos

☐ Generate File Name without Space

Container Format: mpegts

Container Format Description: MPEG-TS (MPEG-2 Transport Stream)

Muxer Settings (if any):

Video Bitrate: 2500

Keyframe interval (frames): 250

Rescale Output ☐

☑ Show all codecs (even if potentially incompatible)

Video Encoder: libx264

Video Encoder Settings (if any):

Audio Bitrate: 512

Audio Track: 1 2 3 4 5 6

Audio Encoder: opus (libopus)

Audio Encoder Settings (if any):

- **IMPORTANT WARNING**: make sure to select the same channel layout as your input (if you have a 4.1 audio source do not select 7.1). If you don't, channel mixing may (or may not) occur. There is an automatic channel rematrixing when either downmixing or upmixing is mandated by a difference in channel layouts between source and output. This channel rematrixing mixes channels in general. Or it can remove a channel (ex: 3.1 source to 4.0 output removes the LFE channel).
- **DOLBY**
  Streams can be encoded in ac3 /eac (using Output > Advanced > Custom FFmpeg recording > stream to URL). But support of the various streaming services or web players has not been tested.
  Modify this wiki if this works.
  Capture of dolby can be tricky if the channels are lumped into two PCM channels; in order to be decoded correctly and encoded all the channels should be held in different PCM channels.

Gaming and Streaming Services Support

Compatible Streaming Services

As of now, the following services have been tested and are compatible with live surround sound streaming:

- **Twitch**,
- **Mixer** (rtmp but NOT with ftl protocol),
- **Smashcast** ...
- **Facebook Live 360 with spacial audio** (requires ambisonics capture device)

Incompatible Services

- **YouTube Live** (discards channels beyond the first two).
- **Facebook Live** (downmixes all the channels).

Streaming Servers and Players

Apart from streaming to services like Twitch or Facebook Live, you might use your own streaming server which delivers streams to your website.

Compatible Streaming Servers

The following servers have been tested and ingest surround sound in rtmp protocol :

- wowza
- nginx with rtmp module

Note however that server-side recordings only keep the first two channels although.

Compatible Web Players

- html5 players (tested with hls or mpeg-dash):
    - videojs,
    - bitmovin,
    - mediaelement,
    - viblast (hls & dash),
    - hls.js

NB: Flash player accepts rtmp surround sound in Dolby; it might work ==> untested.

Streaming Music

For music bands, Djing, ... , bring your production with surround sound to the World !
Up to 7.1 surround sound is available. (For more channels (up to 16.0) check this fork).

Filters and VST

OBS-Studio has built-in audio filters as well as VST 2 support. The filters are compatible with surround sound; the VST also **if** they originally support multichannel. Check the Filters-Guide.

High Quality Recording

Select Custom FFmpeg recording and a PCM format (e.g. pcm_s24le for 24 bit samples) for uncompressed audio.
For compressed formats, you can select also libopus and aac which will work very well (target at least 64 kbs per channel).

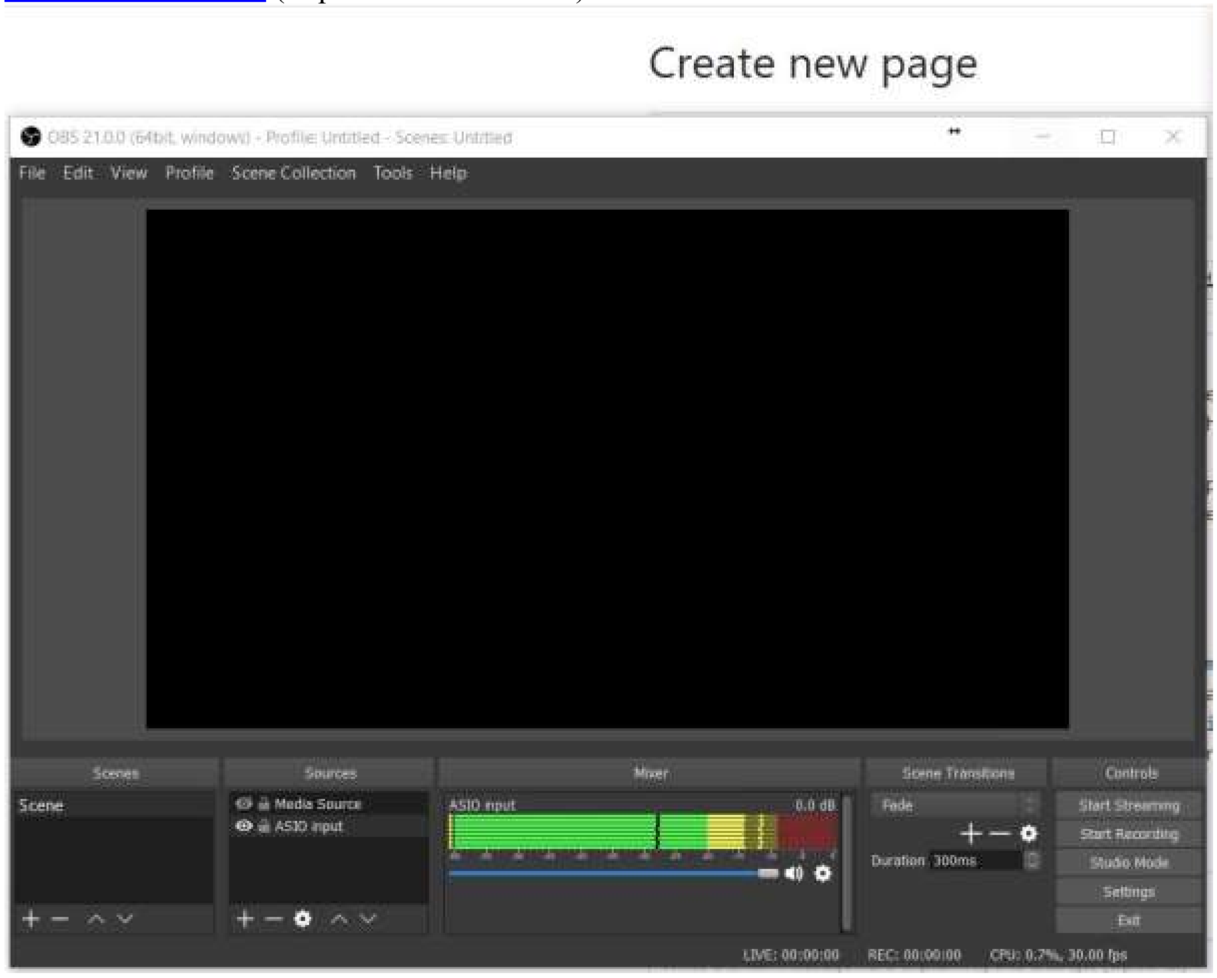
Interfacing with a DAW

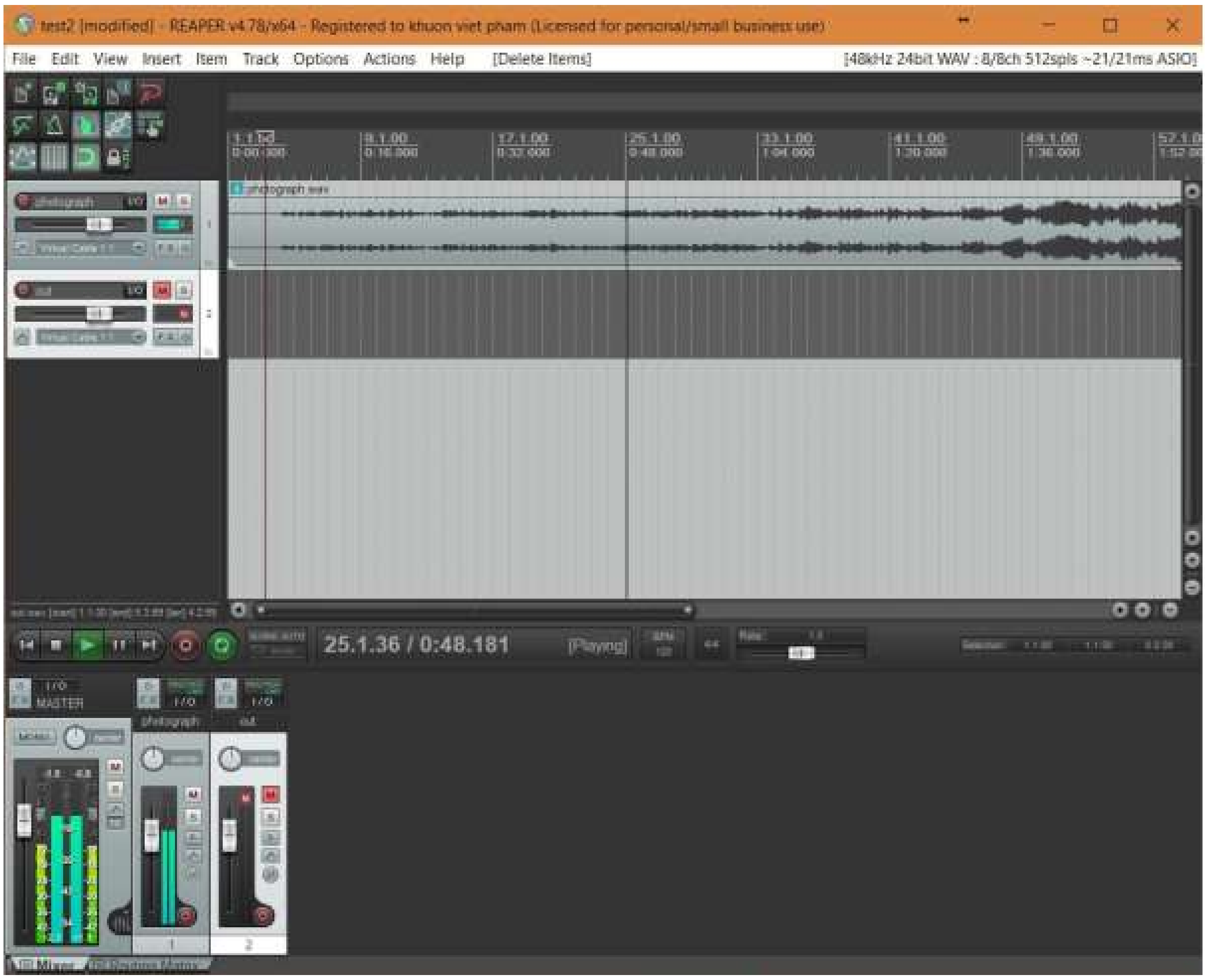
For that you will need some application ensuring the routing.

windows

Reaper :

- SAR (requires having an ASIO driver, for instance ASIO4ALL if you don't own an ASIO card).
- rearoute (from Reaper, with up to 256 channels) if using ASIO plugin
- Voicemeeter (Excellent guide here by Jack0r for Voicemeeter Banana)
- Virtual Audio Cable (requires v.4.50 or later).




test2 [modified] - REAPER v4.78/x64 - Registered to khuon viet pham (Licensed for personal/small business use)
File Edit View Insert Item Track Options Actions Help [Delete Items]
[48kHz 24bit WAV : 8/8ch 512spls ~21/21ms ASIO]
photograph
out
photograph.wav
25.1.36 / 0:48.181
[Playing]
MASTER
photograph
out
Mixer

-
- For routing in Windows, check this excellent guide again by Jack0r. This other guide may be of interest too.

MacOS

Tested with **Reaper** and the following apps ensuring the routing

- Soundflower
- Jack server : check this excellent guide.
- Loopback
- untested: iShow U (guide here).
- untested: WavTap (guide here).

Linux

No DAW tested. Update the Wiki if you tested.

Ambisonics

Although the channel layouts are tagged with position (2.1 5.1 etc), it is possible to encode your channels for ambisonics use provided your decoding application is setup to supply the ambisonics positions according to channel order.

This is the case for Facebook Live 360 with spacial audio.
To use the latter in Settings > Audio, select : channels > 4.0 ; aac codec is mandatory (if you use Recording > Custom FFmpeg > to URL > select flv container, with rtmp URL to FB live 360 and aac audio codec + x264 video codec).
When streaming to FB Live 360 the four channel rtmp stream will be interpreted as carrying ambisonics of order 1 .

In order for this feature to work, obviously you will need an ambisonics capture device.
If you have an aac or uncompressed audio recording with 4 ambisonics, you can also play it by adding a Media Source.
(If it doesn't work, check with ffmpeg that it is decoded correctly.)

Higher Order Ambisonics

No live streaming service supports beyond first-order ambisonics.
For recording, though for order 2 or 3, you can use pkviet's fork.
If you record with libopus (Output > Advanced > Custom FFmpeg recording) up to 255 channels are available with mkv.
You will have to add the mapping_family=255 option though to FFmpeg audio encoder options in OBS-Studio.

Multilingual Streaming

Multichannel support in OBS allows to stream several languages simultaneously.
This is useful for live translation (public talks ...).

The mainstream streaming services do not support such a feature directly.
In the broadcast industry, one usually uses mpeg-ts multi-track streams instead of a multichannel stream (in single track).

There are workarounds though allowing one to use a single track multichannel audio.
Here is one requiring :

- **nginx** with rtmp module;
- **ffmpeg** scripts which will be exec'd by nginx: ffmpeg will split the audio channels and create mono rtmp streams (as many as there are languages).
- each of these mono rtmp streams can then be pushed by nginx to a service like FB or YouTube Live.

For nginx setup with rtmp module, check elsewhere. (Make sure to set it up with a single worker.)

The setup is the following for surround sound capture with each mono channel carrying a language:
(capture tested: sdi/hdmi decklink cards, reaper, Behringer X32, ASIO sound cards).

- obs with surround sound enabled
- rtmp stream to nginx which is setup with an exec script
- `rtmp {`

```
•  server {
•    listen 1935;
•    ping 30s;
•    notify_method get;
•
•    application splitter {
•      live on;
•      allow play all;
•      exec /home/me/splitter.sh $name;
•      exec_kill_signal term;
•      record off;
•   }
   }
```

   - here nginx executes a ffmpeg script which will split the channels and redirect rtmp mono streams to nginx.

FFmpeg example script (for two languages and a stereo source):

```
#!/bin/bash
echo "$(date +"%Y/%m/%d %H:%M:%S"): starting"
on_die() {
  # kill all children
  pkill -KILL -P $$
}
trap 'on_die' TERM

ffmpeg -i rtmp://IP:port/splitter/ \
       -filter_complex \
          "[0:a]pan=mono|c0=c0,aresample=async=1000[a0];
           [0:a]pan=mono|c0=c1,aresample=async=1000[a1]" \
       -map 0:v -copyts -start_at_zero -vcodec copy \
         -map [a0] -bsf:a aac_adtstoasc -copyts -start_at_zero -c:a
libfdk_aac -ab 64k -ac 1 \
         -f flv rtmp://IP:PORT/app/stream_language1 \
       -map 0:v -copyts -start_at_zero -vcodec copy \
         -map [a1] -bsf:a aac_adtstoasc -copyts -start_at_zero -c:a
libfdk_aac -ab 64k -ac 1 \
         -f flv rtmp://IP:PORT/app/stream_language2 &
wait
```

For more languages pick a corresponding channel layout and add relevant streams in FFmpeg script. You should also read the ffmpeg filtering guide. Note that for 5.1 and 7.1 one channel (the fourth) will be encoded as LFE and so is not useable. For up to 16 channels support, check the pkviet's fork.

Detailed list of surround sound features

- Recording and Streaming multichannel audio sources (surround sound).
- Compatible streaming services: Twitch, Mixer rtmp (not ftl), smashcast, FB 360 live
- Compatible protocols:
  - rtmp
  - mpeg-ts tcp udp (others untested).
- Streaming servers tested:

  - wowza,
  - nginx-rtmp
    (rtmp with multichannel audio passes through and can be distributed by these servers to another service or cdn supporting surround sound; however, the recording feature of these servers does not work; only the first two audio channels are kept).
- html5 players tested and working with live surround:
  - videojs,
  - bitmovin,
  - mediaelement,
  - viblast (hls & dash),
  - hls.js
- Compatible containers (for recordings):
  - mkv
  - mp4
  - ts
  - flv (others untested).
- Compatible codecs:
  - ffmpeg aac (native encoder, up to 16 channels),
  - libfdk_aac (up to 7.1),
  - core audio aac (up to 7.1),
  - opus (libopus encoder, up to 255 channels),
  - vorbis (up to 7.1),
  - pcm (others untested).
- OS: compatible with win, macOS, linux (alsa, pulse-audio).
- Misc:
  - Higher audio bitrates (up to 1024 kbs) unlocked to accomodate higher number of channels.
  - audio monitoring, audio filters, VST are all working OOB.

Understanding The Mixer

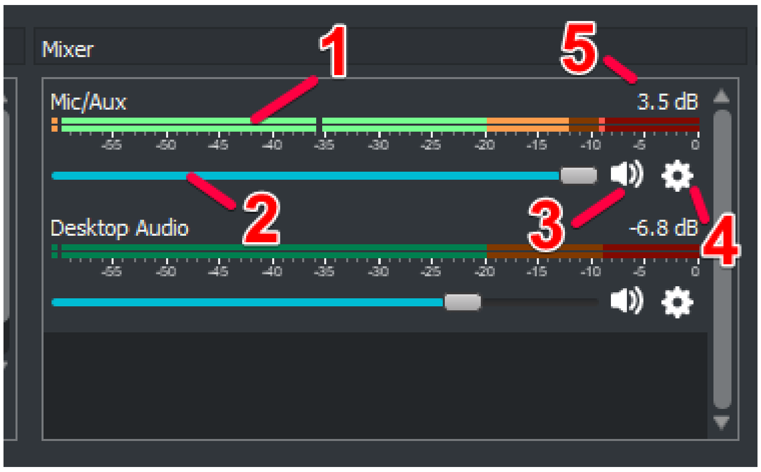
This mixer is available in OBS Studio 21.0 or newer.

Table of Contents:

General Overview

***Clipping** refers to the actual **distortion** that is heard when an audio signal cannot be accurately reproduced by the gear through which it is passing.*

Each entry in the mixer is made up of 5 parts



1. **The (Volume) Meter** - the coloured bars that light up as audio goes through a source
2. **The Fader** - the volume slider for that source
3. **Mute button** - a speaker icon to mute the source without adjusting the fader

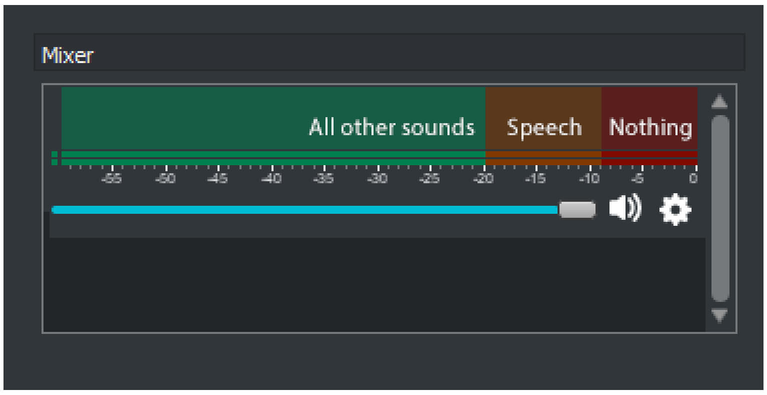
4. **Options button** - provides extra settings for the source
5. **Volume level** - The exact value adjusted by the fader or volume % number, measured in decibels

Most of the time, you want to keep a close eye on the meters. The rest should be 'set once run forever'.

Reading the Volume Meter

Zones

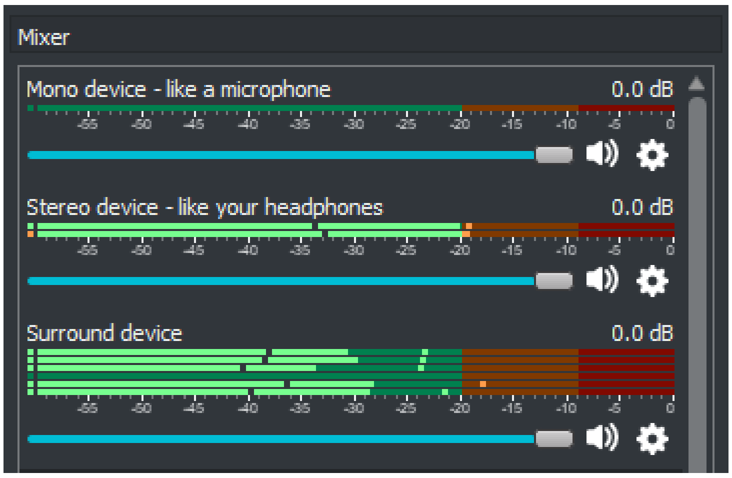
The volume meter is made up of 3 primary sections signified by the green, yellow and red zones.



- **Red Zone** - this area should be avoided as it can cause clipping which sounds unpleasant
- **Yellow Zone** - Speech (your own & friends') should always stay in here, occasionally touching the red zone
- **Green Zone** - Everything else including music, game audio and alert sound effects should remain here
  1) Even if other sounds look to be the same volume as your voice, they may in reality sound louder to viewers

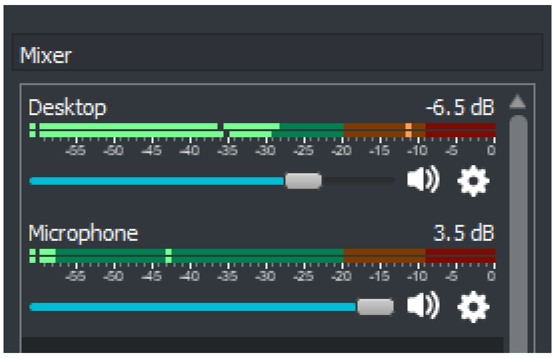
Channels

Each audio source will have at least one volume meter assigned to it.


Mixer
Mono device - like a microphone
0.0 dB
Stereo device - like your headphones
0.0 dB
Surround device
0.0 dB

- **(1) Mono source** - viewers will automatically hear this in both left and right channels (headphones/speakers)
- **(2) Stereo source** - left is shown first, right second. Viewers will only hear these as they're assigned.
  1) If only the first meter lights up, enable "Mixdown to Mono" in the Advanced Audio Properties, otherwise your viewers will only hear that source in their left channel
- **(3 or more) Surround source** - when Settings->Audio->Channels is set to Stereo (default), you won't see surround channels
  1) OBS automatically mixes down surround sources to Stereo unless otherwise specified
  2) Ordered as Front Left, Front Right, Front Center, LFE/Sub, Rear Left (5.1), Rear Right (5.1), Side Left (7.1), Side Right (7.1)

Indicators



1. Left Dot (static): **Input level** - Live indicator of the volume meter's status (green/yellow/red)
2. Black Dot (always moving): **Vu-meter** - Shows 'sound pressure', a more accurate indicator of 'loudness'
3. Main Line (always moving): **Peak Programme Meter** - Has a 'fall-off decay'. After sound stops the bar will slowly go down rather than displaying completely live data, until it reaches -60 dB (empty) or receives new, louder data
   - The 'Audio Meter Decay Rate' (how quickly the bar goes down) can be changed in the Audio screen of Settings
4. Right Dot (sometimes static): **Peak** - Displays the loudest the meter has been in 20 seconds, a great way to see if you're clipping

Setting Audio Levels

There are a number of ways to configure an audio source's volume.

As you adjust the volume at each step of the process, listen to the device both as early as possible (some devices will have a 'headphone' output or a 'monitor' jack) and again when it hits OBS using Audio Monitoring via Edit->Advanced Audio Properties.

- **Always start at the device in question.**
  1) **Microphones:** check if it has a physical knob (usually labeled "Gain")
  2) **Games/consoles:** most game applications and gaming consoles have their own volume sliders usually hidden in a Settings/Options screen
  3) **Physical mixers:** these have individual gain knobs for their own sources, and a 'Master' before it's sent to your computer & OBS
  4) **For other devices** including audio interfaces, check their user manuals to see how to configure them, as they may require third party software by the manufacturer.
- **Your Operating System** (Windows, macOS, Linux, etc) also has its own volume sliders and mixer
  1) Note that some devices may have a 'safe zone' that is well below the 100% mark.
    - In Windows for example, Control Panel->Hardware and Sound->Sound, under the Recording tab, select the device in question (like your mic) and click Properties. In the Levels tab, right click the % value and choose 'decibels'. You want this to be at (or around) 0.0dB for the least amount of clipping. It will retain the setting when you switch back to 'percent'.
      2) Your system's primary volume slider **does not** affect the volume of the sound that OBS hears
      3) Individual application volume sliders **do** affect the volume of the sound that OBS hears

Finally, be sure to record a session as you normally would and then listen back to the file before going live.

- If you still feel the audio needs to be tweaked, now is the time to adjust the Fader within OBS.
  1) If you need something higher than the maximum, you can insert custom percentage (%) values that can go well above 100% using Edit->Advanced Audio Properties.

Other things to keep in mind:

- Microphones will naturally be quieter than anything computer generated – be it games, music or general sound effects. Take that into account when balancing your audio.
- Voice communication software like TeamSpeak, Discord and Mumble allow you to 'boost' the voice volume of other members (both overall and individual users) above the 100% mark if needed.
- Most games do not allow per-user volume control so keep this in mind when playing in a matchmaking system.

Technical Details

decibel relative to Full Scale (dBFS)

Audio is measured in decibel (dB), which is a logarithmic scale which closely resembles how our ears and brain perceive audio volume.

dB is a **relative** measurement. We could put the value 0 dB anywhere on the meter and be correct. In digital audio it is the convention to use 0 **dBFS** (the FS suffix is to indicate this convention) as the maximum volume that the sound card, audio interface, DA/AD-converter can handle. Lower volume levels are shown with negative dBFS values.

OBS internally uses floating point calculations for audio processing, so it does have the ability to process audio that is louder than 0 dBFS. However in the end when OBS records or streams the video, the audio will **need be below 0 dBFS** or the viewer will hear a nasty distortion called clipping (described at this start of this guide).

The Peak Programme Meter (PPM)

The PPM is the main visual feature on the OBS meter, lighting up as part of the overall volume meter.
The length of the bar indicates the peak volume level of what the viewer will hear when playing your video or stream. You can change the volume by moving the fader which is directly below.

The meter is split into three different coloured sections. At -20 dBFS we have the Alignment Level (AL) and at -9 dBFS we have the Permitted Maximum Level (PML).

Traditionally a sinus tone is played through the whole system and gain is configured for each piece of equipment to show the tone to be at exactly the **-20 dBFS Alignment Level (AL)**. This way the level on your mixing desk will match that of OBS. This is important if you have a lot of audio equipment in a chain. **The value for the alignment level was chosen to be near the average audio level for speech.**

The Permitted Maximum Level (PML) is the level where if you go above this value there is a small but potential chance that the sound will get clipped before it reaches the viewer. -9 dBFS was chosen because:

- With most PPMs, including the one in OBS, the audio may have a peak level 3 dB higher than what is read from the meter.
- When reading PPMs it is difficult to see the actual peak, so another 3 dB margin is added. This problem is eliminated in OBS due to the peak-hold feature which makes reading accurate.
- Alignment errors, with multiple pieces of equipment in the chain an extra 3 dB margin was added for difference is levels.

OBS currently implements a "Sample peak program meter (SPPM)", in the future it would be preferred to replace it with a proper "4x Over-sampling peak programme meter", which would make it more accurate for measuring maximum peaks.

Peak and Hold

There is a small line that moves right with the PPM meter, but then stays there when the PPM moves left again. It will stay there for 20 seconds before returning back to where the PPM meter currently is. This allows you to easily check what the maximum level was after you accidentally made a loud noise.

VU-meter

A second small line on the meter, black and inside the bar of the PPM meter, is a VU-meter. This meter was traditionally used to determine loudness, because it was cheap to manufacture, and in OBS easy to implement.

It measure the root-mean-square, integrated over a period of 300 ms. Due to the calculation it shows more closely the sound pressure levels than does a peak-meter.

This meter is less useful, but it kept some structure of the code intact so we can replace it with a proper loudness meter based on ITU-R BS.1770-2.

Input Level

The input level are the small square indicator at the far left of the meter. This is the best place to see if the audio is too loud for the audio interface that captures your microphone.

The colours have the following meters:

- dark green: the input level is less than -50 dBFS
- light green: the input level is between -50 and -20 dBFS
- yellow: the input level is between -20 and -9 dBFS
- red: the input level is between -9 dBFS and -0.5 dBFS
- white: the input level is larger than -0.5 dBFS

If the indicator is missing it means there is no audio streaming toward OBS. This may be simply due to no audio being available yet, waiting for the user to start playback of an audio file. Or it may indicate a problem like the audio interface having been disconnected.

The input level meter is before the volume fader, but it is behind any filters that are in use by a source. To correctly determine input level you will need to disable the filters for this source.

Streaming With SRT Protocol

This feature requires OBS Studio 25.0 or newer.

Table of Contents:

General Overview

- Secure Reliable Transport (or SRT in short) is a relatively recent open source streaming protocol, originally developped by Haivision (first demo in 2013) and promoted by the SRT Alliance which includes many big players in the streaming/video/telco industry. It promises:
  (1) better resilience to network issues (jitter, lost packets, delay, bandwidth fluctuations) with mechanisms for packet recovery (retransmission or ARQ automatic repeat request; and also FEC forward error correction) + internet bonding and
  (2) low latency (as low as twice the round-trip between encoder and ingest server, with sub-second latency usually).

For a video demo, see here:

- SRT is mostly used in the broadcast and corporate world at the moment. See the NAB 2018 SRT panel with ESPN, NFL, Microsoft speakers talking about their use of SRT:

  The NAB 2019 NAB SRT panel can be watched here: https://www.haivision.com/resources/webinars/broadcast-panel. (Hey Haivision, friendly suggestion: it'd be nice to have this vid posted on YouTube instead of having to enter personal info! :P)

- Unlike RTMP, SRT is an open source protocol, and the source code can be found on GitHub. While RTMP development has been abandoned since 2012, SRT development is still very much active.

- As a protocol, it is content agnostic, although the industry uses it along with an MPEGTS container, which is the de facto standard in broadcast industry. (In terms of comparison, RTMP protocol relies on the FLV container.) The MPEGTS container is usually used along with UDP protocol, which makes it fast, but very unreliable and prone to packet loss. It can also be used with TCP, which is more reliable but has larger latency. SRT adds up to these two protocols to transport MPEGTS, with the best of two worlds: the reliability of TCP, and the lower latency of UDP. It also supports encryption and bonding.
- Other competing new protocols are WebRTC, Zixi (closed source) and RIST; the latter two are quite similar to SRT and all go beyond RTMP.
- For further technical details, we recommend this video by Alex Converse, a Twitch engineer:

There is also a white paper which can be found here
or there. The API is fully documented on GitHub.
A very good source of info is the SRT Cookbook.

Can SRT be used with Twitch or my favorite service?

Short answer: **NO** (or not yet?)

Services

Long answer: **None** of the main streaming services support the SRT protocol for ingest. Most still use RTMP (Twitch, YouTube, Facebook...). (Mixer though relies on WebRTC through its proprietary FTL protocol which OBS already supports). If you're using exclusively these services, no need to read further.

At this stage of the adoption of SRT protocol, you'll have to be technically inclined if you want to use SRT. If you are able to set up your own streaming server, maybe redirecting your streams to the main services like Twitch or YouTube and are interested in achieving low-latency with improved network resilience, read on.

The other category of users who could potentially be interested belong obviously to the professional broadcast industry. This wiki entry can be considered as fairly advanced in that it requires access to a server and being able to set it up.

The configuration of OBS itself ranges from easy to medium in terms of difficulty. The server setup is more challenging since it requires system/network admin knowledge.

Encoders

Live software encoders:

- **FFmpeg**,
- **OBS Studio** which relies on the FFmpeg libraries,
- vMix,
- srt-live-transmit (which is a demo app from libsrt developpers; needs to be compiled from source)
- Larix Broadcaster and Larix Screencaster can broadcast SRT from Android and iOS. Larix Talkback feature allows receiving SRT audio return feed back to mobile devices.
- CameraFi Live (Android only).

Hardware encoders are also available at various vendors.

Servers

The following servers support SRT ingest:

- Wowza (paid service)
    - Supports SRT ingest and transmuxing/distributing in RTMP/HLS/DASH
- Nimble Streamer (free, closed source)
    - While Nimble Streamer is nominally free, it is used along with a non free dashboard which is, in all fairness, quite convenient. But it can also be used without the dashboard and just requires modifying a JSON config file.
    - Supports SRT ingest and transmuxing/distributing to RTMP, SRT, NDI, MPEG-DASH, HLS, Low Latency HLS and more.
- SRT Live Server (free, open source)
    - This option only *serves* SRT streams and does **not** transmux to HLS/RTMP/DASH. It is much more rudimentary than the other servers at this stage but it is FOSS and works fine with OBS Studio in our tests.

Additionally, though it is technically not a server, FFmpeg can be used in listener mode to ingest an SRT stream. It won't be able to serve the stream as a real genuine server would do. But it could be used to transmux to RTMP and route to nginx-rtmp for instance, which can then handle the ingest to Twitch/YouTube/Facebook/etc.
ex: `ffmpeg -i srt://IP:port?mode=listener -c copy -f flv rtmp://IP:1935/app/streamName`.
In the same way srt-live-transmit can be used to listen to an srt (or udp) stream and relay to a final srt URL.
ex: `srt-live-transmit srt://IPsrc:port srt://IPdest:port`.

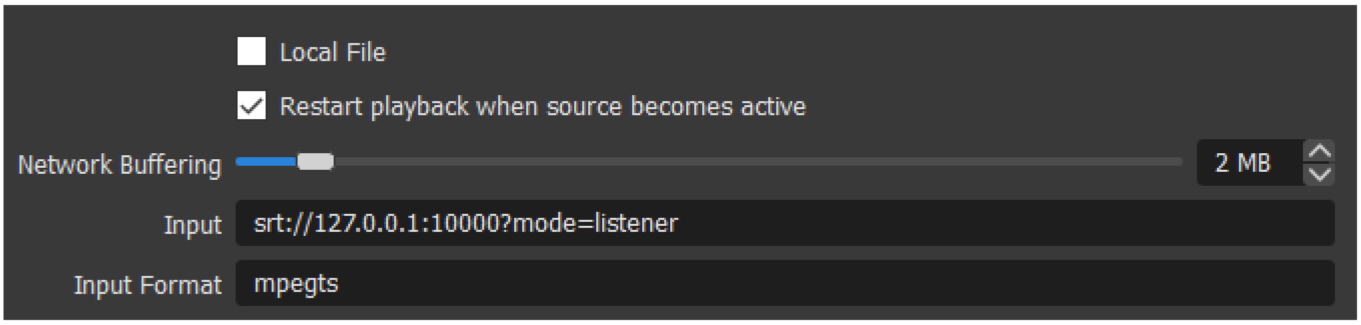
Players

The following players can be used to watch an SRT stream :

- VLC (version 3.0+ on Mac/Linux, version 4.0+ on Windows)
- ffplay (part of ffmpeg tools)
- OBS Studio Media Source (an obvious case-use is to broadcast from any SRT source to an OBS instance).
- Haivision Play Pro - iPhone and Android.
- Larix Player for Android, iOS, Android TV and tvOS.

Receive srt stream within OBS

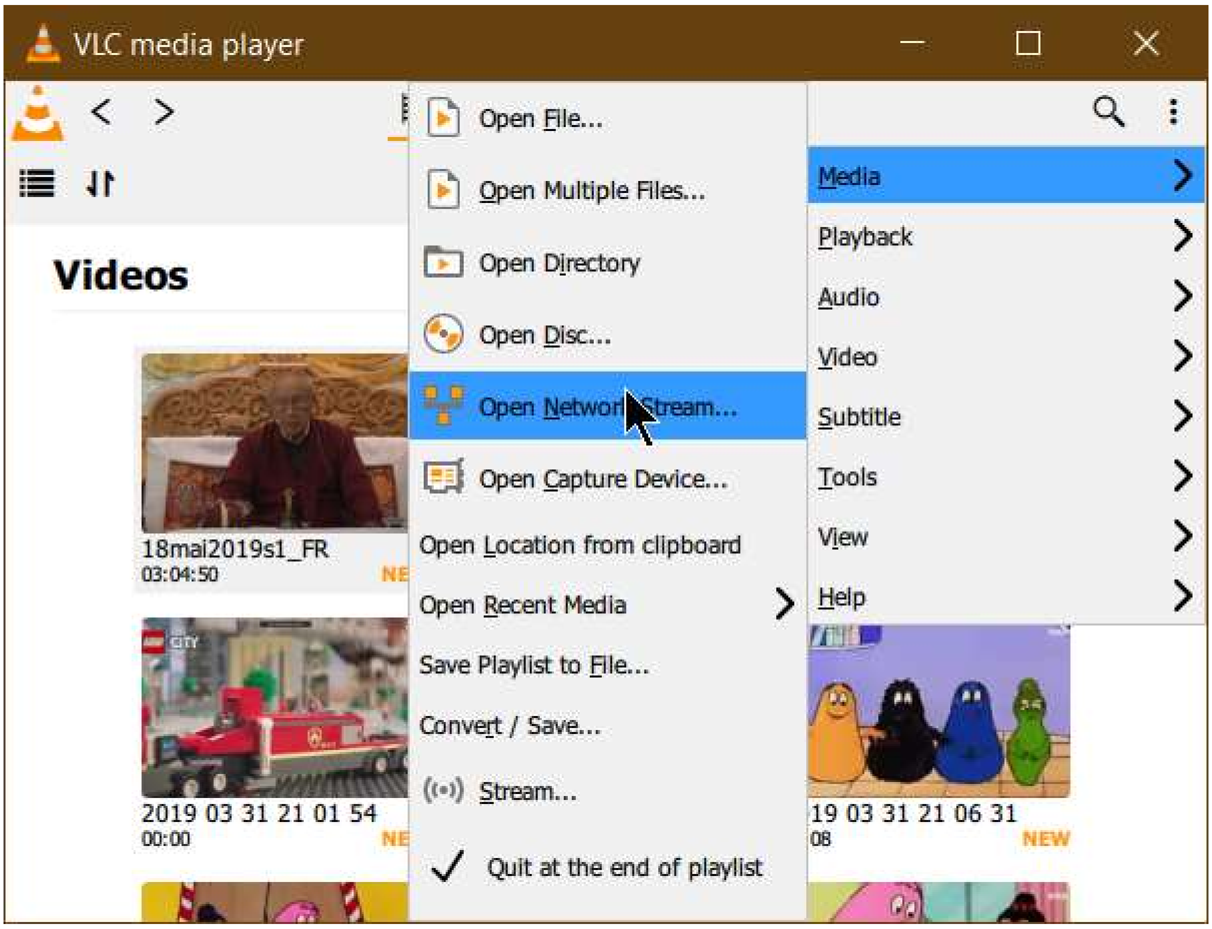
This could be useful to two pc setups (although NDI is probably a more common solution).
In a Media Source, uncheck 'Local File'.
For 'Input', enter the srt URL. If the stream is received from a server (in listemer mode), the srt connexion will be in mode=caller (which is the default one so the option can be omitted).
If however the stream is received straight from an encoder in caller mode, add the mode=listener to the URL (see screenshot).
For 'Input Format', enter mpegts.


☐ Local File
☑ Restart playback when source becomes active
Network Buffering: 2 MB
Input: srt://127.0.0.1:10000?mode=listener
Input Format: mpegts

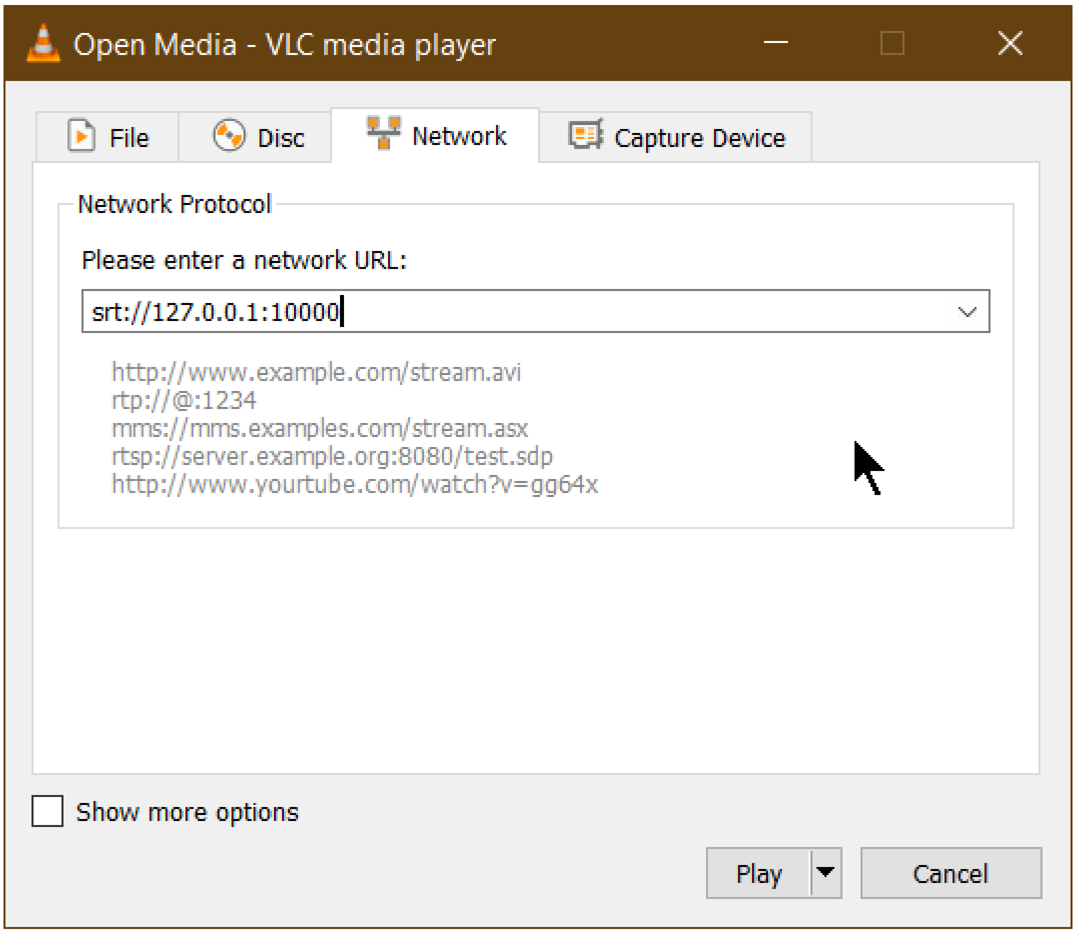
VLC usage

Download VLC 3.0 here or VLC 4.0 here (warning: this is the development version of VLC).
If you just want to test without disturbing your current VLC install, we advise you to download a portable install (zip).

Go to `Media > Open Network Stream`:



enter the SRT IP which has the form `srt://IP:port`.


Open Media - VLC media player
File
Disc
Network
Capture Device
Network Protocol
Please enter a network URL:

srt://127.0.0.1:10000

http://www.example.com/stream.avi
rtp://@:1234
mms://mms.examples.com/stream.asx
rtsp://server.example.org:8080/test.sdp
http://www.yourtube.com/watch?v=gg64x
Show more options
Play
Cancel

ffplay usage

It is required that ffplay be compiled with libsrt support. To the extent of our knowledge, there does not seem to be any such binary widely available, although there are no license constraints.

Just launch the command-line:
```
ffplay srt://IP:port
```

How to set up OBS Studio

There are two ways of setting up OBS Studio to connect to a server. The first is simpler but gives less options at the moment. The second is a bit more difficult to setup but gives more fine tuning capabilities (muxer options for instance are available). Note: ffmpeg may not come with SRT support in older distributions of Linux, so check the repository sources to ensure that ffmpeg comes with libsrt support, as there is no easy way of getting OBS Studio to reference a custom build of ffmpeg.

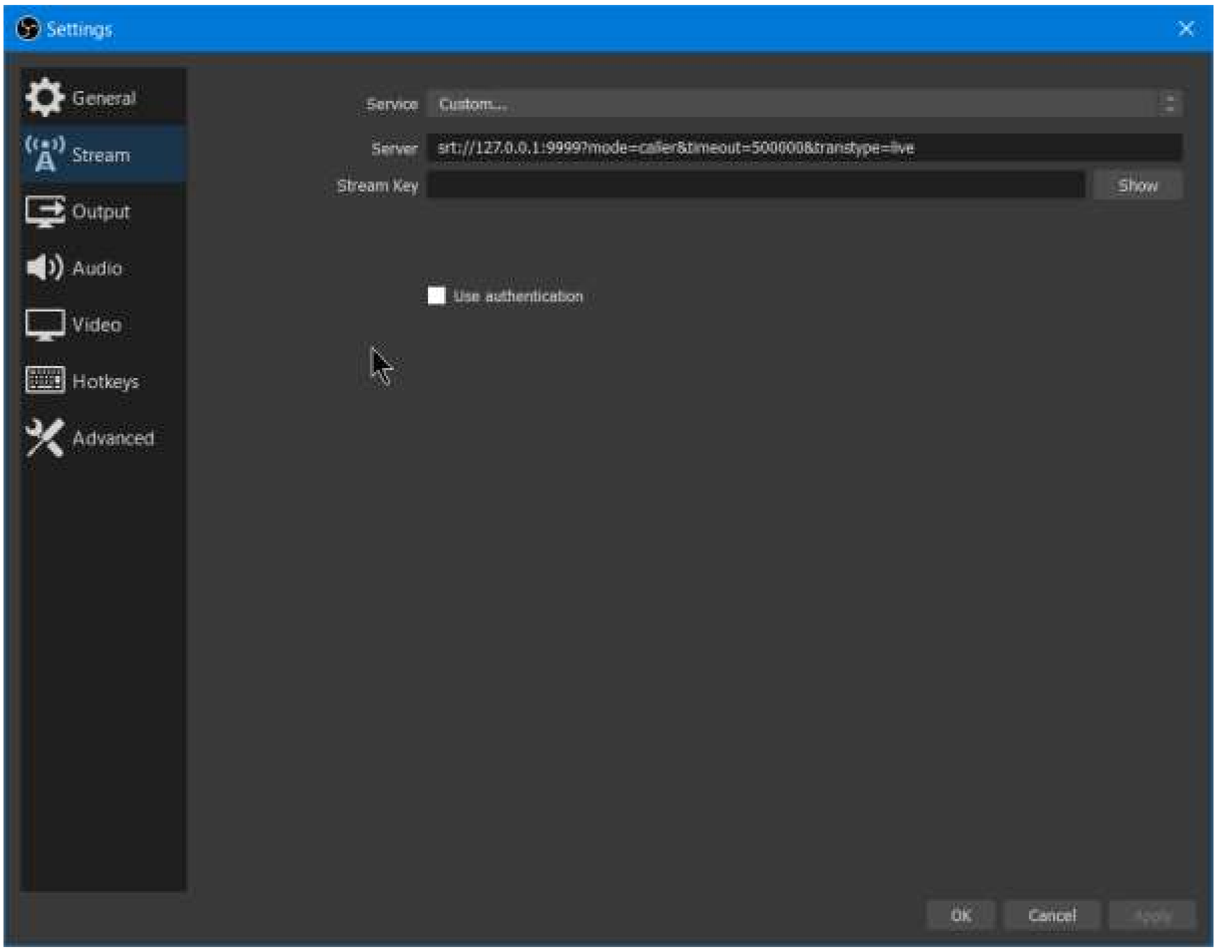
Note that while the discussion focuses on SRT protocol, UDP or TCP can also be used instead.

Option 1: Stream SRT using the Streaming output

Credit: Aaron Boxer, Collabora (SRT Alliance) author of the new SRT output. Output rewritten from scratch by pkv (pkv@obsproject.com) to solve some bugs.

Works fine with: Wowza, Nimble server, vlc, ffmpeg, srt live server, Makito X decoder. (those are only the platforms we tested at obsproject; most should work fine.)

1. Go to `Settings > Stream`
2. In the `Service` drowdown, select `Custom`.
3. Enter the SRT URL in the form: `srt://IP:port` (OBS Studio will also accept any protocol relying on MPEGTS container and supported by FFmpeg, therefore UDP, TCP, RTP, etc.)
4. Don't enter anything for the key. It's not used.



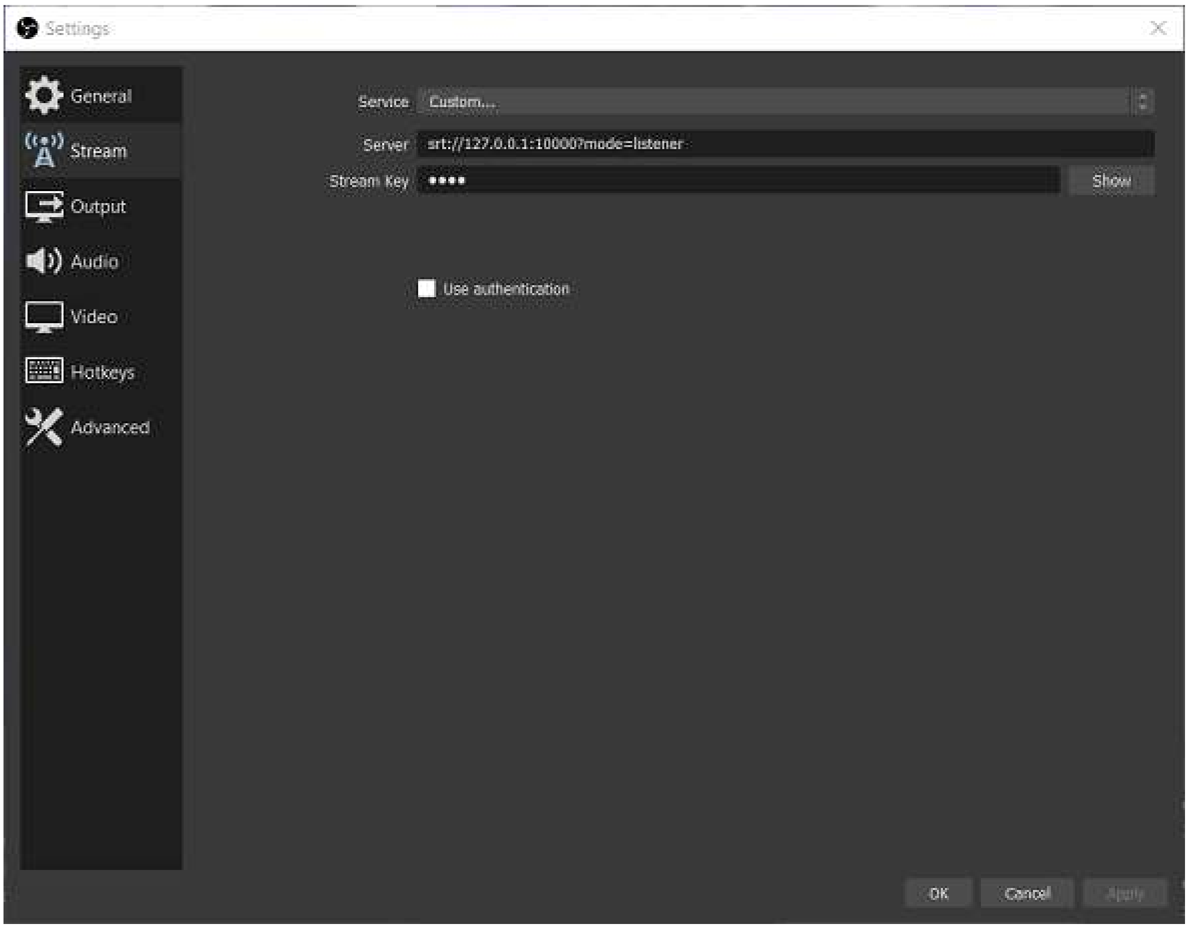
Service: Custom...
Server: srt://127.0.0.1:9999?mode=caller&timeout=500000&transtype=live
Stream Key:

OBS Studio will accept options in the syntax: `srt://IP:port?option1=value1&option2=value2`. The full list of options is those supported by FFmpeg: http://ffmpeg.org/ffmpeg-protocols.html#srt.

The most important option is **latency** in microseconds (µs). It has a default value of 120 ms = 120 000 µs and should be at least 2.5 * (the round-trip time between encoder and ingest server, in ms).
Ex: for a latency of 1 sec, set latency=1000000 .

Another sometimes required option is the **mode**, which can be `caller`, `listener` or `rendez-vous`. `caller` opens client connection. `listener` starts server to listen for incoming connections. `rendezvous` use Rendez-Vous connection mode which is a bi-directional link where the first to initiate handshake is considered caller. The default value is caller and usually need not be set for OBS Studio since it'll be in caller mode normally.

A case where it's useful to set the mode to `listener` is when sending a stream to VLC. OBS Studio then acts as a server to VLC, which is the client. On a LAN for instance, set OBS Studio to `srt://127.0.0.1:port?mode=listener` to establish a connection to VLC which you point to `srt://127.0.0.1:port`.


Service: Custom...
Server: srt://127.0.0.1:10000?mode=listener
Stream Key: ••••
Use authentication

Known issues:

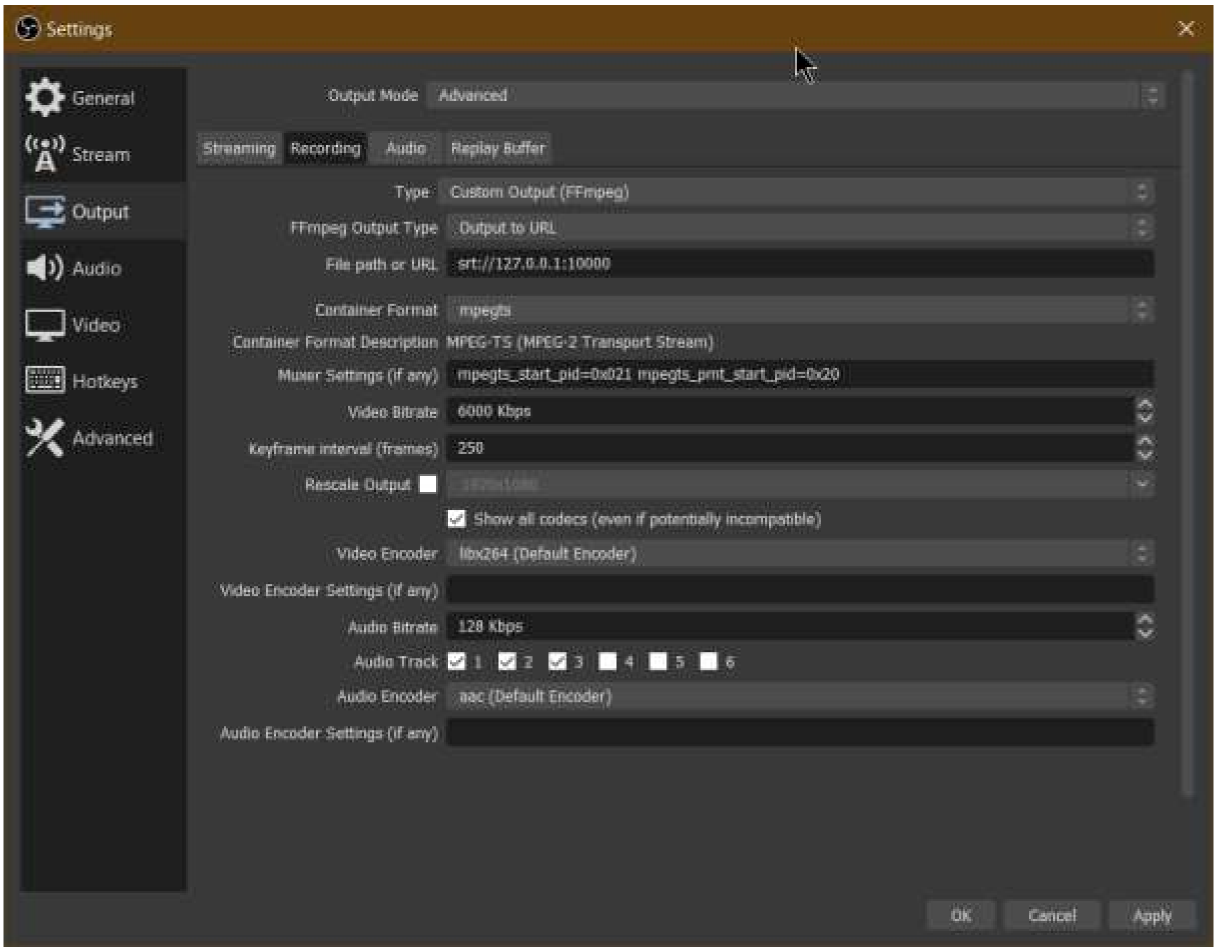
- With v25, a bug with MPEGTS muxer makes the stream not completely compliant with MPEGTS spec. In this case, SRT decoding fails when transmuxing SRT to another protocol/container than the combination SRT/MPEGTS. This is the case for instance with Nimble Streamer and Makito X Decoder. A dump of the MPEGTS stream therefore creates a non-conformant file (this can probably be fixed though by a remuxing with FFmpeg).
  - bug fixed in obs v26 in this PR: PR 2665. Use obs version >= 26.

Option 2: Stream SRT with the Custom FFmpeg Record output

This option is a bit more complicated. It relies on the `Advanced: Custom FFmpeg Recording` output.
Works fine with: Wowza, Nimble server, vlc, ffmpeg, srt live server (those are only the platforms we tested at obsproject; most should work fine). Doesn't work well with Makito X decoder from Haivision.

1. Go to `Settings > Output`
2. In the `Output mode` dropdown, select `Advanced`.
3. Click on `Recording Tab`.
4. In the `Type` dropdown, select `Custom Output (FFmpeg)`
5. In the `FFmpeg Output Type` dropdown, select `Output to URL`
6. In the `File Path or URL` box, type the SRT URL: `srt://IP:port` (options like latency are entered with the syntax `srt://IP:port?option1=value1&option2=value2`).
   For a list of some of these options and a discussion, see the previous section or refer to FFmpeg documentation : http://ffmpeg.org/ffmpeg-protocols.html#srt.
7. In `Container Format` dropdown, select `mpegts`.
8. `Muxer Settings` can be left blank, or you can use the MPEGTS FFmpeg muxer options with the following syntax `option1=value1 option2=value2` (the `option=value` pairs must be separated by a space).
9. The other settings are self-explanatory. Check the box `show all codecs` to display all codecs available to FFmpeg.

Note that several audio tracks can be selected. They can be identified on the ingest server side by what is called a PID. On default value, the video track has `pid 0x100` (=256), and the other audio tracks have `pid 0x101` etc. If you need to change the pid of your tracks, use the muxer option MPEGTS_start_pid in 7.



Output Mode: Advanced

Type: Custom Output (FFmpeg)
FFmpeg Output Type: Output to URL
File path or URL: srt://127.0.0.1:10000
Container Format: mpegts
Container Format Description: MPEG-TS (MPEG-2 Transport Stream)
Muxer Settings (if any): mpegts_start_pid=0x021 mpegts_pmt_start_pid=0x20
Video Bitrate: 6000 Kbps
Keyframe interval (frames): 250
Rescale Output: ☐ 1920x1080
☑ Show all codecs (even if potentially incompatible)
Video Encoder: libx264 (Default Encoder)
Video Encoder Settings (if any):
Audio Bitrate: 128 Kbps
Audio Track: ☑ 1 ☑ 2 ☑ 3 ☐ 4 ☐ 5 ☐ 6
Audio Encoder: aac (Default Encoder)
Audio Encoder Settings (if any):

Known issues:
There can be issues with Makito X Decoder. (under investigation) Use Option 1 for Makito X decoder.

Pros/Cons in comparison with Option 1:

Pros:

- MPEGTS muxer options can be customized (ex: set pid for video and audio tracks).
- Several audio tracks can be streamed (for instance, track 1: main track, track 2: background music, track 3: commentary etc.) while in option 1 only a single track can be selected. OBS Studio supports up to 6 audio tracks.

Cons:

- More complex to set up.
- One can not record the stream since one leverages the Record Output to stream.

Examples of setups

Relay server to Twitch

One can use ffmpeg to easily relay an input SRT stream to a standard RTMP compatible with Twitch (or even other streaming services provider like YouTube ou Facebook). This is especially interesting if you have a bad and unstable connection between OBS and the service. Note that the server you use should have a fast and reliable connection to benefit from this (but this is usually the case). Also note that using a server to proxy stream consumes a lot of bandwidth (consider twice as much as the bandwidth of your video stream) and this can be expensive on some providers like Amazon, GCS or Digital Ocean (it doesn't need a fast CPU though, it's only receiving, rewraping for RTMP and sending).
This approach has the advantage of being really easy to setup on OBS's side if you prepare a server for someone else.
For example the following command can be used on the server:

```
ffmpeg -i
srt://:1234?mode=listener&transtype=live&latency=3000000&ffs=128000&rcvbuf=
100058624 -c copy -f flv rtmp://live-cdg.twitch.tv/app/streamKey
```

In this command you can change:

- `1234`: This is the listening port used by the SRT server. You will need to send your stream to this port. It can be anything, provided the ffmpeg can bind to it.
- `latency=3000000`: The latency in microseconds. Here we use 3000 milliseconds (3 seconds). This is a simple calculus: the higher it is, the more reliable it will be if you drop packets or your latency increases and your viewer will have to wait more to receive your frames. The lower it is, the more sensible it will be to connection problems, but your viewers will also receive your content faster.
- `ffs` and `rcvbuf` are complicated numbers that indicates the sizes of the differents buffers. They are already set for a latency of 3000ms. Basically, when you increase the latency, you'll need to increase these values (because SRT needs to store more content in memory to recompose the puzzle afterwards). You can have informations on how to calculate these here: https://github.com/Haivision/srt/issues/703#issuecomment-495570496
- `live-cdg.twitch.tv`: Nearest Twitch's ingest server. Change to your nearest one found here: Twitch Ingest Informations
- `streamKey`: This is your Twitch.tv stream key. You will need it to replace this with yours so they know who is streaming to where. Do not leak it, anyone with this key can stream to your account.

On OBS, you just need to configure your stream settings to "Custom", and specify your server with this template:
`srt://ipofyourserver:4444`
You can leave the stream key empty, you don't need it.

Run the ffmpeg command you saw above, it will wait indefinitely for an input stream, and automatically stop at the end of the stream. I leave you the choice of managing the restart with systemd or in a Docker container!