fiedler audio

Dolby Atmos COMPOSER

Manual





Table of Contents

- 1. What is Dolby Atmos?
- 2. What is the Dolby Atmos Composer?
- 3. Overview

Monitoring page

Input Configuration page

Master Channel page

Options page

4. Setup & Routing

Session settings

Placing Composer and Beam in your session

Proper routing

Audio processing issues on some DAWs

Latency compensation

Enforce 7.1.2 bed option

Managing connections

Managing the connections list

- 5. Monitoring
- 6. Using the Dolby Atmos Beam

Beam input

Beam output

Beam control column

<u>3D Panner</u>

Top Panner

Side Panner

Operating the Panner

- 7. Panning and reverb with Spacelab
- 8. Input configuration & program level metadata
- 9. Mix processing with the Master Channel
- 10. Loudness measurement
- 11. Export
- 12. Importing and editing ADM/BWF files
- 13. <u>Manual latency compensation</u>
- 14. Additional information
- 15. Video Tutorials
- 16. Trial & Purchasing
- 17. Activating & Moving your licenses
- 18. DAW-specific settings & recommendations
- 19. Modifier keys
- 20. Support
- 21. <u>Installation & deinstallation</u>
- 22. <u>Acknowledgements</u>
- 23. About fiedler audio

1. What is Dolby Atmos

Dolby Atmos is a so-called object based audio format and it is designed for creating three-dimensional immersive audio mixes. Object based means that audio is not present in form of channels with a predefined position in space, like for example stereo, but in form of objects which can move around in space over time, among other things.

This also means that object based audio is not rendered to it's final playback format during production but on the playback side. So Dolby Atmos is delivered to the listeners agnostic of the format they listen to and only the playback device will then convert this Dolby Atmos stream or file to the actual listening format, be it a multichannel speaker setup, a smart speaker system or headphones.

So the idea behind Atmos is that you only have to create one mix and the playback system will render that mix in such a way that it sounds great on any reproduction system. There is no need to create a separate mix for each one of these different playback scenarios.

This is done by having metadata for the discrete channels (e.g. objects) encoded into the Dolby Atmos file and having the playback system mix those channels in the best way for each playback scenario. Since the playback system creates a mix based on your metadata, object based formats tend to be quite future proof and will even work well on playback systems which have not yet been invented.

Dolby Atmos can have up to 128 of such audio channels/objects, each encoded with its own metadata containing all the necessary information for playback systems to properly play back your content. At its core, Dolby Atmos has two kinds of channels: "bed channels" and "dynamic objects". Think of the bed as virtual speaker layout where you pan and place some of your tracks in your session. In Dolby Atmos, the standard bed format is 7.1.2, which means you have 7 speakers around you on the horizontal plane, one LFE channel for Low Frequency Effects, and two height speakers above you.

In addition to the bed channels, Dolby Atmos also has dynamic objects. This type of channel is designed to have the ability to change it's position over time and therefore it is treated differently during playback. Essentially, the playback system gives extra attention to these channels to make sure they are faithfully reproduced in space regardless of the playback system.

2. What is the Dolby Atmos Composer?

The Dolby Atmos Composer is currently the only and complete plug-in solution for producing Dolby Atmos content, on any DAW, on Mac and on Windows. It is fully certified and approved by Dolby Labs and the Dolby Atmos mix can be exported as a legit Dolby Atmos ADM/BWF file. You can either produce for Dolby Atmos right from the start or take an existing mix and expand it with a Dolby Atmos version without actually changing the original.

Also the Dolby Atmos Composer offers features for your Dolby Atmos mixing workflow you will not find in any other Atmos solution. The most prominent among them are the following:

- Instead of using just beds which are limited to a maximum layout of 7.1.2, the Dolby Atmos Composer expands this concept by introducing Composites. A Composite is technically a combination of bed channels and dynamic objects which make layouts possible which are beyond the limitation of Dolby Atmos beds, such as for example 9.1.6.
- The Master Channel allows you to load OBAM plug-ins to process the entire Dolby Atmos mix, just like you would process your stereo mix by putting plug-ins on the master track.
 OBAM is our new plug-in format for immersive audio.
- Deep integration of our immersive reverb plug-in Spacelab, allowing you to have 3D reverberation on any DAW with just 2 clicks.

The Dolby Atmos Composer comes as two plug-ins. The Dolby Atmos Composer plug-in is the centerpiece and usually sits on the master track. It receives audio and panning data from the second plug-in, the Dolby Atmos Beam. The Beam plug-in can be inserted anywhere in your mixing session giving you full freedom to decide what goes into your Dolby Atmos mix.

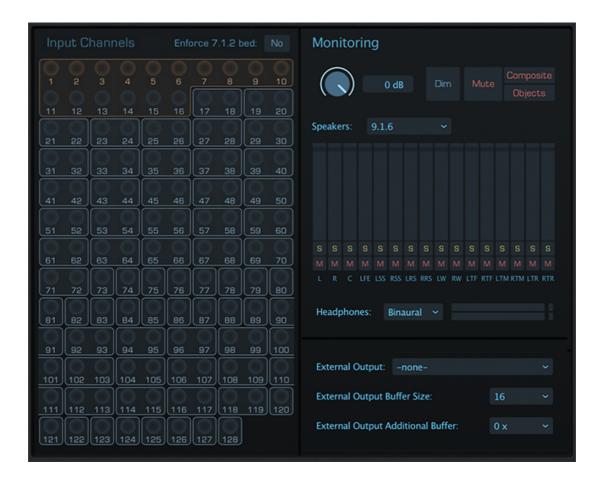
3. Overview

The editor of the Dolby Atmos Composer is split into two parts.



On the left side you'll find the list of incoming connections. Those can be Dolby Atmos Beam plug-ins or Spacelab plug-ins. Below this list are the four buttons giving you access to different pages on the right, the Monitoring section, the Input Configuration, the Master Channel and the Options page.

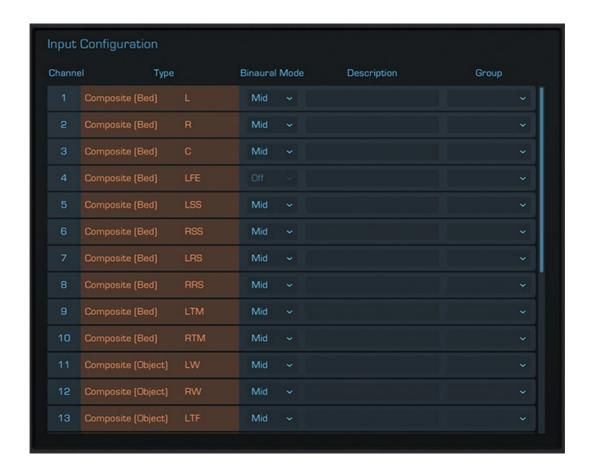
Monitoring page



The first of the four pages on the right side is the monitoring page. It is split into two parts. To the left are the meters for all possible 128 channels of your Dolby Atmos mix. The orange numbered channels are bed channels, the blue numbered channels are dynamic objects. In the image above you see that channels 1 to 16 are grouped with an orange border. That means those channels are the composite, which technically can consist of bed channels and dynamic objects. The channels / channel groups with a blue border are incoming connections configured as dynamic objects.

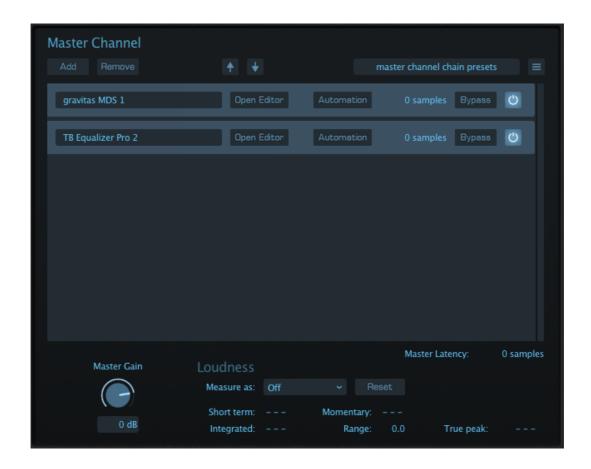
On the right side of the monitoring page you see the controls for monitoring level, Dim and Mute. You can select the speaker layout and the headphone layout as well as a personalized HRTF file and an external audio device.

Input Configuration page



The second page is the Input Configuration list containing all channels in your Dolby Atmos mix. Composite channels are orange and dynamic objects are blue. On this page you can adjust various channel related parameters, such as Binaural Mode, Description and Group assignment.

Master Channel page



The third page is the Master Channel where you can load OBAM plug-ins to process your Dolby Atmos mix as a whole. Below the plug-in list is the Master Gain knob and the Loudness measurement section.

Options page



The last of the four pages is the Options page where you can adjust global settings for your Dolby Atmos mix, the so called Program Level Metadata. Here you can also manage Groups, set the desired file formats for exporting your mix and import an ADM/BWF file.

4. Setup and routing

Session settings

The session setup for the Dolby Atmos Composer is really easy and the process is pretty much the same on every DAW. First of all you need to decide whether you want to produce in 48kHz or in 96kHz. Other sample rates are currently not supported by Dolby Atmos. The preferred buffer size for mixing in Dolby Atmos is 512 samples at 48 kHz and 1024 samples at 96 kHz. But you can use buffer sizes of up to 4096 samples in case you need it.

Placing Composer and Beam in your session

The Composer plug-in usually sits on the master track since it is the last element in a Dolby Atmos mix.

The Composer plug-in ignores all incoming audio and doesn't let anything go through which is why you won't hear anything when instantiating it on the master track of an already existing mix. That is because anything which you want to be part of your Atmos mix comes in through direct connections from either the Dolby Atmos Beam or Spacelab.

These direct connections circumvent the mixing engine of your DAW because even if the DAW is multichannel-capable, panning (meta)data cannot be transported from one channel to another and multichannel audio transport becomes available on any DAW.

The Dolby Atmos Beam plug-in can be instantiated anywhere in your session and serves as a three dimensional panner, sending both audio and 3D positioning information to the Composer plug-in.

Proper routing

Any experienced mixing engineer will tell you that mixes can be structured in a thousand ways. There might be cases where, for whatever reason, you need to put the Composer on a track that is not the master track. If this applies to you, you'll need to make sure that all tracks with a Beam or Spacelab on them are routed to the track with the Composer plug-in. By this I mean you can either route the track's output or a send from that track to the Composer track. This routing does not have to be direct which means that in between the Beam or Spacelab track and the Composer track can be other tracks such as group tracks. But eventually the routing has to reach the Composer track.

By doing this, you make sure that all instances of Beam and Spacelab are processed before the Composer and all your audio and metadata arrive there in sync. If you don't do this, you might experience crackling noises or strange delays in your sound.

Audio processing issues on some DAWs

Some hosts turn off audio processing for plug-ins when they think that nothing is happening on the channel where a plug-in is instantiated. For example, this can happen when no audio is arriving at the channel where the Composer is instantiated and so the host switches off the plug-in. Our Composer, Beam and Spacelab plug-ins detect if this has happened and display a warning. If it happens you can either route some audio there or instantiate some generator plug-in on the track to keep it processing.

Latency compensation

The Dolby Atmos Composer automatically compensates for track latencies that tend to happen when you use different plug-ins in your sessions with varying latencies. There are a few exceptions however. To know more about them and how to manually compensate please see the chapter on Manual Latency Compensation.

Enforce 7.1.2 bed option

On top of the input channel meters is the Enforce 7.1.2 bed option. This is meant for the rare case that your distributor requires you to have a 7.1.2 bed in your Dolby Atmos mix but your bed/Composite does not contain all of the channels of the 7.1.2 layout, for example when having a 7.1.4 Composite which technically consists of a 7.1 bed and the four height channels as dynamic objects. So with that option enabled your exported mix will contain a 7.1.2 bed plus the 4 dynamic objects representing the height channels of 7.1.4.

Managing connections

- Beam connection



- Spacelab connection



It is good practice to have all your instances named so that you can easily identify them as you mix. When you instantiate a Beam on a track, Beam adopts the name of the track if the DAW communicates that information to the plug-in but you can also change it manually here (1). The connection names can also be changed in Beam and Spacelab and these changes are communicated to the Composer, so you only have to do it once.

The rows of Spacelab instances show the sources inside Spacelab along with their names (7). These source names can be changed in Spacelab. Please check the Spacelab manual and tutorials to learn more about Spacelab.

Whenever you instantiate a Beam or Spacelab plug-in, it automatically connects to the Composer. They appear in the Composer's connections list on the left. Using the Connect button (2) you can manually disconnect them in case you want to exclude them from your Dolby Atmos mix, for example if you want to export different versions of your mix.

For monitoring, you can mute (3) or solo (4) each instance and, if you're using Spacelab, you can also mute (8) or solo (9) each of it's sources.

As a convenience, we have included "Open Beam" and "Open Spacelab" buttons **(5)** so you can open the editor of any instance of these plug-ins directly from the Composer. This way you do not have to search through the whole mixer of your DAW.

For each connection you can choose in what way it will be represented in the Dolby Atmos mix **(6)**. You can choose one of the many possible speaker layouts and let the connection be mixed into the Composite or for Beams you can also choose them to be dynamic objects instead.

Spacelab sources can also be switched to dynamic objects (10). In that case the dry signal of the source goes to the dynamic object channels and the reverb part goes to the Composite.

Managing the connections list



If you have a lot of connected instances, it can be hard to find a specific item. To help with this, we have included a search function at the top left corner of the connections list (1). Just type a part of the name into the field and all non-matching rows go dark. Now you can scroll through and easily find the instance you are looking for.

The Dolby Atmos Composer maintains the order of the instances in the Connections list and this order is initially determined by the order in which the Beams or Spacelabs have been recognized by the Composer. You can manually change the order in two ways. The first way is to use the up and down arrows (2) for moving the selected instances one line up or down.

The second way of changing the order is to double click on the row number of any selected connection and type in the target row number where you would like to move your connections.

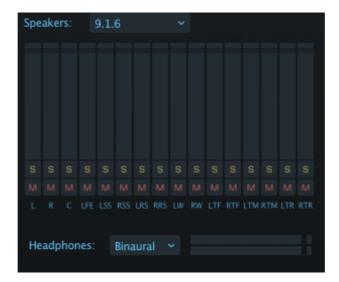
If a selection is not consecutive it will be made consecutive upon changing it's position.

The Unmute button (3) on the top of the Connections list unmutes all instances. If Spacelab is present, all of its sources are also unmuted. And, as you would expect, Unsolo button (4) unsoloes everything.

5. Monitoring



With the top controls you can set the monitoring volume (1), dim the monitoring with one click (2), mute the whole Atmos mix (3) or mute either Composite or dynamic objects (4).



When instantiated on a multichannel track the Dolby Atmos Composer will set the appropriate speaker layout if there is a match. Otherwise headphone monitoring is enabled by default using the binaural setting. You can of course change both manually and you can also monitor both simultaneously.

- If both a speaker layout and a headphone layout is selected, the speaker channels will appear first when looking at the channel order, followed by the headphone channels.
- If the speaker layout matches the layout of the track the output channel order is set according to the plug-in format. The visual order does not change though.
- If the speaker layout does not match the track layout or when using the External Output feature, the channel order is as shown below the meters.



In case your DAW does not support the speaker layout you want to monitor on, for example when using stereo-only DAWs, you can use the External Output feature of the Dolby Atmos Composer to directly access your audio hardware. Here you can select any of the available audio interfaces on your system and thereby circumvent the output limitations of your DAW completely.

We generally recommended using the same audio interface with your DAW that you plan to use for the external output of the Dolby Atmos Composer. This helps avoid timing drift and audio dropouts between different devices. If you must use multiple audio interfaces, you might need to use wordclock or other means to tightly synchronize the clocks of the different audio interfaces.

Some audio drivers, such as RME drivers, are multi-client capable but only if the access is not coming from the same application or process. If you want to output to such drivers you might need to set your DAW to output to a virtual audio device, such as Blackhole or Virtual Audio Cable and then set the Composer to output to your real audio device.

Keep in mind that some drivers do weird things which means that this feature might not work well with some audio devices.

Upon selecting an external audio interface, the External Output Buffer Size usually is set to the buffer size at which the interface is operating. That said, it is possible to change it in case you need to set it to a different value. For example, you may need to match the buffer size used by your DAW after a change.

In case you experience drop outs or interrupted audio you might need to increase External Output Additional Buffer which is initially set to Ox. Increasing the value adds an additional buffer of the set Buffer size above multiplied by the factor set here. Bear in mind that this also increases output latency.

6. Using the Dolby Atmos Beam



The Dolby Atmos Beam plug-in serves two purposes: It brings audio into the Composer plug-in from anywhere in your DAW and it is a sophisticated 3-dimensional panning tool for Dolby Atmos.

If your DAW offers Post Fader Inserts it is recommended to insert Beam in those, because then your pan and volume automation is picked up and included in your Atmos mix.

On the left side you see the input and output configuration. The silver column right next to it contains the controls for object positions, spread and volume. On the right side is the panner and above the panner are the buttons for undo and redo.

When instantiated Beam tries to recognize the track format and set the input names and object positions accordingly. Beam supports up to 16 channels.

Beam input



On the top of the input section you find the input channel list. In this list you can change the input names and mute or solo each of the input. Mute and solo are not just for monitoring purposes and their setting is stored with your session.

Below the input channel list you'll find a "Reset Objects" button. It opens a drop down menu of speaker layout presets for setting the input channels to predefined positions. Especially if the track format is not recognized or if for any reason you want to manually set this, you can do so by selecting the desired layout.

If the Beam plug-in is instantiated on a stereo track, a "Mono" button will become visible next to the "Reset Objects" button. When "Mono" is switched on, the number of input channels or objects is reduced to one and only the audio from the left channel is sent to the Composer. This is very handy when working with a DAW that does not have mono channels and you want to avoid extra and unnecessary objects.

Below that you see the selector for the LFE channel. If the Beam connection is routed to the Composite this channel will be mixed onto the LFE channel of the Composite. Note that if you have set Pan Mode to "Objects", the audio for a channel set as LFE will be ignored and that channel also does not appear as an object in the Composer. That is due to the fact that LFE channels can only be part of a bed and not dynamic objects.

The last control in the input section is Pan Mode where you can select whether this Beam instance is mixed to the Composite or appears as dynamic objects in your Dolby Atmos mix.

Beam output



On the top of the output section you find the field for setting the name of the Beam. When instantiated Beam tries to retrieve the track name from the DAW, which is no problem in most cases. You can change it always to whatever you want and the change is communicated to the Composer.

Next comes the selector for the Composite format of this Beam, which can be different for each Beam in your project.

The Dolby Atmos Composer will mix all incoming connections routed to the Composite. This Composite consists of all channels from all Composite layouts of all these connections. Hence the name Composite.

Below the meters you find the selector "Plugin Out". Here you can select if and how Beam passes incoming audio through back to the DAW. By default it is set to "Beam Vol". That means that the audio is passed through with the volume applied via the Volume Knob to the right. If you set it to "Pass Thru" audio will be passed through unchanged and if you set it to "Muted" the output of Beam will be silent.

Beam control column



Between the input and output sections on the left and the panner on the right is the control column with the three knobs for positioning objects, Spread and Volume.

Spread and Volume are global controls for all objects while Azimuth, Elevation and Distance only affect the selected objects.

Panning in Beam is done with polar coordinates using Azimuth, Elevation and Distance. If Pan Mode is set to "Composite" the panning algorithm of Beam will place the objects accordingly. If Pan Mode is set to "Objects" the polar coordinates get converted to the cubic Dolby Atmos coordinate system for dynamic objects. If you want to know more about that please see the Additional information section further below in this manual.

Azimuth is the angle that determines the position of the object in the horizontal plane. O° places the object directly in front of you while 180° or -180° places the object behind you. If more than one object is selected, the azimuth knob is set to O° and when turning the knob, all of the selected objects move together keeping their spatial relationships.

Elevation is the angle that determines how far the object is above or below you. O° means that the object is on the horizontal plane. -90° means that the object is directly below you, a position also called "voice from hell," while +90° is above you, a position called "voice of God." Again, when more than one object is selected, the elevation knob is set to O° and the selected objects move together while keeping their spatial relationships.

The third parameter is called Distance and —as the name implies— it sets the distance of the object to the listener. 100% means that the object is at its maximum distance while 0% means that the object is inside the head of the listener. The Distance value can also go to –100% for creating a movement through the listener. –100% is also the maximum distance but on the exact opposite side as to where it would be according to the Azimuth and Elevation values. Once again, if more than one object is selected, the distance knob is set to 0% and the objects move together while maintaining spatial relationships. With more than one object selected this knob has a range of –200% to +200% to get the full motion range for all selected objects.

If Pan Mode is set to "Composite" and an object gets closer to the listener, more speakers of the composite are involved in the playback of that object. This increases the perceived size of the object. If Pan Mode is set to "Objects" the Atmos parameter "size" of the object is changed accordingly creating a similar effect. In that case size is a metadata parameter of the corresponding Dolby Atmos dynamic object.

The fourth knob is called "Spread." It always affects all objects and as you increase this value it causes the objects to behave similarly to as if they would be coming closer to the listener.

Volume is the last knob and, with it, you can adjust the volume of all objects sent to the Composer. If Plugin Out is set to "Beam Vol" also the audio passed through Beam and back to the DAW will be adjusted in the same way.

3D Panner

The panner in the Dolby Atmos Beam comes in three flavours. When you open the Beam editor you will see the 3D Panner first.

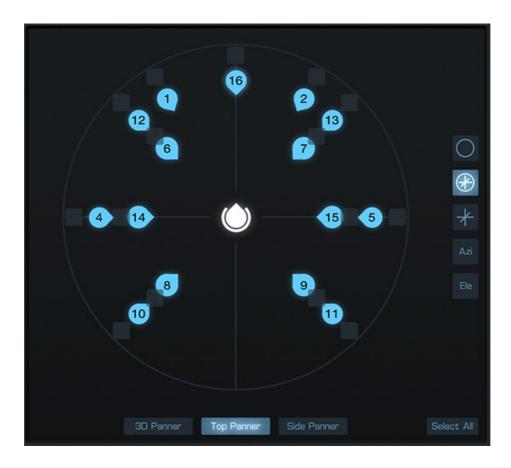


In the 3D Panner you can turn the view around by right-clicking & dragging with your mouse. If you're working on a laptop and don't have a mouse, you can command-or-control click and drag.

The viewing angle is important not only for having an idea where the objects are but also for certain panning modes. More about the panning modes further below.

For better visualisation of the viewing angle not only the listener itself is shown but also small speakers indicating the speaker positions of the Composite.

Top Panner

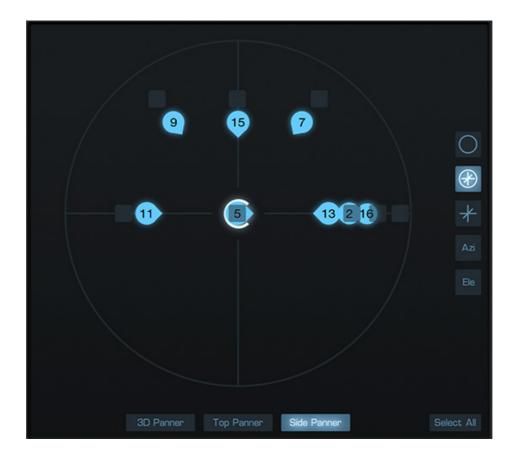


The Top Panner is as the name implies a 2D panner with viewing angle right from the top. This viewing angle lets you easily asses the objects position in relation to the horizontal plane around you.

The small grey squares represent the speakers in the Composite layout. As you can see the front of the whole space is on the top and the listener, represented by the white symbol in the center, is pointing there.

The objects are numbered according to the input lists on the left. They appear bigger when closer to the top and smaller when closer to the bottom.

Side Panner



The Side Panner is, like the Top Panner, a 2D panner. But here the viewing angle is such that you see the whole space from the right side, just like looking into the right ear of the listener.

Consequently the front of the space is to the right and the rear to the left. In the image above you can see the listener behind the right side speaker and an object at the same position.

Here also the grey squares represent the speakers of the Composite and the objects are numbered according to the input list on the left of Beam.

Operating the Panner

There are several ways to select objects. You can select the corresponding channels in the channel list to the left and the respective objects become highlighted. You can also click the "Select All" button below the panner to quickly select everything. Or you can directly select objects in the panner display by either clicking or shift-clicking them one by one or by using the lasso tool. Clicking into the empty space deselects everything.



In any of the three panners you can choose from five different panning modes. By default the second mode is selected. When this mode is active the objects move in an orbit around the listener following the mouse movement. Since the sound of objects changes drastically when objects change their distance to the listener, the second mode moves the objects in a way that they maintain the distance to the listener as they move spherically around the it.

In the third panning mode, objects also follow mouse dragging but here on a straight line through space. That means that their distances to the listener and therefore their perceived sizes change depending on where you move the mouse.

In the first panning mode, the relative position that objects have to each other are maintained and the whole arrangement is rotated around the listener.

When working with the 3D-panner, you may want to adjust your view according to the planned movement of objects when using one of the first three panning modes. The other panning modes are independent of the viewing angle.

The fourth mode is for changing azimuth only. That comes in handy if you do not want to alter the height perception of the objects but just create some kind of rotation or horizontal movement.

The top panner is optimized for azimuth automation. In the fourth panning mode you can grab the objects and rotate them around the listener indefinitely doing circular movements with the mouse.

And the fifth mode is for changing elevation only. Note that elevation is implemented in an unlimited way similar to Azimuth. This means that when an object's elevation goes beyond its normal boundaries, the object comes around on the other side. That way vertical circular movements can be done

The Side Panner is optimized for elevation automation. In the fifth panning mode you can move the objects on a vertical circle indefinitely doing circular movements with the mouse.

Please check out our video tutorials about panning as they demonstrate visually how it all works.

7. Panning and reverb with Spacelab



As soon as Spacelab is instantiated on a session where the Dolby Atmos Composer is present, both the Composer and Spacelab recognize each other and establish a connection. By default 9.1.6 is selected as the format for rendering reverb in your Dolby Atmos mix but you can change it in the Dolby Atmos Composer connections list to any other format Spacelab supports, even your very own speaker layouts created in Spacelab.

Spacelab has two operating modes, one is called Object mode, which is the default and where you have access to all the sophisticated panning features of Spacelab. The other is called Classic mode which is designed to be used when you only need the reverb portion of Spacelab and you want to use it as a send and return effect, as you would with other reverbs. The classic mode technically does not work when connecting to the Composer. So if you have this mode selected, it will automatically switch back to Object mode upon connecting to the Composer. But no worries, Spacelab still continues to work as intended since the Classic mode is just a subset of the Object mode and internally simply uses one source with the channel configuration you had selected as the input speaker layout of Spacelab.

All Spacelab sources are panned within Spacelab to the selected speaker layout by default, except if they are dynamic objects. If you convert a source into a bunch of dynamic objects, extra channels in your Atmos mix are reserved for them. The reverb will still go to the speaker layout which is routed to the Composite but the dry sound of those sources is going to the newly created dynamic objects in your Atmos mix. You can convert a source to dynamic

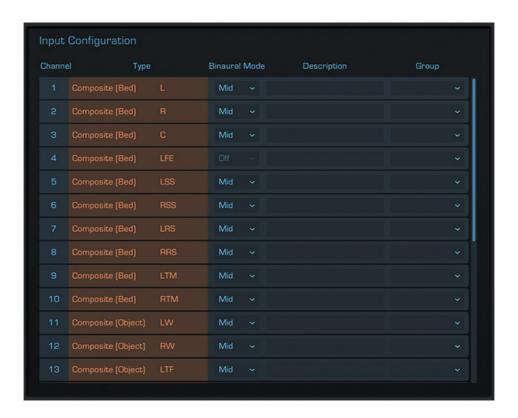
objects either in the Dolby Atmos Composer or in the source setup window in Spacelab (please check out the Spacelab tutorials and manual).

You can also move around the listener in Spacelab. In Spacelab Ignition the listener has 3 degrees of freedom and in Spacelab Interstellar it enjoys 6 degrees of freedom. When the listener moves, the positions of the objects relative to the listener change and this relative position is used to calculate the object position for Dolby Atmos. In that way complex sceneries with multiple movements of both objects and listener can be automated and rendered in a human comprehensible way. This is true for both room related and listener related sources in Spacelab.

Please checkout the Spacelab tutorials and manual for further information on all the features of Spacelab as they translate perfectly to the Dolby Atmos world using the Dolby Atmos Composer integration.

8. Input configuration & program level metadata

In the Dolby Atmos Composer you have full access to all the usual parameters you need to finetune your Dolby Atmos Mix.



On the Input Configuration page you have access to the channel related features of Dolby Atmos. In the Binaural Mode column you can change how each channel is rendered when listening to your mix in binaural. Dolby Atmos offers 4 different options for each channel. "Off" means that no binaural processing is applied to the channel while the remaining three modes differ in the perceived distance.

As a side note, with the current implementation of Apple Spatial Audio, Apple's way of Dolby Atmos reproduction on headphones, these settings unfortunately are ignored because the Apple algorithm basically renders your mix first to the 7.1.4. speaker setup and then converts that to binaural.

In the Description column you can input an arbitrary text for describing the content of the channel. This text however is the same for all channels belonging to the Composite.

In the Group column you can select a group for each channel from a predefined list of groups. This list can be changed on the Options page. The group must be the same for all channels belonging to the Composite.



On the top right corner of the Options page you can see the place where you can manage the available groups. The names can be arbitrary as well and you can add and remove groups as you see fit. Using the Export button of each group you can disable or enable them for export giving you the possibility to export well defined parts of your Dolby Atmos mix one by one.

To the left of the groups management section is a set of controls which allows you to configure how your mix will be rendered on certain speaker layouts. The first two dropdown menus let you choose the algorithm for downmixing from 5.1 to stereo and from 7.x to 5.1 and 5.1.x. This is used for monitoring and when exporting your mix to a multichannel wave file. To exactly know what these algorithms do please refer to the Additional information section of this manual.

Below that you find the controls to set trim and balance values for five different speaker layouts. First you need to select which layout you'll be editing using the dropdown menu.

If the "Automatic" button right to the dropdown menu is switched on, the Dolby Atmos renderer sets these values internally and the knobs are greyed out. If the button is switched off, the knobs become available for manual adjustment.

The Trim knobs are attenuations in dB. One knob sets the attenuation for the surround speakers while the other sets the attenuation for the speakers in the upper plane, or "height" speakers. All associated "surround" and "height" speakers are attenuated equally by the values set here.

The balance values are in percent and determine the front-to-back balancing ratio. This applies to both the ear-level listener plane and for the overhead "height" plane. Positive values emphasize the front while negative values emphasize the back.

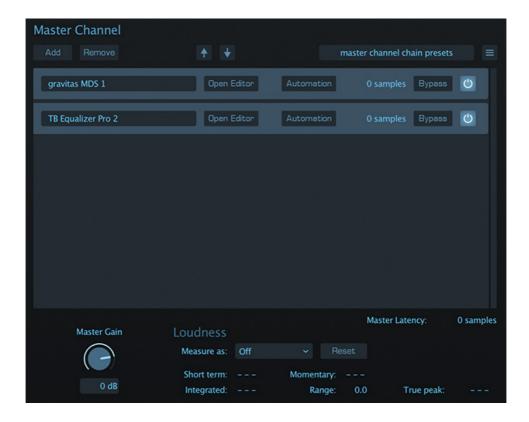
Below that are the three master settings: Sample Rate, Frame Rate and Timecode start. "Sample Rate" can be set to either 48 kHz or 96 kHz as these two are the only options allowed for Dolby Atmos content. If your DAW session does not match the set sample rate an error is displayed next to the sample rate dropdown telling you that your session is at the wrong sample rate. Also the error button becomes visible and you are prevented from exporting until correcting your session's sample rate.

It is also worth mentioning here that the optimal buffer size you should set your DAW to is 512 samples if you are working in 48 kHz or 1024 samples if you are working in 96 kHz.

The frame rate dropdown menu lets you select your project's frames per second setting. This option is only relevant if you are producing Dolby Atmos content for video or film. The various timecode values shown throughout the plug-in such as "timecode start" as well as the "inpoint" and "outpoint" are calculated based on your selected frame rate.

"Timecode start" should be set to the timecode with which your DAW session starts. This value and the frame rate must be set correctly in order for the inpoint and outpoint values to be correctly displayed as timecode.

9. Mix processing with the Master Channel



The Master Channel sits directly before the Dolby Atmos renderer, but after all the incoming connections have been mixed together. The channels of the mix pass through the OBAM plugin chain in the Master Channel and are then leveled by Master Gain. Master Gain by the way can be automated. After that, the mix goes into the Dolby Atmos renderer integrated into the Composer where the rest happens.

As a side note, there are actually two chains in the Dolby Atmos Composer, one chain is for the mix you are working on while the other is for imported ADM/BWF files.

Another side note is that any OBAM plug-in which supports an external sidechain input can receive audio through the sidechain input of the Dolby Atmos Composer, when working on a mix.

Use the "Add" button to add a plug-in and the "Remove" to remove a selected plug-in from the chain. With the arrow buttons further right you can move a selected plug-in up or down in the chain. Right to the arrow buttons you find an initially empty field and by clicking on it you can load previously saved plug-in chains as a preset. With the "hamburger" button you can save and otherwise manage those plug-in chain presets.

Once a plug-in is added, it is automatically given a name, which of course you can change to whatever you like. Next to the name field, you see the buttons for opening the editor of the plug-in, for configuring automation parameters, and for bypassing or deactivating the plug-in.

Also there is a label stating the latency in samples of the plug-in. This number can change depending on the settings inside the plug-in. Please note that in cases when the latency is greater than O samples, bypassing will not change the latency time while deactivating the plug-in will eliminate the latency caused by it. This happens because bypassing simply bypasses the plug-in while keeping it in place whereas deactivating a plug-in takes the it completely out of the chain.

At the bottom of the plug-in list, you will find "master latency" which is the latency of the entire chain. Currently, the maximum possible latency is 192.000 samples. If there is any latency, all incoming panning data is synchronized so that everything remains in time.

When working on a mix the parameters of the plug-ins can be automated by the dynamic automation feature of the Dolby Atmos Composer. Clicking on the "Automation" button of a plug-in opens the menu seen in the image below where you can dynamically assign a parameter of the plug-in to an automation lane of the Composer by selecting the lane from the dropdown menu. Master Gain is on lane one so the lanes here start at number 2.



The image on the left shows that Input Gain from the first plug-in is on Lane 2.

The number before the bracket is the position of the plug-in in the chain to which this parameter is connected. If you reorder the plug-ins in the chain, those numbers change accordingly.

This way your automation stays intact even if you change a plug-in's position.

10. Loudness measurement



On the Master Channel page you find the loudness measurement section at the bottom. Before you can start measuring you first need to select the right format. After that, measurement starts as soon as sound passes through.

Before checking your mix for compliance with the distributor's requirements please do not forget to reset the measurement using the "Reset" button adjacent to the format selector. Then you need to play your song from the intended start to the end without stopping to get the right values for measuring.

Of the five values shown here, integrated loudness and true peak are the most important. The integrated loudness must be calibrated to a certain target value a distributor may require you to match before they accept delivery while true peak shows you if there is any level overshoot.

11. Export

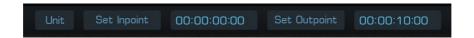
For exporting you first have to tell the Dolby Atmos Composer what formats you want to export. You can export the ADM/BWF file and/or the speaker output and/or the headphone output and you can select that on the Options page.



All of these export options can be selected simultaneously and you will obviously need at least one for the export to work. By default ADM/BWF is selected which is the delivery format of your Dolby Atmos mix.

The speakers and the headphones renderer outputs can be set to export as one single multichannel file or as a set of mono files, one for each channel. Once you have selected one of the two options, you can then select your desired format. This will switch the renderer to the selected format and monitoring will also be set to this format. You can see this on the monitoring tab accordingly. So, if you have exported to a different format than what you normally use for monitoring, you may need to switch this back to your normal monitoring format before carrying on with the mixing process.

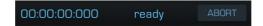
Next, you have to tell the Composer which part of your DAW session you want to export. Remember that you don't have to export the entire session and you can export a smaller section if you like. You can do this by setting the inpoint and outpoint on the bottom of the Composer editor.



That can be done either by moving the song position pointer to the desired positions and clicking the on-screen buttons "Set Inpoint" and "Set Outpoint", or by entering the values manually. If you enter the values manually you can do so either in time code or in samples. The "unit" button allows you to switch between both. When you enter the values as timecode the frame count aligns with the selected frame rate on the Options page.

On some DAWs the on-screen buttons only work during playback, not allowing a precise setting. This is due to the fact that those DAWs only communicate project position during playback. In this case you have to resort to manually entering the values.

Once the range for export is set the next thing is to click the Export button and select a filename and folder for the files to export. When this is done the Dolby Atmos Composer switches to export mode showing the label "ready".



You can export in realtime by playing back the whole export range or you can use your DAW's offline export function over that same range. In both cases you need to make sure that the whole range is covered by the process. That means that your export, no matter if realtime or offline, starts at or better a bit before the inpoint and ends after the outpoint.

If your export does not start at or before the inpoint the Composer will not even start writing your export files and if the playback or offline export does not actually reach at least the position of the endpoint the Composer will not complete the export.

If your export has started you will see a percentage showing how much of the set export range already has been exported. With the ABORT button you can make the Composer abort the export process.



If your playback or offline export hits the endpoint the percentage display will disappear and a message will be shown about the success of the export process.



Since the export files are always written by the Composer you can safely delete the files exported by your DAW in case you have used offline rendering.

If you choose to export the renderer output, be it speakers or headphones, an appendix is added to the selected filename indicating the content of the exported files.

Some DAWs switch off audio processing when they consider that there is nothing to process. That might happen if you set the outpoint at the end of your song and nothing is there anymore, no clip, no region etc. In that case your export might not be finished. You have to make sure that audio processing is always on for the Composer. Some DAWs, such as Pro Tools, offer an option to switch off the feature of automatically stopping audio processing and on others such as Logic Pro you have to make sure that something is there to make the DAW keep processing. See the section about DAW-specific settings.

12. Importing and editing ADM/BWF files

Apart from all the features for mixing Dolby Atmos and exporting your mix to a legit Dolby Atmos ADM/BWF file, the Composer also offers editing existing ADM/BWF files, no matter with which workflow they were created. The Composer is not limited to editing ADM/BWF files originating from the Composer itself but you can edit any legit Dolby Atmos ADM/BWF file.

On the Options page you find the Import button at the bottom. Clicking on it opens the load dialog where you can select the ADM/BWF file for import. Once the file is imported the Composer goes into "file mode". All the connections from Beam and Spacelab on the left disappear and playback controls become visible at the bottom.



Apart from playing back, either by clicking the play button or hitting the space key, you can move the playback pointer to any place of your loaded file to play back the desired section.

When in "file mode" the Input Configuration page is renamed to Channel Configuration. It offers you to change Binaural Mode, Description and Group of each channel just like when you are mixing. In case you need you can edit the list of available groups on the Options page.

The Master Channel is also available for processing your imported ADM/BWF files, just as you would use it for your mix. The external sidechain input and plug-in automation however are not available. Also there is no display of the plug-in latency and the master latency because of the lack of relevance as internally everything is synchronized. For "file mode"-specific features of the different plug-ins please refer to their the product manuals and tutorials.

The loudness measurement section on the Master Channel page has an additional function, the "Measure ADM" button. By clicking this button you can measure the loudness of your imported file including all the plug-ins and Master Gain to quickly check for compliance with the distributors requirements. Before doing that you must have selected the format in which to measure the loudness.

On the Options page you can change downmix and trim values and you can also manage available groups for selection on the Channel Configuration page. Notice that sample rate and frame rate cannot be changed, but you can set a new timecode start if you need to.

Now you can re-export the imported file with all the changes you have made, as ADM/BWF and/or as re-render wave files as you can do with any Dolby Atmos mixing session. The only difference is that you do not need to set the inpoint or outpoint, since start and end of the imported mix are already known.

When you are done working with the imported file, just click "unload" on the Options page and the Composer switches back from "file mode" to normal operation and your connections become visible again together with all the settings of your Dolby Atmos mix.

13. Manual latency compensation

Some DAWs do not communicate a latency compensated song position to the plug-ins. Currently known DAWs with this flaw are:

- Pro Tools (not entirely, but to some extent, please see chapter about <u>DAW-specific settings</u>)
- Samplitude
- Sequoia

The list will be updated as soon as the bug is fixed in those DAWs.

This bug makes automatic latency compensation impossible. You can however compensate manually. You have to calculate the latency occurring up to the Dolby Atmos Beam (or Spacelab) and input that latency in samples on the about screen of Beam (or Spacelab). This will notify the Dolby Atmos Composer of the occurring latency so that it can synchronize everything accordingly.

Our tutorial video "Delay Compensation in Pro Tools" demonstrates this process in detail. The process is the same on any DAW and you can find it here: youtu.be/A55pMjfSjow

In section 18. (<u>DAW-specific settings & recommendations</u>) you can find a description on how to make automatic latency compensation work with the Dolby Atmos Composer in Pro Tools.

14. Additional information

System Requirements

Plug-in Formats: VST3, AU, AAX

Supported Operating Systems: macOS 10.14 through 15 / Windows 10, 11

CPU: Intel min. 2 GHz, x64 with at least SSE3 support, or

Apple Silicon M1 or higher

Display/Graphics: min. 1440 x 900 px, OpenGL 3.3 or newer

Memory: min. 4 GB RAM

IMPORTANT: The Dolby Atmos Composer plug-in only works correctly and connects with the instances of Dolby Atmos Beam and Spacelab when only one instance of the Composer is running at a time! You cannot run more than one instance of the Dolby Atmos Composer plug-in simultaneously, not in different sessions nor in different DAWs! Some DAWs, such as Logic Pro, need to be restarted when switching to another session, because just closing the session does not seem to terminate the old instance of the Dolby Atmos Composer completely.

The following list contains the Azimuth and Elevation values for the speakers found in the composite formats of the Dolby Atmos Beam and the Dolby Atmos Composer.

Speaker	Azimuth	Elevation
L/R	30° / -30°	O°
С	O°	O°
LS / RS	110° / -110°	O°
LSS / RSS	90° / -90°	O°
LRS / RRS	135° / -135°	O°
LW / RW	45° / -45°	O°
LTF / RTF	45° / -45°	45°
LTM / RTM	90° / -90°	45°
LTR / RTR	135° / -135°	45°

Description of 5.1 and 5.1.x downmix options

- "Standard (Lo/Ro) default" downmix to 7.1 and then to 5.1 using the coefficients:
 - * Ls = $0 dB \times LSS + 0 dB \times LRS$
 - * Rs = O dB × RSS + O dB × RRS
- "Dolby Pro Logic IIx" downmix to 7.1 and then to 5.1 using the coefficients:
 - * Ls = LSS + $(-1.2 \text{ dB} \times \text{LRS}) + (-6.2 \text{ dB} \times \text{RRS})$
 - * Rs = RSS + $(-6.2 \text{ dB} \times \text{LRS}) + (-1.2 \text{ dB} \times \text{RRS})$
- "Direct Render with room balance" Renders from Dolby Atmos to 5.1 directly applying an updated Dolby rendering algorithm that reduces the comb filter effects associated with phantom imaging of objects positioned halfway between the front and rear of the room. Room balance refers to how the Renderer deals with content that is panned between the midpoint and rear of the room. With this setting, the content is presented at a constant level in the surround speakers between the rear and midpoint of the room, avoiding any need for phantom imaging until it is in the front half of the room.
- "Direct render" Renders from Dolby Atmos to 5.1 directly accurately re-creating the sound field at the central listening position using phantom imaging between the surround speakers and front speakers.

Description of the 5.1 to 2.0 downmix options

The coefficients for the two-channel downmixes from 5.1.x are:

- "LoRo"
 - * Lo = L + $(-3 dB \times C) + (-3 dB \times LS)$
 - * Ro = R + $(-3 dB \times C) + (-3 dB \times RS)$
- "Lt/Rt (Pro Logic II) and Lt/Rt (Pro Logic II) w/Phase 90"
 - * Lt = L + $(-3 \text{ dB} \times \text{C}) (-1.2 \text{ dB} \times \text{LS}) (-6.2 \text{ dB} \times \text{RS})$
 - * Rt = R + $(-3 \text{ dB} \times \text{C})$ + $(-6.2 \text{ dB} \times \text{LS})$ + $(-1.2 \text{ dB} \times \text{RS})$

The phase 90 filter used provides the all-pass 90-degree phase-shift filtering for the LS/RS feeds into the downmix, which reduces undesirable signal cancellation, improves imaging, and enables proper matrix decoding. It is strongly recommended to use the 90-degree phase shift for any Lt/Rt downmixes.

Differences between Dolby Atmos Composer and Dolby Atmos Composer Essential

Dolby Atmos Composer	Full Version	Essential Version
Supported Sample Rates	48 kHz, 96 kHz	48 kHz
Individual Composite format per connection	yes	no
Channel Description & Channel Groups	yes	no
Downmix & Trim Settings	yes	no
Simultaneous Monitoring and Export of Speakers & Headphones Output	yes	no
Frame Rate & Timecode Start	yes	no
Integrated Loudness Measurement	yes	no
Master Channel with OBAM plug-ins	yes	no

Dolby Atmos Beam	Full Version	Essential Version
Top Panner & Side Panner	yes	no
Manual LFE Channel Selection	yes	no
Individual Composite Selection	yes	no
Pan Mode Selection	yes	no

15. Video Tutorials

Check out our video tutorials on our YouTube channel.

Channel: youtube.com/@fiedler-audio

Dolby Atmos Composer tutorial playlist: DAC tutorial playlist

16. Trial & Purchasing

After downloading the installer and installing the plug-in you have a 14 day trial period. The plug-in is fully functional during the trial period. To start the trial period you need to click "Try" on the about screen of the plug-in which opens after first instantiation or opening the editor. On the about screen you can also see the remaining days of your trial. The about screen can be opened manually by clicking on the product logo or on the fiedler audio logo.

The above mentioned way to start your trial requires an active internet connection. If for some reason you do not have an internet connection on the computer you are using for your trial you will instead be prompted with a way to start your trial offline. The dialog windows which open will guide you through this process which is basically a challenge & response type activation. You will first have to save a file called "comp-id.xml" which contains a digital fingerprint of your computer. This file you have to upload to our website to get the response file with which you can then start the trial offline by loading it into the plug-in in step 2 of the whole process.

Once the trial period ends the plug-in stops working and you need to activate it with a serial number. To purchase a license please visit our <u>website</u> and click on the "Buy Now" button of the desired product. A popup will open and you will be able to make your purchase. The payment options offered depend on the country and the purchase is processed through Fastspring (<u>www.fastspring.com</u>).

After successful payment the serial number will be sent to you automatically via email. If you are planning to buy several different products please check out our bundles to get discounts.

Note: If the trial period has expired but you didn't have the chance to properly evaluate the plug-in, you can request an additional trial period by contacting us through the contact form on our homepage. You will then get a trial extension serial number which you have to copy into the serial number field on the about page and hit "Try" (not Activate!).

17. Activating & Moving your licenses

After purchasing the Dolby Atmos Composer you will receive a serial number via email. To activate the plug-in just copy the serial number, paste it into the license number field on the about screen and hit "Activate". The window will close automatically and the plug-in is activated. A regular license allows simultaneous activation on two computers.

For that process to work you need an active internet connection. If for some reason you do not have an internet connection on the computer you want to activate you will instead be prompted with a way to start your offline activation. The dialog windows which open will guide you through this process which is basically a challenge & response type activation. You will first have to save a file called "comp-id.xml" which contains a digital fingerprint of your computer. This file you have to upload to our website to get the response file with which you can then activate offline by loading it into the plug-in in step 2 of the whole process.

If you need to move your license to another computer you can deactivate the plug-in to free one of the seats of your license on the old machine and then activate it on the new computer. To do so please open the about screen of the plug-in on the old machine by clicking on the product logo or the fiedler audio logo and then click onto the "Deactivate" button. Again, this works out of the box with an active internet connection but if you do not have an active internet connection on this system you will have to go through the same process with challenge and response as you would have with activation. There is no limit regarding the amount of deactivations so you can move freely between machines.

IMPORTANT: Uninstalling the plug-in does NOT deactivate it. If you have not deactivated the license as described above, the license is still active on that machine.

18. DAW-specific settings & recommendations

Looking at the available parameters for automation in some DAWs you can see a parameter called "DO NOT USE". Please do as it says and do not touch it. This parameter is used for notifying the DAW that something in the plug-in has changed and forcing the DAW to mark the session as "dirty". This will require the user to be asked for saving the session upon closing it. If you recorded automation on it by accident please delete the recorded automation data to make it work correctly.

Some DAWs such as Cubase, Nuendo, Ardour, Mixbus and Digital Performer 11 have so-called Post Fader Inserts. These are plug-in slots on a track sitting after the pan pot and the volume fader. This great feature allows you to place the Dolby Atmos Beam and Spacelab plug-ins into these Post Fader inserts to receive all the pan and volume automation which is especially handy when augmenting an already existing stereo mix with a Dolby Atmos version.

Pro Tools

To make automatic latency compensation work in the Dolby Atmos Composer on Pro Tools you have to instantiate it either on an Aux Input track or on a Routing Folder track. It is recommended to make this track your master track by routing the entire track hierarchy of your mix there.

It is also recommended to switch off "Dynamic Plug-In Processing" in the Playback Engine settings. This way it is made sure that the connections between the Dolby Atmos Beam, Spacelab and the Composer are always in time and the export process is always working correctly even if the outpoint is beyond the end of the entire session.

The Dolby Atmos Composer offers dynamic automation of the parameters of OBAM plug-ins in the Master Channel. These parameters become only visible in the automation menu once assigned to the respective automation lanes. To know how to do that please check out the chapter about the Master Channel in this manual.

Reaper

In Reaper you have to switch off Anticipative FX processing in two places: Preferences ->Audio->Buffering & Preferences ->Audio->Rendering. If you do not switch off this option you will likely experience audible artefacts.

Reaper also sometimes resets the Composer before doing offline exports. To avoid this behaviour where the export process does not start switch off "Inform plug-ins of offline rendering state" in Preferences->Plug-ins->VST.

Logic Pro X

It is highly recommended to switch off "Only load plug-ins needed for project playback". This option can be found in File->Project Settings->General. When doing so, loading your project will take longer since all plug-ins will be loaded at once but your Dolby Atmos mix will only then correctly be recalled in the Dolby Atmos Composer.

When closing a project containing the Dolby Atmos Composer and starting/loading another project also containing the Composer Logic Pro does not kill the instance of the Composer of the old session and still keeps it in memory. That will cause the Composer in the new session to not work properly. So you have to close Logic Pro completely after closing the old project and load/start the new project only in a freshly opened Logic.

Ableton Live

The Dolby Atmos Composer uses dynamic automation for OBAM plug-in parameters. After assigning an automation lane to such a parameter it becomes available for automation (please see chapter about the <u>Master Channel</u>). To be able to access this parameter in Live you have to first click on the triangle button of the Composer instance to make the automation parameters visible, then you have to click the "Configure" button which just became visible with the parameters and then touch the assigned parameter using it's control element in the OBAM plug-in editor once to make the parameter appear in the parameter list on the right (green). After doing so, switch off "Configure" again.



19. Modifier keys

Knobs and sliders can be dragged in a fine tuned way using Shift Key and/or Cmd/Ctrl Key. Both Shift and Cmd/Ctrl can be combined for an even finer control.

Double click on a Slider or Knob resets it to it's default value.

Hovering with the mouse over knobs, buttons, sliders etc. reveal quick hints about their functions.

20. Support

If you need help with operating our software please check out our <u>video tutorials</u>, the <u>knowledge base</u> on our homepage and don't hesitate to contact us through the <u>contact form</u> on our homepage.

If you think that you have encountered a bug in our software please first make sure that you have the latest version installed. You can check the version of the software on the about screen. The about screen can be opened by either clicking on the product logo or on the fiedler audio logo in the editor. If you are on the latest version and the bug is still present please contact us through the <u>contact form</u> on our homepage. Please provide information about the software you are using, the operating system, the main hardware specs of your computer and a detailed description of how to reproduce the bug if possible. Thanks in advance!

21. Installation & deinstallation

When installing the plug-ins, the installation program will copy the plug-in into the appropriate plug-ins folders, and in most cases your host will recognize them automatically.

If you want to uninstall our plug-ins you can do so on Windows using the Control Panel.

On macOS, plug-ins are installed in the standard plug-in folders in the Library folder.

Audio Units: /Library/Audio/Plug-Ins/Components

VST3: /Library/Audio/Plug-Ins/VST3

AAX: /Library/Application Support/Avid/Audio/Plug-Ins

To uninstall the plug-ins on macOS you have to manually delete them from these folders. To also delete the presets and other settings you have to go to the folders

/Library/Application Support/Fiedler Audio and

~/Library/Application Support/Fiedler Audio

and delete the appropriate folder(s) inside.

Note: Since OS X 10.7 (Lion), the system and user Library folders are marked as hidden by default. To make them visible again in Finder, open Terminal (found in /Applications/Utilities) and enter the following commands:

chflags nohidden /Library chflags nohidden ~/Library

22. Acknowledgements

A huge thanks to all our beta testers for their relentless testing of the different beta versions! Special thanks go to Thomas Wendt for making our plug-ins visible to the world.

Furthermore we would like to thank all our users for their support and loyalty over the years. You have made all this possible.

23. About fiedler audio

Fiedler Audio was founded 2013, with the goal of delivering the highest quality products for musicians, audio engineers and sound designers. We are dedicated to the creation of professional music and audio software that expands the horizons of musicians, DJ's, audio engineers and producers. Our greatest desire is to enable amateurs and professionals alike to realize their dreams and ideas at the highest level, wherever they may be – whether in the studio, at a gig, in the comfort of their living room or in the park, our software offers new and innovative ways to evolve.