Audiolense 5.0 - Introduction
Audiolense is a speaker & room correction system with digital crossover functionality, and a registered trademark of Juice Hifi.

Audiolense adopts the sound of your audio system to your listening environment.

A three way stereo correction with digital crossovers

Audiolense comes in three versions. The 2.0 version provides room correction for a conventional stereo setup. The surround version provides room correction and bass management for conventional setups from 2.0 to 7.2. The XO version provides high end sound correction with flexible digital crossovers for active systems. It is a highly customizable solution with a relatively low getting-started threshold. And as soon as you’ve found your favourite settings, they are stored until next time.
The best way to familiarize yourself with Audiolense is to check out the various features in the gui, experiment, read this help file and sign up to the Audiolense User Forum. The major purpose of this help file is to provide some getting started assistance.

Revision History & New in Audiolense 5.0

New in Audiolense 5.0:

- The core algorithms has been completely reworked
- TTD correction per speaker and optionally per driver.
- Partial correction has been extended to include partial TTD correction
- Clock drift correction in measurement is now practically perfect.
- Sophisticated pre-ringing prevention TTD correction, including:
  - Hands-on noise removal from the measurement
  - A test run that enables the user to inspect removal of problematic reflections
  - A test run that enables the user to inspect how partial correction and dip limiting is handled.
  - An option for selective removal of problematic reflections
  - Improved TTD pre-filtering that is much more robust and effective than earlier with regards to pre-ringing
  - Increased flexibility in bass management, including multi-driver subwoofers and cascaded bass offloading.
- As a result, it is much easier to achieve a TTD correction without Audioble pre-ringing.
- Target Designer enables various combinations of minimum and linear phase targets.
- Improved crossover handling with cleaner pulse, especially combined with TTD Correction.
- TTD combined with minimum phase crossover is now included.

New in Audiolense 4.11 - 4.13:

- Improved correction boost limiting. More predictable boost.
- Support for MiniDSP - Open DRC / miniShark - with with correction saved as binary mono files.
- Improved resampling

New in Audiolense 4.9 and 4.10:

- 64 bit version with better capacity for large speaker systems
Clock drift correction, for better timed measurement when different devices are used for speaker and microphone.

**New in Audiolense 4.8:**
- Refined bass management. The optional 10dB LFE boost is shared between subwoofers.

**New in Audiolense 4.7:**
- Implemented choice between using target as eq and not eq in partial corr
- Changed hardware id generation. File is now stored in the Audiolense folder under My Documents and hopefully more easy to access than previously.
- Reduced default measurement level to -20dB
- Removed data from chart views.
- Added an “export simulation” function. This will enable the user to e.g. load the simulation in REW for further examination.
- A number of bug fixes

**New in Audiolense 4.6:**
- Much improved partial correction (Optional, Audiolense XO)
- Minor tweak to improve the integration with the JRiver Convolver.

**New in Audiolense 4.2-4.5:**
- Two different methods for pre-ringing suppression (Optional, Audiolense XO)
- Optional low pass crossover for subwoofers – default is 250 Hz (Audiolense XO)
- Improved crossover handling that leads to better correction and better stop-band rejection in the crossovers.
- Improved frequency correction (Audiolense XO And Surround)
- Improved memory management -> improved stability with large setups.
- Choice between minimum phase and linear phase measurement.
- The measurement module has been prepared to handle very long measurements for large systems.
- Improved peak identification and speaker driver alignment with digital crossovers.
- The multi seat correction can now combine measurements with different sample rates.
- Optional 64 bit float correction filters. 32 bit float is default.
- Optional 10dB amplification of the LFE channel.
- A number of bug fixes and incremental improvements

**New in Audiolense 4.1:**
Separate pre-ringing control (Audiolense XO only)
Various optimizations to TTDC.

New in Audiolense 4.0: Try before you buy & True Time Domain Correction
Try before you buy functionality: You can now create filters and prepare 90 seconds of corrected music for playback through your ordinary audio player.
In Audiolense XO:
- Group Delay Correction has been renamed to True Time Domain Correction. We feel that this term is more representative of the actual benefits and how it really operates.
- True Time Domain Correction towards minimum phase behavior. See separate chapter.
Improved and more analysis functions.
Various improvements of the GUI

Main Features
State of the art room correction.
Psychoacoustic oriented correction
Digital Crossovers (Surround and XO version)
Multichannel measurement functionality
Speaker and driver level and time alignment
High quality measurement, with quality control.
Easy to use graphical user interface. The whole process is visualized
Draw your own target curves. An easy-to-use Target Editor is provided
Bass management (not in the 2.0 version)
Multi seat correction in Audiolense XO (correction based on several measurements)
True Time Domain (TTD) correction in Audiolense XO (Corrects the time domain as well as the frequency domain)

2.0 Version
- 2.0 setup
- 5 frequency correction procedures, including for Open DRC and miniSHARC.

Surround version
- supports 2.0 – 7.2 setup and has
- 8 correction procedures
  - 3 with frequency correction & linear phase crossovers
  - 3 with minimum delay correction & crossovers
- OpenDRC and miniSHARC
- Very flexible bass management

**XO Version:**
- Includes all features in the Surround and 2.0 version
- 3 predefined correction procedures
- Correction procedure designer: Create your own correction procedures
- Full support for active systems with digital crossovers
- Full, partial or no room correction on a per speaker basis in time and frequency domain.
- True time domain + frequency correction
- Minimum phase crossover filter
- Variable correction filter length
- Custom frequency dependent window settings for measurement & correction filtering
- Multi seat correction
- RIAA correction for vinyl playback or vinyl ripping

**System Requirements and recommendation**

- **Windows Vista or newer. Windows 10 is supported.** Sound card, microphone and microphone preamp for measurement purposes is also required.
  - Windows’ Smart Screen may block the installation. You can disable the Smart Screen from the Control Panel. The User Account Control may block / warn against the installation. Ignore or click “more info” to proceed. Contact us if you’re in doubt – and we will verify that the install files are still 100% OK.
  - If the program doesn't run normally, you may need one or two of these:
    - Dotnet framework 4 (or later)
    - Visual C++ Redistributable for Visual Studio 2015

- **Recommended software solution for real time convolution and playback:**
  - Ordinary playback from PC
    - John River Media Center with own Convolution plugin for Audio and Video playback
    - Foobar with Foobar’s own convolver.
    - Solutions exists to use the correction filters with Apple PC’s, but filters must be generated on a Windows platform.
  - OpenDRC, MiniSHARC

- **Recommended Microphone and Microphone Amp:**
  - Any calibrated quality microphone and microphone that you can connect to your Windows PC. Calibration files can usually either be uploaded in Audiolense directly or modified in a text editor before uploading. Use the sample mic cal file as a template.
Recommended Sound Card / DAC:

- Professional sound cards or DACs from quality providers such as Lynx Studio or RME for high quality playback.
- Any other DAC or sound card of preferred quality. Preferably with an ASIO driver. There are still bad drivers out there, but those who have ASIO drivers usually have good WASAPI drivers as well.
- Audiolense also works well with budget solutions. Embedded sound cards in stationary pcs often have decent sound quality.

Seven steps towards making and using a basic correction filter

The first times the Audiolense is run it should be operated in the following sequence:

1. Configure the speaker setup
2. Measure the speakers.
3. Filter the measurements.
4. Draw a target curve (or open existing targets)
5. Generate correction filters.
7. Load the filters into John River Media Center and play some music.
   - Get config or IR: from `c:\program files\juice\correctionfiles`. These works with JRiver and Sourceforge Convolver with a preset sample rate.
   - The config files from `c:\program files\juice\correctionfiles` supports automatic switch of sample rate and playback formats when used with John River Media Center.
   - The correction filters are by default made in the wav format with 32 bit floating point precision per default, but can be switched to 64 bit float and the OpenDRC format from the setup menu. They can be used with other convolution engines as well, but the configuration files are tailor made to Convolver.
   - See next chapter for more info.
Different playback methods

Several different playback methods are available. The only basic requirement is to have a convolution plug-in somewhere in the playback chain. Four usual methods are:

- JRiver Media Center with embedded convolver provides the most flexible and user friendly solution.
- External source + Bidule or Console or any other stand-alone VST host + Convolver plug-in.
- Foobar with Foobar's own convolver.
- Any other player or configuration that can host a convolution plug-in that can load filters in wav format.
- Audiolense filters can also be used on Mac / OSX and Linux. Ask on the forum for best practice.

Audiolense produces three types of files for usage with Convolver and other convolution plugins. Whenever Audiolense is being used, a file of one of these types should be loaded into Convolver or another convolution plug-in.

*.wav files (*.bin files for OpenDRC & MiniSHARC)

They are found in the Audiolense/Correction Files/ folder. They can be used directly in all decent convolution plug-ins available. If you are only correcting passive speakers and the number of speakers matches with the playback format, you can use this file directly. But remember, there will be one file for each sample rate.

*.cfg files (in the Audiolense/Correction Files folder):

These config files are sample rate specific and setup specific. It will be one file for 44100 Hz stereo playback and another for 48000 Hz 5.1 playback. These files can be used with Sourceforge Convolver in any plug-in host. These are also the ones to use with JRiver. JRiver will switch to matching samplerate if such filters are available. If not, JRiver will resample the correction filters whenever needed.

*.cfg files (in the Audiolense/Correction folder)

These files contain a correction filter list that enable automatic switching between different sample rates and different playback formats. They are at present irrelevant most of the time.

**JRiver Media Center (JRiver)**

This is the recommended method for audio and video playback. You can also set up JRiver to correct other playback sources on your computer. The following description is for Media Center 18. This version supports automatic change of sample rate.
Player -> Playback options -> Playback options:

Choose Output mode: ASIO if your sound card supports it. If your sound card doesn’t support Asio you can switch to Wasapi or use the Asio4All wrapper. Found here: [http://www.asio4all.com/](http://www.asio4all.com/) Other rendering methods may work well too. Select your sound card and output under Output mode settings.

Player->Playback options ->DSP Studio:
• You have to choose an output format that doesn’t have more output channels than what is supported by your sound card.
• If you have a stereo sound card and want to set your system up for 5.1 playback, you have to let JRiver downmix to 2.0 format. In which case you enter the setup menu in Audiolense and delete the 5.1 format.
• If you have enough output channels to support all playback formats you can choose whether the re-mapping is done by JRiver or Audiolense. Check the format tab in Setup ->Edit Speaker Setup or Setup ->Edit Playback Setup, depending on which version of Audiolense you’re using.
• You can generate filters in Audiolense with the appropriate sample rates, or you can let JRiver will resample the odd sample rate.
• Use no less than 24 bit or 32 bit Bitdepth in JRiver.
• There is drag and move functionality in the left window. Output Format and Convolution must be used. The various dsp functions are executed in the order they appear. Unless you have very specific needs and know what you’re doing, the Convolution should be at the bottom..
• Browse & load filter. We recommend that you choose a config file that supports 5.1 playback if your sound card has 6 output channels or more. This will work equally well for 5.1 and 2.0. If other formats are needed they can be made in Audiolense, under the setup menu.
• Keep the "Normalize filter volume" unchecked. Checking it will often lead to nasty sounding digital clipping.
• Check the Automatically switch filter … button.
• The audio correction has latency from a few milliseconds to several hundred milliseconds. JRiver has functionality for establishing proper lip sync. Refer to Jriver support forum and help file.

**MiniDSP – OpenDRC and miniShark**

Choose appropriate filter length.
Check that 32 bit float is chosen:
Choose Bin as correction file Format:
Sourceforge Convolver can be downloaded here:

http://convolver.sourceforge.net/

Sourceforge Convolver is an open source convolution engine that can be used with a number of plugin hosts and players. Further details about Convolver, including usage with other software players and usage with Asio drivers is on the same site. Sourceforge Convolver is not needed with JRiver.

Audiolense generates corrections based on four (five) inputs

1. Information about the speaker setup.
2. A measurement
3. A target
4. A Correction procedure chosen from the “Filter Measurement” button.
5. And lastly, a microphone calibration file if it is available. There is a samplemic.cal file available that shows the file format and text format required.

Once these 5 inputs are completed, there is nothing else that will influence the end result. So now you know.

Measurement

Audiolense uses the sine log sweep method. This is in our opinion the most accurate methods for impulse response measurement of speakers. It has a very good signal/noise ratio and harmonic distortion is separated from the impulse response. Mechanical noise from speakers, windows or other objects should be avoided. Fans, AC systems and other Choose a sound pressure level during measurement that produces the cleanest and clearest sweeps. The method is quite robust to moderate background noise and good corrections has been measured and made with people talking nearby. It is not very hard to achieve a good measurement. But for the very best result we recommend silence. The mandatory steps for a first time measurement are indicated in the figure below:

Before you start:

✓ Read the sound card’s user manual. 75% of all Audiolense getting started trouble shooting is sound card related.
✓ Turn the volume low before you press the “Run Measurement” button. Use your normal volume control.
✓ Place the microphone in the sweet spot – where someone’s nose will be when they listen to music. Use a microphone tripod or let the microphone rest on the back of a chair. Whatever holds the microphone should not conduct vibrations. Hook it up to the mic preamp and the sound card analog input.
✓ Try to avoid nearby (< 1 meter) reflective surfaces.

You are now ready to operate the measurement module.

An Asio based measurement is less likely to be subject to windows resampling and other interference from the operative system. Asio4All is a free Asio wrapper that can provide Asio support if this isn’t provided by the sound card manufacturer. Wasapi is also resampling safe. It will force you to measure with the same sample rate that is chosen as default in Windows.
**Step 1: Select the right sound card** (usually the same for recording and playback).

Several audio interfaces are available for playback and measurement. Most sound cards supports Windows MME and Direct Sound and some cards supports Asio. Sometimes there is a mismatch between sound card and the player in Audiolense. If one doesn’t work, try the next on the list. Use Asio if it is available and works.

Enter input channel if channel 0 isn’t used for input.

If you know that you’re using separate audio devices for recording and playback, or if you’re unsure, check the “separate play and recording device” option under advanced settings. That will prevent any clock drift related problems. When the clock drift correction is engaged, one of the speakers / tweeters will be measured twice.

**Step 2: Check the speaker connection if there are digital crossovers involved.**

- Note: You can select input channel
- Note: You can override the default output channels. This can be activated under the Advanced Settings menu. For Asio measurement and playback, and sound cards with analog and digital outputs, the digital outputs are usually numbered on top of the analog outputs. For instance; a sound card with stereo analog and stereo digital out will usually have channel number 0 and 1 for the analog outputs and 2 and 3 for the digital outputs.

**Step 3: Start a 3 second. Over and over until the following has been accomplished:**

- Adjust playback volume. Repeat until it is reasonably loud and as clear and clean as possible. Use your normal volume control.
- Adjust the input level from the microphone amp.
- Adjust the input level on the sound card. It should be loud and clear, but not over 0 dB.

**Step 4: Run measurement.**

Choose the desired sweep duration. 3 sec’s is good for step 3 and quick tests. Then complete the job by measuring for 5-10 seconds or as long as you wish.
The measurement module, with the 4 main steps indicated.

Warning: Be careful with the tweeters if they are being measured separately!

All tweeters have very limited low frequency capacity and some tweeters have very limited power handling capacity. So don’t measure them on a loud level and with a long duration. Tweeters may be measured with a sweep that is 10 dB lower than the rest.

All measurements are done at your own risk.
The difference between a valid and an unsuccessful measurement

Try to stay in the green area during measurement (below)

A measurement quality report will be presented right after the recorded sweep:

The measurement process is monitored with regards to noise level, input signal level, s/n ratio and dynamic range of the extracted impulse response. In addition, the potentially worst part of the recorded sweep is presented in the chart to the right. This chart will expose digital and analog clipping, microphone connection issues and a few other mishaps. In this very instance, the clicking noise from the laptop’s keyboard while taking a screenshot during measurement drove the microphone input to clipping levels and created a highly irregular sweep as well.

The three errors and two OKs at the bottom are templates of what to look for in the chart. The first is digital clipping and will be captured by Audiolense. The second has been clipped in the analog domain and you have to see that one. The third is a mess,
due to some resampling that Audiolense has no control over. The first OK sample is a clean, low frequency sweep, and the second is a clean high frequency sweep.

In addition to the measurement report it is advisable to examine the impulse response (below). A valid measurement has about 100 msecs' of silence, followed by a distinct peak and thereafter a tail with all the room reflections, resonances and diffuse field. It will always look very similar to the figure below:

Successful measurement. Remember to study the impulse response
Excessive pre-ringing before the main spike can indicate poor synchronization between playback and recording during the measurement process. There has been instances where the recorded stream has returned on a different sample rate than the signal that went out.
Measurement of timing

One of the important tasks of the measurement is to capture the relative timing of the measured speakers and drivers. To do this precisely requires an “all in one” measurement. This again requires that the microphone has a fixed position. In a multichannel setup, due to the directivity of the microphone, this procedure may lead to artificial measured high frequency roll off. This can be negotiated in two ways:

- Make individual targets for e.g. front and rear speakers – targets that accounts for the high frequency roll off.
- Measure the speakers individually or in groups, e.g. front speakers – surround left – surround right.
  - A brief all-in-one measurement that is taken just before the grouped measurement will capture the necessary time alignment info to for instance a speaker-by-speaker measurement.
  - In the batch filter generator it is possible to combine the individual impulse responses from one measurement with the timing information from another measurement.
  - The automatically generated timing info can be overridden manually by simply entering delay values in the measurement table and save the measurement thereafter.

The measurement of speaker polarity

All corrections in Audiolense are made such that all speakers and drivers have a positive polarity after correction. Therefore, polarity check and peak identification are important input from the measurement to the correction procedures.

There are a few speakers out there where the polarities of the speakers are ambiguous. By default, Audiolense detects the first significant peak. The first impulse response below is regarded as having inverted polarity, since the first significant spike is negative. The second – a woofer - has a first significant peak that is positive, and is thus regarded as having positive polarity. The timing of the two speakers is associated with the identified peak.

![Impulse Response](image)

The polarity check and peak identification will be effective in most circumstances. But under certain conditions and with certain correction procedures the peak identification may fail and the correction will be less than optimal. It is possible to disable the automatic polarity correction from the main form under the Measurement menu. When the automatic polarity correction is
disabled, Audiolense will work on the premises that all speakers and drivers are connected with correct polarity and will use the first positive spike for timing reference.

**Measurement through a system with significant delay**

During measurement, Audiolense expects that the speakers start to output the measurement signal at about the same time that the microphone recording starts. Audiolense can handle delay differences between the output and input of several hundred milliseconds automatically. Situations with extreme latencies can be managed by manually adding delay to the output or input stream in the measurement module, under the "Advanced" menu.

**How to make good-sounding filters?**

Three things accounts for 90-95% of the end result:

- ✓ A good measurement, where the speakers haven't been driven outside their comfort zone, and where the microphone was on and plugged into the same input as specified in the measurement form. Read about measurement above.
- ✓ A target that fits the speakers and your preferences. Target tweaking is very important to achieve a good result.
- ✓ If digital crossovers are applied: Crossover frequency and crossover width can have a profound impact on the sound quality.
- ✓ The final 5-10 percent is related to the correction procedure and is dealt with in a separate chapter.

We often get questions about why the filters attenuate the output so much. The short answer is that in order to avoid digital clipping, the average sound pressure level has to be attenuated by, typically 6-10 dB to give enough headroom for the single frequency that needs the most amplification. The frequency response of the correction filters – as they appear in the main form will give a 100% accurate picture of how the filters will attenuate the output. The frequency correction of the simulation will give the wrong picture of the same.

If you see excessive attenuation in the correction filter, you should look for anomalies in the frequency domain.

**Target**

The key success factor in finding the best target is to experiment and to judge by your ears. But here are a few guide lines.

The figure below shows four different targets:

* A neutral target (blue), a popular target shape (red), a neutral target that rounds off in the top (green) and a target that is flat through the lower midrange (grey)
All these four shapes have merits, depending on preferences and time spent with Audiolense.

The blue target usually sounds neutral. A straight line, falling 1-2 dB from 100 Hz to around 10 kHz. Usually it pays off to be more diligent above 10 kHz than I’ve done here. Sometimes a slight rounding off, following the natural tendency of the tweeter above 10 kHz sounds best. The green target is quite similar to the blue, but with a shape in the top that follows the tweeter’s
round-off in the top. The default setting in the correction filters will deal effectively with both, but the results may be slightly different.

The red target has a shape that is preferred by a lot of users. A fairly flat shape through the midrange and a bass lift, starting somewhere down in the upper bass.

The gray target is in some ways the opposite of the red. Flat from ca 50 Hz to 1-2 kHz and a downslope from there. Preferred by a few professional users and long timers.

Going all the way from the familiar house sound to a 100% neutral setup will often not a satisfactory result. It can change the sonic signature to a degree where you don't recognize your speakers. A good point of departure when starting with Audiolense can therefore be a target (not depicted above) that is somewhere between neutral and the response of the uncorrected speaker. You will appreciate a somewhat smoother frequency response but still recognize the signature of your old speakers.

The overall slope of the target has more impact on the warmth + bass pressure vs. brightness + air dimension, while the curvature has a large impact on the timbre and the sound stage. The less curvature through the pass band, the more neutral timbre you will get. The red target (above) may sound colored on male voices, rhythm guitars etc if you're sensitive to that sorts. But it will also produce a large sound stage, and perhaps also a live like sound stage like what you will hear on e.g. a rock concert.

Good bass performance is achieved with a reasonable nice and smooth rounding off. If the target looks like how a boom box vented alignment measures in location, it will sound like one too. If the target is similar to a low Q sealed alignment it will sound like one. The shape of the curve in the bass is absolutely worth experimenting with. Sometimes you can deepen the bass without degrading the sound in general. Other times not. And sometimes you can make it perceptually deeper by starting the round-off earlier and rounding it off more gently. The speed of bass is centered clearly above 100 Hz. If you want “fast” sound you need enough energy around 2-400 Hz.

Most of the time the treble sounds pretty good if the correction does not work against the natural slope of the tweeters above, say 5 kHz. The slope in the top end can have a huge impact on long term listening enjoyment and the sense of air. When the voicing is generally good it may be worthwhile to do some listening-based fine tweaking in the top. Usually it is possible to achieve subtle but significant improvements over the uncorrected or minimum corrected tweeter.

Note that all targets drawn level out towards 0 Hz and the top frequency. This is recommended, since steep slopes at the extremes sometimes will lead to less precision in the calculations.

**Crossover design**

The crossovers in Audiolense are defined in terms of the octave bandwidth of their overlap region. Narrow width corresponds with steeper, high order filters and wide overlap corresponds with lower order crossovers.

The crossovers in Audiolense have two special features: They are linear phase by default – for perfect integration and timing. And they are optimized to enable minimum time domain ringing for a given steepness. All crossovers have ringing in the time domain, and the crossovers in Audiolense performs much better in this regard than conventional crossovers with similar steepness.
Crossover design is all about making the best use of each driver and making them blend as seamless as possible. Different drivers have different strengths and weaknesses and there is no universal best practice. Audiolense provides a very flexible crossover framework. Basically each low pass and each high pass section can be specified independently of the rest of the makeup. And full / partial driver overlap is supported. Here are a few general crossover properties to consider.

Narrow width filters provide:
- A reduced workload to the driver, so it can be utilized closer to its frequency boundaries. A tweeter can for instance be high passed at a lower frequency.
- An improved vertical off-axis response. Particularly at high frequencies there is a tendency for in phase / out of phase variations that will hit the ceiling and floor before it arrives at the listening seat.

Greater width filters provide:
- An overlap region that sometimes makes up for a sonically more seamless integration.
- A smoother horizontal off-axis response. Relevant in cases where the low frequency driver has a reduced dispersion compared to the high frequency driver at the crossover frequency.
- Better time domain behavior. Filters with a wide overlap region do the job faster. That means less ringing before and after the impulse response peak. This is illustrated further down.
Two alternative crossover configurations. 2 octave width @ 200 Hz and 3 kHz (red) and 3 octaves/200Hz – 0.1 octave/2kHz (blue).
The time domain behavior of the two sets of crossovers is depicted in the two following figures.
Time domain behavior of the left speaker with 2 octave crossover width. Zoomed around zero to reveal the activity before and after the peak.
Time domain behavior of right speaker. Wide 200 Hz crossover that is very fast and narrow 2 kHz crossover that has more ringing.

The wide 200 Hz cross has a very good time domain behavior, as can be seen by the low pass curve (red). All the ringing is from the very steep 2 kHz crossover. This figure has the same scale as the one above.

If the drivers had no distortion free they would cancel each other's ringing, and the added result would be a perfect impulse response. Drivers are hardly perfect, but the major part of the ringing will be canceled because drivers are mostly linear and recently low distortion in their behavior.

The tradeoff between steep filters and fast filters is a universal tradeoff. The Audiolense filters compares very well in this regard. They are relatively shallow above the -6dB point - which is the part that has most influence on the time domain ringing. And they are steep below the -6dB point, which provides a very efficient offloading of the driver outside its operating range. A two octave width filter is very steep compared even to most digital filters, but it is almost as fast as a good 1st order filter.
Advanced topics (Audiolense XO)

Correction Procedure Designer

The Audiolense 2.0 version and Surround version is equipped with predefined correction procedures. The procedures are capable of very good frequency correction that compares very well to most room correction solutions available. Three variants - with varying degree of correction - are available. Audiolense Surround has the three above correction procedures with linear phase crossovers, and an additional 3 procedures for minimum XO latency. The latter filters have the smallest possible delay. They compare well to most filters, but are not quite as good as the default which uses linear phase crossovers. The minimum xo latency filters are there for situations where a very short latency is required. With these settings, latency that approaches live audio is possible.

Since Audiolense 3.0, a psychoacoustic interpretation of the frequency response is being used for the frequency correction. This has been a very successful feature and has now fully replaced the earlier direct sound approach.

Audiolense XO version comes with three predefined procedures, including a true time domain correction. But it also has a Correction Procedure Designer that enables the user to define custom procedures in order to squeeze the last percentage of sound quality out of the system. See below:
Correction procedure designer. Note that the designer is only included in Audiolense XO. Prevent treble boost and prevent bass boost are default in Audiolense 2.0 and Audiolense Surround.

**True time domain Correction**

The True Time Domain correction Cleans up the impulse response in the time domain and corrects the speakers towards the time domain behavior in the target.

A good frequency correction goes a very long way in this game, but usually time domain correction improves the sound quality even further.

In most setups a true time domain correction will provide better perceived dynamics, 3D and air than the alternative. The difference is clearly audible in A/B comparisons, but e.g. new users of Audiolense 2.0 tend to describe the sonic improvements in a very similar way as those who listen to time domain correction with Audiolense XO.

When digital crossovers are handled by Audiolense, the correction is made in two steps. First each driver is time frequency corrected individually, then the sum of the individual drivers are corrected as a combined unit.

When True Time Domain Correction is checked, the whole speaker will be time domain corrected as one single entity. This is the only TTD option for passive speakers. The option “TTD Correction per driver” will also be visible when digital crossovers are used. Checking this option will also time domain correct each driver individually before the speaker is assembled and corrected as a whole. The experience so far is that it varies from system to system whether this option is beneficial or not.

**Filter Length**

Short filters can sometimes be a bottle neck with regards to low frequency control. A 65536 taps will provide a frequency resolution of 0.7 Hz at 44.1 kHz sampling frequency and will usually provide generous headroom as far as filtering capacity goes. At the other extreme, a 4096 taps filter has a frequency resolution of around 16 Hz @ 44.1 kHz sampling frequency. This will usually give very precise to just below 100 Hz. It will usually give significant improvement down to 20 Hz for sample rates of 48 kHz or 44.1 kHz, but the precision will be lower in the deep bass, and there may be instances where 4096 taps is less than desirable. OpenDRC and MiniSHARC uses 6144 and 10240 taps respectively. Simulations with filters of this length always show very good results down to somewhere below 100 Hz and reasonably good control further down. while 6144 taps is safe for a full range correction for sample rates of 48 kHz and lower, the number of taps may become a limiting factor at higher sample rates of if digital crossovers are to be included in the filter.

A short filter may affect the quality of the crossover – particularly how steep and how deep it cuts off the stop band. This may not be an audible issue in most cases, but crossovers that are steep and low frequency does take their time to complete the job.

Be aware that a very long filter combined with short windows (see below) will only keep the computer busy calculating a lot of zeroes. The filter should be long enough to provide enough room for the correction window and crossovers. The crossovers are
8192 taps by default, they will be allocated their own space in the correction filter if the filter length is minimum 32768, something that increases the precision of the correction and the stop band behavior of the crossovers.

**Boost limiting**

*The Max boost setting* determines how high a dip is allowed to be lifted by the correction filter. A too high setting will consume the dynamic range of the setup. Typical setting is below 12 dB and 6 dB is a good figure most of the time. Higher values work – and sometimes a higher value is needed to get the desired result. The correction boost will also be limited by the measurement and correction window – and quite often to an extent where a 6dB max boost setting hardly makes a difference.

*The Prevent treble boost* function will prevent any treble boost above ca 15 kHz. A target that is flat out to max frequency will not lead to excessive treble boost as long as this function is enabled. The max boost setting applies if this is unchecked.

*The Prevent bass boost* function will basically prevent bass boosting below the region where the bass has significantly dropped in sensitivity. The algorithm involves some fuzzy logics and may not hit the bull’s eye in all setups. In any case we recommend a target with a proper roll off in the bass for the best possible performance.

**Minimum Phase Crossovers**

Basically all analog crossovers have a minimum phase character. And they all have phase shifts. And this causes the low pass and high pass sections to be somewhat of phase through the crossover region. The advantage is basically zero pre-ringing and minimum time delay in the generated filters. Latency inside 10 ms can usually be achieved, and with careful speaker placement and crossover tweaking 5-6 ms should be within reach. Using minimum phase crossovers without TTD correction will – together with with 16 partitions in Sourceforge Convolver - will usually be enough to achieve good lipsync on video playback from external sources.

**Measurement & Correction Window**

Looking at the first tab in the procedure designer, there are two tables for measurement and correction windows. (With the True Time Domain option disabled, only the top table will be applied and visible).
You can customize the measurement & correction window. Shorter windows will lead to less correction and less detailed correction. The correction filter will become shorter and simpler. Longer window will have the opposite tendency. This window influences the smoothness of the filtered frequency response and defines the effective duration of the correction at any frequency.

You can choose the size of the time window in the bass (here: 10 Hz) and in the high treble. You can also specify an optional mid frequency and the associated duration. This will often come in handy if you wish to progressively increase the scope of the time domain correction towards the lowest frequencies. You can also adjust the low frequency (here set to 10 Hz) can also be adjusted. This is the end point of the window, and anything that falls outside (in this case 800 ms). If we change the low frequency to 100 Hz, the filtering will stop at 80 ms.

One more thing to take note of: When the per driver option is unchecked, each individual driver is only frequency corrected. Here's how these filter functions work in the time domain:
Time domain filtering with 3 cycles low and 2 cycles high. This is what the TTD Correction “sees”

The TDW filtering also leads to frequency domain smoothing:
Same filtering as above. What we see here is the frequency response produced by the first 3/2 cycles that reaches the sweet spot.

This time domain filtering is used on both sides of the impulse peak when the situation calls for it. This is the case when TTD or linear phase crossovers are used and the filters are being finalized.

If we used the above frequency response as a basis for frequency domain correction, we would not get the best possible result. Later arriving reflections have an influence on the perceived frequency response, and sometimes quite substantially. Therefore, a more psycho-acoustically correct frequency smoothing technique is used in combination. As a result, this is what Audiolense sees and corrects in the frequency domain.
A frequency smoothing based on psycho-acoustic principles leads to a smoothed response that sits high in the comb filter region and avoids overcorrection of dips.

The measurement & correction window helps smoothen the filtered measurement in the frequency domain and determines how much time the correction filter is given to complete the job.

The settings also affect how the frequency response is smoothed before it is corrected. The TDW is used together with other methods to smooth the frequency response. The ultimate goal of the procedure is to obtain a smoothed response that correlates well with what we hear and what we would want to correct. A longer TDW will smooth the response less, and hence the correction will be more detailed. A shorter TDW will provide an even smoother response which will also result in a more moderate correction.
The size of the measurement & correction window is not where you need to spend a lot of tweaking time. These window settings are usually not very influential on the sound quality. But this window should always be larger than the TTD subwindow when TTD is selected. 2-3 cycles extra is usually a good figure.

**The true time domain sub-window** defines a true time domain window that will be time domain corrected. More about that down below, under pre-ringing prevention.

The chart below shows good symmetry between the filtered measurement and the correction filter – and a smoothed simulation that behaves as we wish.

The smoothed evaluation is as close to a smoothed measurement after correction as it gets. First, the raw measurement is corrected, using the correction filters. Then the corrected measurement is smoothed and filtered the exact same way as the raw measurement was handled. Apples to apples, and also practically identical to what you will see in a properly performed, similarly filtered control measurement of a corrected speaker. This is how we want it to be. Depressions in the filtered measurements that are only partially corrected are also healthy and normal.
Pre-ringing prevention
Audible pre-ringing is a recurring problem with TTD correction. This part addresses what it is and how to prevent it.

What it is
Enter from the main menu Advanced TTD Correction -> Test TTD Behavior.
This function has been run, and is shown below.

Group delay of time domain filtered measurement (red) and group delay of the TTD prepared measurement (blue)
The red graph is the filtered measurement, the blue is filtered and prepared for TTD correction. The highlighted peaks can cause audible pre-ringing. This test is not 100% reliable. Sometimes, with full range speakers, the actual results turn out better. Digital crossovers involves additional processing that can lead to problems even though they don’t appear here.

Below, we have made a correction with the same settings:

This is how the simulated result looks:

To take a closer look, we can study the step response (from the Analysis menu)
Step response simulation and target with 3/5 TTD Window, selective prevention disabled.

The above result – with a generous TTD window and without selective pre-ringing prevention - produces a corrected impulse & step response that follows the target very closely – as it should. In this specific instance, the pre-ringing we see is audible only on very rare occasions.

If you hear pre-ringing you should also examine the actual correction filter(s):
H2 Make correction procedures that prevents audible pre-ringing

It is often a good idea to examine the correction filter itself. Significant distortion levels are not uncommon in speakers below 100 Hz. Sometimes, audible pre-ringing may occur even though the simulation looks good. Audiolense does not account for noise and harmonic distortion that may accompany an otherwise well-functioning correction. Whenever that is the case, filters with very little activity before the main peak is needed.

Same setting, with selective preringing prevention:
Selective preringing prevention checked

A result as shown in the chart below can either be achieve by shortening the TTD window or enabling the selective pre-ringing prevention. And you should tweak the correction procedure until you all peaks have been replaced by notches.
Most likely successful pre-ringing prevention here.
Step response with 5/3 TTD window and selective preringing prevention. Looking good in this case.

The corresponding correction filter is shown below:
Correction filter 5&3 TTD window and selective pre-ruing prevention.

The above chart zooms in on the left side of the correction filter. It clearly shows that problematic resonances that lead to excessive pre-ruing in the correction filter has been successfully removed.

**Remove noise from the measurement**

From the main form select Advanced TTD Correction -> Manually remove measurement noise.
Noise removal in front of the measurement peak gives the TTD correction an easier task, and improves the various simulations and evaluations substantially. The reset buttons are usually very precise. You can zoom in on the pulse and click with the mouse to define where the real signal starts. Get this point as close to the main peak without altering the frequency response substantially. Blue indicates IR and FR after noise removal, red before.

**Final advice regarding pre-ringing prevention**

The pre-ringing prevention that is now implemented is new functionality in Audiolense (January 2017), and it will take some time before we have a wide experience base about situational best practice. All systems measure differently and most systems have quirks. A limited set of speaker measurements have been used during the development. The one shown here is easy to work with, but also has a couple difficult frequencies. A 3 way speaker from the same room has also been used, and a (before this) very difficult to correct 5.2 system with horn speakers in a more lively room. These three systems seem to respond well to the selective preringing prevention.
A good place to start is 5/3 window or 3/2 window – and test with and without the prevention enabled. Extremely short TTD window can be used. They will not look as good on simulation, but if the window is made short enough (1 cycle or less), any pre-ringing should be inaudible.

Digital crossovers has additional challenges. This is especially the case when bad reflections occur in the crossover region. So pay attention to where you place the crossovers (and where you place the drivers). Cancellations in either can lead to problems if they are strong enough. Another trick that may help is to try minimum phase crossovers. Minimum phase crossovers will alter the relationship between low pass and high pass, and sometimes this helps.

### Partial Correction

The second tab in the procedure designer is assigned to partial correction. Partial correction is activated by the checking the option in the “Filter options” section. Below it selected on both speakers of a stereo pair. No correction above 15 and 3 kHz, respectively.

<table>
<thead>
<tr>
<th>Speaker</th>
<th>Partial Correction Speaker</th>
<th>Partial Stop Frequency</th>
<th>Transition Width (Octaves)</th>
<th>+/− Amplification (dB) of Uncorrected Frequencies</th>
<th>TTD Stop Frequency</th>
</tr>
</thead>
<tbody>
<tr>
<td>Front Left</td>
<td>✓</td>
<td>15.000 Hz</td>
<td>0 Octaves</td>
<td>0.0 dB</td>
<td>15.000 Hz</td>
</tr>
<tr>
<td>Front Right</td>
<td>✓</td>
<td>3.000 Hz</td>
<td>0 Octaves</td>
<td>0.0 dB</td>
<td>3.000 Hz</td>
</tr>
</tbody>
</table>
The target curve can be used as an equalizer in the no correction zone. This was not chosen here.

Partial TTD is a new feature in Audiolense 5. It will typically lead to less correction activity in the filters. If you e.g. sees too much high frequency correction in the filters, you can try a 15kHz setting like above. Also, it is now possible to frequency correct all the way up, and stop the TTD correction at a lower frequency. Partial TTD will lead to less perfect looking simulations and may also introduce low level artifacts. If the tweeter already has a close to minimum phase output (not uncommon on a flat baffle), you may get a cleaner correction filter by not using partial TTD. OTOH, the partial TTD can be a very effective way to negotiate non-minimumphase behavior on horn-loaded tweeters.

Result of partial correction 15 kHz and 3 kHz, respectively

When digital crossovers are used, we generally recommend that the no correction zone starts above the highest digital crossover point. The drivers will sum perfect at the crossover point, but due to phase shifts in the drivers, it may not sum
perfect right beside. This can be seen in the figure above, where correction only is applied below 500 Hz. The evaluation shows a clear and wideband recession on the left side of the 3.5 kHz crossover point.

Partial correction is pretty straight forward as in a frequency correction procedure. There are more pitfalls when partial correction is combined with TTD correction. A separate analysis has been devised to examine how the measurement is manipulated prior to correction. It is found under the Advanced TTD Correction Menu and is labelled Test TTD behavior and frequency mod. This display shows how the measurement is moderated prior to being corrected. It show how it is moderated in the frequency domain and time domain, and thus includes the dip boosting, the no correction zone and the pre-ringing prevention. Note that this display has the same limitations as the Test TTD Behavior display that was presented earlier. The hard parts of the operation only operates on the speaker combined unless TTD per speaker has been checked. Anyway, here are screen dumps that displays how full range speakers with 6 dB dip limiting and no correction above 15 kHz is modified:

Procedure used
Result of the measurement pre-processing. Left speaker only

The frequency responses shown here are used for the time domain correction only, and do not include the psychoacoustic function that is always used to get close to a correct perceived frequency balance. We can see that the dip limiting and the no correction above 15 kHz are in effect. Also, the deviation in the low frequency roundoff reflects the correction limits that will be in operation in the low frequency stop band and pass band.

But the time domain view is where it gets interesting:
Time domain view of the measurement pre-processing with partial correction. Red is windowed measurement, blue is pre-processed.

The function isn’t 100% perfect as there are small traces of pre-ringing in the pre-processed signal. This is nevertheless a promising start. What you should look for here is an impulse that is less complex than the original – an impulse where very little high frequency correction will be needed. Sometimes a bit of experimentation with the partial frequency and transition region is needed to get a purest possible pulse here.

**Multi seat Correction**

The last tab on the menu (below) is assigned to multi seat correction.
Multi seat correction tab

Only measurements stored in the default measurement folder will be available for multi seat correction. The subject measurements are dragged and dropped between the two containers above, so the user can choose which measurements to include in the multi seat pool. The multi seat pool is common for all multi seat corrections. Each measurement is stored with a parameter indicating whether it belongs to the pool or not.

Before we proceed, let’s click the “Show multi seat impact” button:
Estimated impact of multi seat correction for all seats (measurements)

This chart shows how each measurement or listening position will be affected by not being fully corrected as a pure sweet spot correction. A straight line indicates that the result will be the same as for a single measurement based correction. We have selected the "bak" measurement to be the sweet spot measurement here, and as we can see, it will hardly be compromised by the other measurements in the pool. This is due to two factors:

1) We have checked the "ignore worst case" option

2) We have applied a sweet spot preference of 3 dB, which favors the sweet spot at the expense of the other seats.

The "ignore worst case" option ignores the worst case(s) at any frequency. How many worst cases are ignored depends on how many measurements are in the pool. 3 or less measurements in the pool and they are all accounted for.

The multi seat settings can be changed without closing the estimated multi seat impact window. After unchecking the "ignore worst case" and setting sweet spot preference to 0 dB we get this result:
Estimated impact of multi seat correction for all seats (measurements) Here with "ignore worst case" unchecked and 0 dB sweet spot preference.

With this setting we get rid of the most annoying frequency deviations: The dreaded peaks. No seat is exposed to frequency peaks anymore.

Multi seat correction is the art of compromise. Measurements in the pool that are very different from the rest will affect the frequency response for all the rest – after correction. The relative differences between the listening seats will be the same after correction. But they are all likely to benefit from the same correction, and if the multi seat regime is well managed, the audibility of the internal differences may be substantially reduced. Choosing which measurements to include in the pool, sweet spot preference, the “ignore worst case” option - and the “Estimated Impact” view are the tools for finding the best overall solution.
In accordance with the multi seat correction, a multi seat simulation function is also available from the main form. It can be enabled for automatic generation by pressing the simulation hotspot in the main form. Or it can be generated directly from the Correction menu after the correction has been generated. Here’s how it may look:

Multi seat simulation with “Ignore worst case” disabled and 0 dB sweet spot preference

As can be seen in the simulation above, the end result doesn’t get quite as good as suggested by the estimated impact. This is due to the time domain restrictions that are imposed on the correction filters.

Note that this multi seat simulation can also be used on single spot corrections – as long as a few measurements have been entered into the multi seat pool.
Minimum phase, linear phase, mixed phase time domain correction

A "minimum phase" time domain correction, will give the fastest possible rise time and a longer settling time for the early part of the decay. A "minimum phase" time domain correction is made by combining a minimum phase target with a True Time Domain correction. A linear phase correction will correct towards linear phase behavior where all frequencies hit the listener at the same time with the same phase. A mixed phase is anything between the two others.

The illustration below shows the impulse response of a minimum phase target:

Here’s the same target, only in linear phase version:
Linear phase target impulse response

Observe the fast rise time with the minimum phase target and the perfect symmetry with the linear phase target. The advantage of linear phase is that all frequencies arrive at the listener in phase and at the same time. The disadvantage of linear phase is a slow rise time. The minimum phase has fast rise time but it has phase distortion close to the high and low extremes of the speaker's bandwidth.

The step response magnifies what happens before and after the main peak of the system, but it also depicts better how the system responds if a sudden broad band burst (a square wave like sound) is played back through the system.
Step response of minimum phase target

Step response of linear phase target
Observe that the step response is plotted on a longer time scale than the impulse response.

The band limited nature of real world speakers is such that you can’t have perfect rise time behavior and perfect phase behavior. However, if the music is more bandwidth limited than your audio playback system you can have a more timely behavior than these graphs indicate. This makes a good case for bandwidth that exceeds below 20 Hz.

All targets are prepared in the target designer. Mixed phase option (highlighted here) is activated from the top menu.
Hidden functions and details you might like to know

The user interface and the program logics are organized to be as intuitive as possible. But there are a few not-self-documented functions.
All charts have a zoom function. Click, hold and drag the mouse from top left to bottom right of the zoom area.

Zoom out by click, hold and drag up and left.

All charts can be edited. Double-click to enter the editor. (Double click with the right button in the Target Designer).

The targets (in the target designer) are best drawn with mouse and keyboard

- Mouse point and click to add a point in the target curve
- Use mouse to select active target point
- Hit “Delete” button to delete active target point
- Use arrow buttons to move active target point.

The evaluation graphs (simulation) in the main form are very reliable. They have been benchmarked with true measurement of corrected speakers, and the result is very close to identical. The delay shown by the evaluation is the true delay from filters and speaker driver alignment.

Bass management. Be aware if the wav correction filters are used directly that the bass management is not just performed by the subwoofer correction filter. If bass management is involved, it is highly recommended to study the config files generated to be used with Convolver. They are written in text format and shows which input goes through which filter and to which output - and sometimes attenuation may be involved as well. The easy solution is of course to use Convolver with the config files.

A subwoofer is a speaker in its own right. And the subwoofer correction filter is valid for that speaker as a stand-alone – to correct the LFE channel. In addition the sub is often used to play frequencies for the main speakers. Audiolense generates separate correction filters for the two tasks.

The subwoofer will use the same "offloading" correction filter to offload bass from all speakers that has the same crossover. This offloading filter will be optimised towards the first speaker using it. In certain situations, this filter will not produce optimal results for all speakers. This can be negotiated by applying slightly different crossover values e.g. 80.1 Hz instead of 80 Hz, and thus get another tailor made sub woofer integration for the subject speaker.

It is possible to use several subs to offload from a speaker. But any speaker that shall be offloaded should be categorized as a "Small" speaker.

A speaker categorized as "Fullrange and sub" can also be used to offload Small speakers.

The offloading part of the bass management can be specified on a speaker by speaker basis - where each speaker is offloading to one or more sub woofers. Alternatively, the bass can be specified to be routed to all subs from each of the Small speakers.

The surround version uses wider crossovers for the minimum delay correction in order to minimize the total delay in the playback.
✓ **Crossover alignment** (only relevant when crossovers are used) When frequency correction or minimum delay crossovers are used, or partial correction is enabled, some of the drivers will usually be time-adjusted to provide the best possible driver integration at the crossover frequencies. This adjustment does not apply to full range time domain correction where all frequencies are time aligned anyway.

✓ **Partial correction / no correction from 0 Hz.** A correction with no correction from 0 Hz on all speakers will simply produce a combination of crossovers and targets, and time alignment and crossover alignment is disabled.

✓ **Crossovers in Audiolense XO** You can basically specify any crossover configuration you like. Audiolense will adjust crossover filters automatically so that it all adds up to 0 dB.

✓ **EQ functionality:** If partial correction is used, the target curve can be specified to function as an equalizer in the region where no correction is applied.

✓ **Output channel order can be manually overridden:** This function is enabled in the measurement module, from the “advanced” menu. The generated correction will be based on the same channel order that was used for measurement.