

Anti-Mode 2.0 Dual Core Owner's Manual



Recycling information

The product you have purchased is marked according to the **Waste Electrical and Electronic Equipment Directive** (WEEE Directive).

There are take-back systems in place that help to preserve nature and natural resources when products are disposed of appropriately.

If you need to dispose of this product, use the take-back system that

has dedicated collection facilities for electronic equipment. Do not put the product into household waste disposal!



Also, the product has been manufactured using parts and processes that follow the directive of the Restriction of the use of certain Hazardous Substances in Electrical and Electronic Equipment (RoHS).

Intended Use

This product has been designed for normal indoor use and to be connected to other equipment with cables not exceeding 3m (10 feet) in length. If you use cables of extended length, check that their quality is sufficient and observe electrostatic discharge precautions when connecting or disconnecting them. Use of the device outdoors, in humid or other extreme environments, may cause reduced performance or risks to the user of the equipment.

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1. Introduction

You are in possession of a remarkable audio device capable of – no less than – making wonders to the sound quality of most audio systems. The list of the advanced features of the Anti-Mode 2.0 Dual Core is long; including jitter-less digital output, audiophile-grade D/A converters with digitally controlled analog volume control, etc. But perhaps the most important feature, regrettably commonly overlooked by the industry, is the ability to correct the room resonances and other adverse effects generated by the listening environment. Perfection in sound can only be achieved through the mastery in the art of room acoustics!

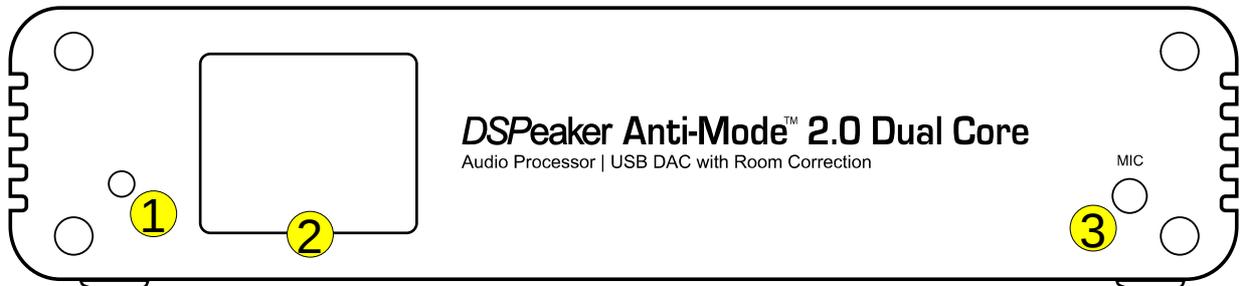
We have worked hard to ensure that the Anti-Mode 2.0 Dual Core is not only powerful and versatile but also intuitive and easy to use. However, it is a good idea to read this manual in whole before operating the unit.

1.1 What's included in the box

- ✓The Anti-Mode 2.0 Dual Core
- ✓A power supply unit suitable for your country / region
- ✓A calibration microphone
- ✓An infra-red remote controller
- ✓A USB cable
- ✓This owner's manual

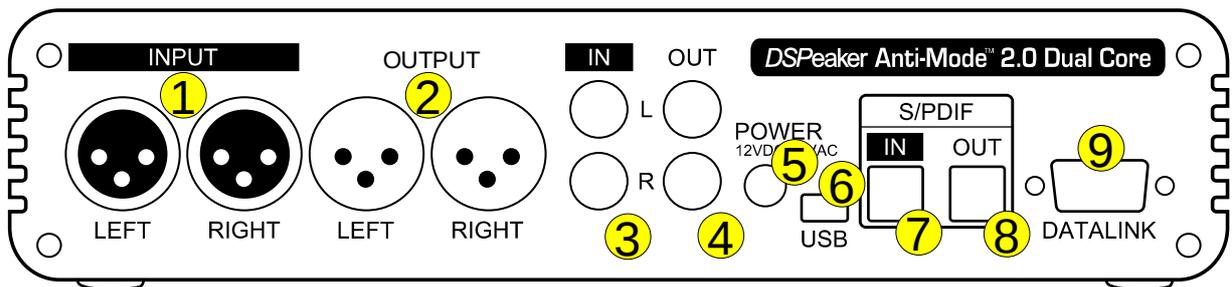
2. Overview

2.1 Front Panel



- (1) Infrared sensor
- (2) TFT display
- (3) Calibration microphone connector

2.2 Rear Panel



- (1) Analog balanced XLR input*
- (2) Analog balanced XLR output
- (3) Analog RCA input*
- (4) Analog RCA output
- (5) Power supply input (12VDC or 9VAC)
- (6) USB connector
- (7) S/PDIF Toslink input
- (8) S/PDIF Toslink output
- (9) DSPeaker datalink connector

*) see the warning below.

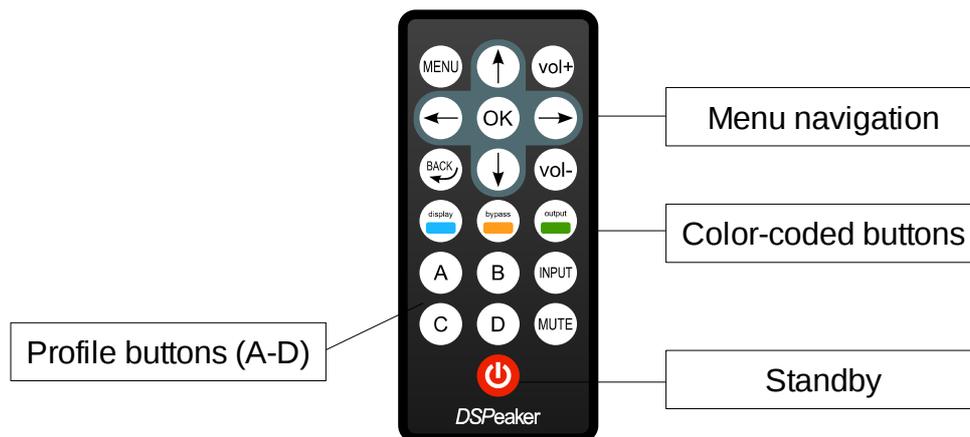


IMPORTANT: Never connect both RCA and XLR inputs (1) and (3) at the same time! The analog INPUT connector types are meant to be exclusive alternatives to each other.

2.3 Getting Started

Anti-Mode 2.0 Dual Core can be utilized in many different kinds of configurations. Before powering the device, please connect all the inputs and at least one of the outputs that are relevant to your system.

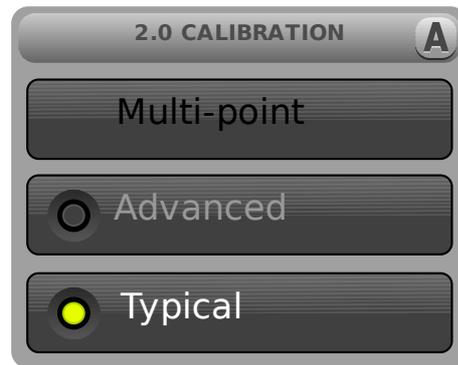
Virtually all of the user interaction is carried out by using the remote controller. Below are some commands that are commonly used throughout the interface:



- The **MENU** button brings out the main menu (press **MENU** again or the **BACK** button to return back to home screen from the main menu)
- Directional arrow buttons change the active menu item
- The **OK** button accepts any changes made and advances to the next screen
- The **BACK** button returns to the previous screen
- The three color-coded buttons (**blue**, **orange** and **green**) select similarly colored items in the various EQ editor screens. The primary actions of these buttons are active in the home screen only (**display**, **bypass** and **OUTPUT**.)

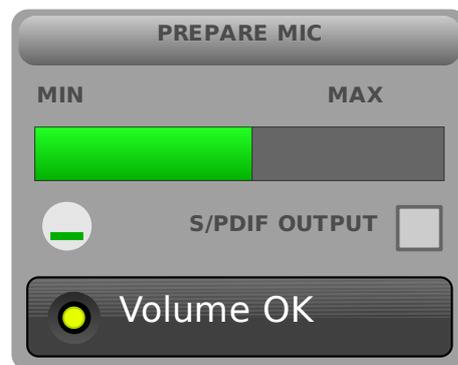
When the device is powered up for the first time, the language selection screen appears. Use the arrow buttons on the remote to select the language and press **OK** to confirm. If needed, the language can be changed later from the Settings menu.

It is highly recommended to perform the room calibration process right away. Follow the steps described in the next “Typical Room Calibration” chapter.



2.4 Typical Room Calibration

- Press **MENU** on the remote controller. The main menu screen appears.
- Use the arrow buttons to highlight *Calibration* menu item and press the **OK** button to select it.
- Press **OK** to accept the *Typical* calibration mode (The *Advanced* mode allows more options to choose from. That and *Multi-point* calibration mode will be described later in this manual.)
- The speaker configuration screen appears. Use the arrow buttons to select the option that best matches your speaker setup. If you are unsure, select the *2.0 Stereo* configuration. Press **OK** to confirm the selection.



- The microphone check screen appears. At this point, plug in the provided calibration microphone to the microphone jack and position it at the primary listening spot. The microphone should be positioned at ear height, with no reflective surfaces near the tip and fixed in place.

Do not hold the microphone or the microphone cable in your hand during the measurement! Rather find a secure position where the microphone stays intact throughout the whole calibration process.

- Make sure either analog or digital output is connected to powered speakers or subwoofers. If the analog output is used, a test tone should now be audible.
- If needed, press the **OUTPUT** button on the remote to change to digital output. Please consider the warning below prior to doing so.



WARNING: In order to preserve the dynamic range, digital output signal is never scaled. Consequently, the digital output level is always at 0 dB (i.e. maximum.) When using the S/PDIF output, please be sure to **adjust your amplifier's volume controls** to a moderate level in order to avoid damaging your ears and/or your audio equipment!

- Use the arrow buttons or the **VOL+** and **VOL-** buttons to adjust the calibration volume (analog output only).
- The field at the lower part of the screen displays "Volume OK" when Anti-Mode detects a sufficient volume level for a reliable calibration.
- If you can't hear the test tone (hissing noise), please make sure the output is connected and the amplifiers or powered speakers are functioning properly.
- The provided volume indication is a recommendation. Sometimes even lower volume level may be sufficient for a decently accurate measurement.
- When you are ready to start the calibration process, press the **OK** button to proceed.



FOR BEST RESULTS: When calibrating a subwoofer system: 0.2 Stereo, 0.2 Dual-Mono, or a single subwoofer, make sure that the built-in lowpass filters of the subwoofer(s) are bypassed or set to the highest cutoff frequency. You can restore the setting after calibration or use Dual Core's Lowpass Filter Tool.

- Depending on configuration, the calibration procedure may start with a speaker distance detection. If the distance detection is successful, room acoustic analysis begins immediately after it.
- During the acoustical analysis phase, Anti-Mode generates several frequency sweeps. After the first sweep, the initial room response graph is shown in red.
- It is normal that the analysis phase lasts several minutes.

- You can cancel the calibration process at any time by pressing the **BACK** button. This will revert the room correction parameters back to their previous calibration state. (However, possible new speaker delay values from the distance detection step stay in effect. Please note that this behavior may change in a future firmware revision.)
- Once the analysis phase is finished, the corrected room response is shown in black. The limit of the automatically corrected range is displayed with the vertical dashed bar. Press the **OK** button to dismiss the screen. (Both the before and after graphs can be viewed later in the *Room Response* menu accessible from the Main menu.)
- Congratulations! Your system is now optimized for the most accurate sound. Press the **MENU** button (or the **BACK** button) to return from the main menu to the home screen.



PLEASE NOTE: After Anti-Mode gets rid of the room resonances, it may appear the result sounds good but is lacking some bass. One possible reason for this is that the listener has grown used to the exaggerated bass response previously caused by room resonances. If you feel that you need to increase the overall bass level, you can do this using the house curve tool or designing a custom EQ to lift the low-frequency band.

3.Home Screen



The default home screen view consists of three components. The currently active input is displayed at the center.

➤The active input can be switched by pressing the **INPUT** button on the remote.

Similarly, the active output is shown at the bottom of the screen.

➤Press the **OUTPUT** (**green**-coded) button to switch between the digital and analog output modes.

Various state information is shown at the top of the home screen. The icons are from left to right:

(1)Input sample rate (in kHz)

(2)Input bit depth

(3)House curve indicator

(4)Tilt curve indicator

(5)Lowpass or highpass indicator

(6)Custom EQ indicator

(7)Infrasonic filter indicator

(8)Speaker configuration indicator. A yellow triangle in this position means that Anti-Mode room correction is NOT active.

➤Press the **BYPASS** (**orange**-coded) button on the remote to toggle the room correction on and off.

(9)Displays the currently active sound profile (designations "A" to "D"). Press the corresponding button on the remote to select a new active sound profile. See the "Using Sound Profiles" section for further information.

➤Press the **DISPLAY** (**blue**) button on the remote to toggle screen dimming. The dimmed mode is less obtrusive with much less information on display.

4.Settings Menu

Access the settings menu from the home screen by pressing the **MENU** button on the remote, then using the arrow buttons to highlight Settings and pressing **OK**. Among other things, this menu contains powerful tools for customizing the sound signature.

	The House curve tool. An adjustable target curve for the low frequencies (bass).
	The Tilt curve tool. An adjustable target curve for the higher frequencies (treble).
	Enable or disable the built-in infrasonic filter. Select between three most common cut-off frequencies (10 Hz , 15 Hz or 20 Hz).
	The Lowpass or highpass tool. Create lowpass or highpass filters with the desired characteristics. Up to two independent filter slots are available. Only recommended for advanced users.
	The Custom parametric equalizer tool. Create filters with freely adjustable center frequency, bandwidth and gain. Up to 16 independent filter slots are available. Only recommended for advanced users.
	The Digital headroom setting. Most tools will automatically increase headroom to avoid signal clipping. Manual adjustment is offered for advanced users.
	Audio Settings. Analog input settings can be configured from this menu.
	Change the menu language.
	Factory reset: clear all data from the selected sound profiles (the Anti-Mode correction, custom settings etc.) Use with caution: this operation cannot be undone.
	Display firmware version and CPU usage information.
	PC link: export the measurements to a computer via USB or perform system update. System update requires separate software which is currently available for Windows only. Visit www.dspeaker.com for more information.



PLEASE NOTE: Any changes made in the various EQ editors are generally not updated in real-time. In order to hear the difference in the sound output, accept the changes and close the corresponding editor by pressing **OK** on the remote controller.

4.1 House Curve Tool

The House curve tool provides a quick method of changing the target curve of the bass response. The motivation behind a house curve is in the sensitivity of the human auditory system and the hearing threshold - in addition to personal preference, of course. When sound pressure level is low enough, we are no longer able to hear the lowest bass frequencies.

It is easy to add a boosting equalizer to the lower bass region using the house curve tool.



WARNING: As with any low-frequency boosting EQ, employing an engaging house curve must be done with caution. Excessive low frequency lifting may introduce clipping of the amplifier or even cause damage to the woofer. Add house curve boosting in small increments at a time.

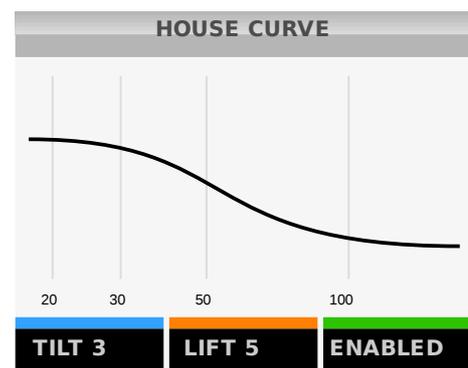
The parameters of the House Curve tool are:

✓**Tilt:** Defines the frequency range of the low frequency boost (the effect is illustrated on the screen). Adjust using the **LEFT/RIGHT** buttons or the **blue**-coded button.

✓**Lift:** Defines the maximum boost in decibels (the effect is illustrated on the screen). Adjust using the **UP/DOWN** buttons or the **orange**-coded button.

✓**Enabled:** Enables or disables the applied house curve. Toggle with the **green**-coded button.

Please note that modifying the parameter values will automatically enable this equalizer. Press the **green** button (followed by **OK**) to disable it again.





EXAMPLE: Boosting the lower bass under 50Hz by about 3 dB

How to do it step-by-step:

- Open main menu by pressing the **MENU** button on the remote controller.
- Navigate to the Settings menu using the arrow buttons and press **OK** to select it.
- Select the House Curve Tool using arrow buttons again (the first item on the upper row.)
- Press **OK** and the House Curve tool opens. Using the **LEFT** and **RIGHT** arrow buttons, set Tilt to 2 or 3. The approximated response is shown on the screen.
- Using **UP** and **DOWN** arrow buttons, set Lift to 3.
- Once ready press **OK** to apply the new house curve

4.2 Tilt Curve Tool

The Tilt curve tool provides a quick method of changing the overall frequency response slope of the treble. The motivation behind a tilt curve is (along personal preference) to compensate differences in various recordings. One can, for example, make a profile with treble tilted down to tame too “bright” recordings.

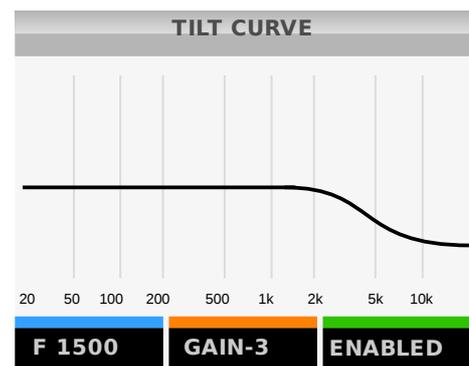
The parameters of the tilt curve are:

✓**Freq:** The transition frequency of the effect (illustrated on the screen). Adjust using **LEFT/RIGHT** arrow buttons or the **blue**-coded button.

✓**Gain:** Attenuation (negative values) or emphasis (positive values) of the selected band. Adjust using **UP/DOWN** arrow buttons or the **orange**-coded button.

✓**Enabled:** Enables or disables the applied tilt curve.

Please note that modifying the parameter values will automatically enable this effect. Press the **green** button (followed by **OK**) to disable it again.





EXAMPLE: Adding a -3dB treble tilt-down above 5 kHz

Background: Some recordings seem too bright or 'toppy' in their tonal balance. To compensate this the high frequencies can be toned down using a subtle high frequency attenuation filter.

How to do it step-by-step:

- Open main menu by pressing the **MENU** button.
- Navigate to the *Settings* menu using the arrow buttons and press **OK** to select it.
- Select the *Tilt Curve* using the arrow buttons again (the first element on the lower row.) Press **OK** to open the Tilt Curve tool.
- Now you can adjust the pivot frequency using **LEFT/RIGHT** arrow buttons, until the value is 5000Hz.
- Use the **UP/DOWN** buttons to set attenuation to -3dB. You can see the approximated frequency response of the filter on the screen.
- To save the filter, press **OK**.

At any time, you can disable the tilt curve by opening the Tilt Curve tool again and pressing the **green**-coded button to disable the effect, followed by **OK** to apply the changes.

4.3 Infrasonic Filter

The Infrasonic filter tool is a quick way to apply a subsonic highpass filter that filters out the signals below the audible band. The motivation behind an Infrasonic filter is to protect the woofer cones and to eliminate the effect of infrasonic harmonics and/or inter-modulation distortion. It is recommended to use an infrasonic filter whenever excessive house curve or low frequency boosting parametric equalizers are applied.

The controls of the infrasonic filter are:

✓**Frequency:** The frequencies below this threshold are attenuated.

✓**Enabled:** Enables or disables the infrasonic filter.

Use the directional buttons on the remote controller to change the parameters. Press the **OK** button to store settings or the **BACK** button to cancel.

Note: You can also design a custom infrasonic filter with the Lowpass/Highpass tool, which gives more control over the parameters (see High/Lowpass filters section below.)

4.4 High/Lowpass filters

The Anti-Mode 2.0 Dual Core features fully customizable highpass and lowpass filters. These filters can be used to control the pass-band of the speaker or subwoofer. For example, with a custom lowpass, it is easy to implement a more precise digital cross-over filter for a subwoofer. A custom highpass filter can be used to attenuate lower frequencies and thus prevent them from reaching the main speakers.

To design a custom lowpass or highpass filter, navigate into the *Highpass/lowpass* section of the *Settings* menu. Note that you can design two independent filters by selecting a different filter slot (use the **LEFT/RIGHT** arrow buttons to highlight a slot and press the **OK** button to accept).

This allows you to design extremely steep lowpass or highpass filters, for example duplicating a maximally steep filter results in 48 dB / octave slope.

The controls of the high/lowpass filter tool are:

✓**Frequency:** the cut-off frequency of the filter, at which point the signal is attenuated exactly 6 dB. Activate with the **blue**-coded button, adjust with the **LEFT/RIGHT** arrow buttons (coarse) and **UP/DOWN** (fine).

✓**Slope:** the steepness of the filter (6, 12, 18 or 24 dB / octave). Cycle with the **orange**-coded button.

✓**Settings:** open the settings submenu.

Settings submenu:

✓**Type:** selects the filter type: highpass or lowpass. Use the arrow buttons to toggle the selection.

✓**On/off:** enables or disables the filter. Use the arrow buttons to toggle value.

4.5 Custom Parametric Equalizers

The Custom Parametric Equalizers (PEQs) are very versatile filter structures that can implement a wide range of different bandstop, bandpass, highpass and lowpass filters.

All user-defined parametric EQs are organized in 16 filter slots. The first step of the Parametric EQ tool is the filter list screen. Use the arrow buttons to highlight an inactive filter to create a new custom filter, or select an existing filter to change its parameters, then press the **OK** button to accept the selection. Note that there are

more slots than the 6 visible ones, just scroll down with the **DOWN** button. Active filters are represented with green boxes in the filter list, inactive filters are grayed out.

The filter name (such as EQ01↓S) gives general information about the designed filter. The number after the EQ refers to the number of the filter slot (ranging from 01 to 16). The arrow points either up or down to indicate its type. A filter indicated by an up arrow has heavier impact on CPU usage. The last letter indicates whether the filter is stereo ("S"), left channel only ("L") or right channel only ("R").

After selecting a filter to edit, the filter overview screen appears. An approximated response of the filter is displayed. The following actions are available:

✓**Edit**: activates the edit mode (see below).

✓**Filter name**: the filter slot of the current PEQ.

✓**Settings**: Opens the settings submenu.

Edit mode controls are:

✓**Center Frequency**: the frequency at which the filtering effect is applied. Adjust with **LEFT/RIGHT** arrow buttons (coarse) and **UP/DOWN** buttons (fine)

✓**Bandwidth**: approximately the 3 dB bandwidth of the effective band. This is basically equal to setting the Q value of the filter, but for simplicity user can adjust the bandwidth instead of the Q-value. ($Q\text{-value} = F_c / B_w$). Adjust with **LEFT/RIGHT** arrow buttons (coarse) and **UP/DOWN** buttons (fine).

✓**Gain**: filter gain in decibels. The value is negative for a cutting PEQ (bandstop) and positive for a boosting (peaking) PEQ.

Press the **OK** button (or the **BACK** button) to return to filter overview.

Options in the settings submenu:

•**Channel**: Defines the channel this filter is associated with (left, right or both).

•**On/off**: Enables or disables the designed filter.

Press the **OK** button (or the **BACK** button) to return to filter overview.

Press the **OK** button to accept any changes, close overview and return to the filter list. Use the **BACK** button to discard any changes made. **Please note** that any new filters or edits will not affect the sound audibly until the filter list is closed.



EXAMPLE: Adding a psycho-acoustic 3 kHz dip

Background: The idea behind making a 3 kHz dip originates from the human auditory system and diffuse field characteristics. The human hearing is less sensitive to diffuse fields in the neighborhood of 3 kHz. Because of a flat microphone response, compared to the original listening situation for example in a concert hall, many recordings have too much energy around 3 kHz. A cutting PEQ at 3 kHz can be introduced to compensate. Whether or not this is beneficial is left to be judged by the user. This example shows how to create the cutting PEQ or any other parametric equalizer.

How to do it step-by-step:

- Open the menu display by pressing the **MENU** button
- Navigate to the “Settings” menu using the arrow buttons and press **OK** to select it
- In the settings menu, select the “Parametric EQ” using the arrow buttons. (The second element on the lower row.) Press **OK** to proceed. The Parametric EQ filter list appears.
- First you need to assign a filter slot for the filter. If there are no previously designed Parametric EQ filters, the first slot is automatically selected and you only need to press **OK** to proceed to the PEQ designer.
- At the filter overview, open the “EDIT” submenu by pressing the **blue**-coded button on the remote controller.
- Adjust the center frequency to 3000Hz by pressing the arrow buttons. (You can press and hold the buttons to adjust faster.)
- Press the **orange**-coded button to activate the bandwidth adjustment. Using the arrow keys set the bandwidth to about 1000 Hz.
- Press the **green**-coded button to adjust the filter gain. Using **UP/DOWN** for +/- 0.1dB adjustments and **LEFT/RIGHT** for +/- 0.5dB, set the gain to -4.2dB.
- The custom equalizer is now ready. Save it by pressing **OK** twice. You are returned to the filter list. The designed filter gets enabled when you go back to the settings menu by pressing the **BACK** button.
- Now you can try out how the psycho-acoustic dip works for you. You can also fine-tune the parameters and try to find out if there is a setting that suits your ears the best.

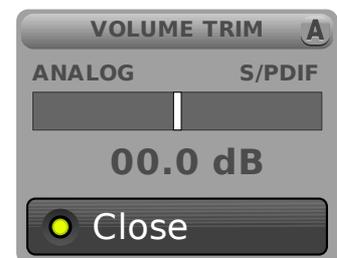
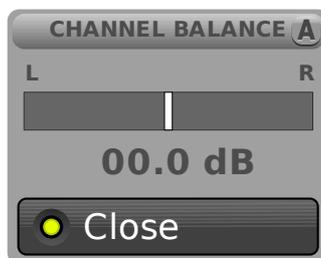
4.6 Digital Headroom (Advanced)

All internal digital processing is performed with 32- or 40-bit accuracy with the appropriate number of bits reserved for headroom. However, if emphasizing (raising) filters are used, some input signals could cause saturation / clipping at the output. The unit automatically calculates the best headroom setting from the gains of the active filters. However, if you only use the analog input and do not use the full input range, you can use a smaller headroom.

The headroom value is automatically re-calculated any time you change filter settings.

4.7 Audio Settings

Analog input settings can be configured from this menu.



Input sensitivity setting adjusts the maximum voltage level of the analog input. Default value is low, which ensures maximum compatibility with different audio sources. Use high sensitivity if you connect a device which follows the line-level standard to gain a bit higher volume level and better signal-to-noise ratio.

Use manual channel balance adjustment to match the levels of the left and right speakers. Changes in this dialog take effect immediately. Press the **OK** button on the remote to accept the new adjustment or press the **BACK** button to discard changes.

Volume trim menu allows compensation of different volume levels between the digital and analog inputs. For example, if the digital input is subjectively louder, adjust the volume trim value towards the left side until the volume levels are equal. Changes in this dialog take effect immediately. Press the **OK** button on the remote to accept the new adjustment or press the **BACK** button to discard changes.

Notice: channel balance setting is associated with the current sound profile. Input sensitivity and volume trim are global settings, i.e. the values are shared by all four sound profiles.

4.8 Language Selection

When you first turn on the unit, you are asked to select a language for the user interface. Using "Language Selection" you can later change the language. Currently (firmware "May 30 2012") the available languages are English, Finnish, Swedish, Dutch, and German.

4.9 Factory Reset

The factory reset menu lets you clear data from selected or all sound profiles.

Use with caution: the operation cannot be undone.

4.10 Information

Use this menu to display firmware version and CPU usage information.

4.11 PC Link

By selecting "PC Link", you can export the measurements of the currently selected profile to a PC or Macintosh computer via USB. Anti-Mode 2.0 Dual Core will appear as a removable disk and you can copy the measurements as text files that can be imported to popular PC / MAC measurement programs. The measurements are named accordingly to indicate the profile and measurement type: the four files in each profile are for example profile A as.

"a-after.txt" - The automatically generated response after the Anti-Mode room correction

"a-before.txt" - The automatically generated initial response of the room

"a-user1.txt" - The user made room measurement

"a-user2.txt" - The secondary user made room measurement

Equivalently for other profiles, the naming starts with the profile name, for example "b-after.txt" is the corrected response of the profile B (if it has been calibrated).

System update is also performed in "PC Link" mode. The system update requires separate computer software which is currently available for Windows only. Visit www.dspeaker.com for more information.

5.Using Sound Profiles

Anti-Mode 2.0 Dual Core supports up to four independent sound profiles. A sound profile includes everything that affects the sound: the room correction filters and all possible custom filters (highpass/lowpass filters, house curve etc.) Also all room measurements are stored in the profile.

Profiles can be utilized for various things. For example, you can calibrate up to four different listening spots using a profile for each location and switch between the spots quickly and conveniently. Or, you can create separate equalizer presets for music listening and movie watching etc. Sound profiles can also be of assistance when comparing the effects of different custom filters.

Initially, only one profile (A) is available. To create another, navigate to the Sound Profile dialog located in the Main Menu.

	Select this icon to create a new profile. The newly created profile will be identical to the currently active profile. The new profile will populate the first empty slot available. If all four profiles are in use, one of them must be deleted before a new profile can be created.
	Select this icon to delete the currently active profile. Please note that profile A cannot be deleted even if other profiles would still exist.

After the new profile has been created, you can change its characteristics by starting the room calibration process or modifying one or more settings in the Settings menu.

Note: if you wish to create an empty sound profile without inheriting any characteristics, you can use Factory Reset located in the Settings menu to clear the newly created profile from all previous data.

6. Advanced Calibration

Advanced calibration gives the user more control over the calibration parameters. The procedure is similar to the typical calibration, but with some additional steps. Refer to "Typical Calibration" section of this manual for an overview of the basic process.

To perform the advanced calibration, choose the Advanced Calibration option at the Calibration menu.

➤ After choosing the relevant speaker configuration, the extra settings dialog will appear.

Level compensation affects compensation for wide dips in the bass response. Compensation has three levels, "Off", "Norm" and "Max". If the setting is "Off", no compensation to boost dips is applied, and the bass level remains as low as it would be without room effect. This gives maximum headroom for the user to design custom boosting filters, such as House Curve or Parametric EQs. The "norm" setting applies a mild compensation, which is the same amount that "Typical" calibration is using. This setting usually gives the best audible result as the perceived level of the bass is closer to the original than with the uncompensated "Off" setting. In the "Max", maximum compensation is used, which gives about +2dB more gain for boosting filters. This setting is beneficial in spaces where part of the problem is the general lack of bass, possibly resulted by the absence of proper acoustic barriers. It is good to remember that destructive reflection nulls can not, will not, and should not be attempted to be corrected by any DSP.

➤ Select the desired amount of compensation and press the **OK** button to proceed.

➤ The next dialog controls the time-alignment of channels.

There are three possible options: Auto-detect, Off and Manual. It is advised to use the auto-detect for subwoofers and "Off" for the full-range speakers, as the full-range speakers are assumed to be equidistant from the listener. Note that optimal acoustic delay is not always the same as physical distance to the subwoofers: because of this the automatic detection is advised to be used always on subwoofers. If the manual setting is chosen, the compensation delay for each channel for subwoofers / speakers can be entered to the corresponding fields. The manual channel compensation delays are meant primarily for more complex systems, such as a 2.2 system with two Dual Cores or equivalent. In the Typical calibration, automatic detection of delay is always employed when

calibrating two subwoofers (either stereo or dual-mono) but not in full-range 2.0 mode. You can still use time-alignment and auto-detect for the full-range 2.0 setup by choosing the Auto-detect option. The auto-detection always calculates compensation to produce the minimum lag, meaning it will only delay the channel that is “too early” compared to the other, whereas manual adjustment of the delays allows delaying both channels. This is of course unnecessary for any systems where all of the audio is going through Anti-Mode 2.0 Dual Core.

➤ Press the **OK** button to proceed to the next dialog.

Now you can select the upper frequency limit of the automatic correction. Using the arrow buttons, set the upper limit of the correction between 80 and 500Hz. Bass signals have no directionality at the lowest frequencies, but they get directional toward the mid-range bass and above. Because of this, Anti-Mode 2.0 Dual Core employs correction of the summed output on lowest frequencies (where all audio material is essentially monophonic), but the correction becomes more channel-dependent towards lower mid-range and higher frequencies. After all it is a different concept to measure either left or right channels separately, compared to measuring both in-phase (which applies to the majority of any bass content). All this directionality versus summed (in-phase) effects are automatically handled by Anti-Mode 2.0 Dual Core controlled by the selected correction frequency range.

➤ After selecting the desired settings, press **OK** button on the remote to proceed to the volume level check. From here on, the calibration process is similar to the Typical calibration described earlier.

7. Multi-point (Wide Area) Calibration

Anti-Mode 2.0 Dual Core features a wide area optimization method which allows for a larger “sweet spot” to be corrected. Fortunately, wide area algorithm does not require you to measure dozens of different places around the listening area. In most cases it should be enough to make the main calibration in the center of the target listening area, and when it is complete, more points can be added to the calibration data.

➤ To perform a multi-point calibration, the first step is to calibrate the device normally, following either the Typical or Advanced procedure.

➤ After the first calibration, you can add more points by choosing the “Multi-point” option at the Calibration dialog. Place the microphone to one edge of the listening area and continue calibration.

➤You can add as many points as you want to the calibration, and the system automatically weights them by comparing them to the main calibration data. The maximum recommended number of additional points is 7, more points would not contribute much to the correction model.

Please note: if the listening area consists of rather two alternate areas, instead of one larger one, it is recommended to perform separate calibrations into two separate profiles. In other words, the alternative area is calibrated into another profile, and the user can then switch the optimum correction to one of the spots at a time.

8. Measurements

The Measurement menu allows you to measure and view the response of currently active filters. You can also view the room responses measured during the automatic Anti-Mode calibration process. The calibration measurements always show the most important frequency range in terms of room correction – from 16 Hz to 200 Hz.

You can create and view two custom measurements. The measurement frequency range can be the same as for the calibration responses, but sometimes it can be useful to measure the whole audio band. For example, it may be helpful when trying to find the optimal places for acoustical treatments or for example the toe angle for the loudspeakers.

Note that “Measurements” is not the same as “Calibration”: with the Measurements tool you simply make measurements without applying any correction automatically.



PLEASE NOTE: All room response measurements are always shown just as they were measured, without any “sugar coating”. No smoothing is applied to make things look prettier than they actually are.

To make a custom measurement, navigate to the Measurements in the Main menu.

➤Highlight “Make New” and press the **OK** button.

➤In the next menu, you can define the range of the measurement: bass (16Hz - 200Hz) or full-range (20Hz - 20000Hz). You can also select the target channel (left, right or both) to be measured.

➤Remember to check that the microphone is connected. Once you are ready, you can start the measurement.

With full-range measurements, the alignment of the microphone is more sensitive than with the bass-only measurements. The microphone should be pointing towards the ceiling.

➤Once the first measurement is ready, a secondary measurement slot becomes available in the Measurements menu. This measurement has the same range as the first one, but you can select the channel(s) to measure. This enables you to, for example, compare a room response of the left channel to that of the right channel, compare the effects of the designed custom filters, different points within the listening zone and so on.

➤NEW (Jun 07 2012 >) By pressing left/right arrow button you can activate tracker cursor to view the exact frequency and magnitude value. Pressing up/down will move the cursor with smaller step

➤Close the measurement dialog with the **OK** button to store the measurement.

➤Remember that each sound profile has its own measurement data. This way, with up to four profiles available, you can store a maximum of 8 custom measurements. This is in addition to the 4 "before / after" responses of the Anti-Mode calibration measurements that are automatically generated.

9.Audio Connections

Because of the versatile connections and the wide range of possible uses for the Anti-Mode 2.0 Dual Core, it is hard to list all combinations. Each case is at least subtly different. Examples of the basic cases will be added to the next version of the manual.

Consider the following guidelines when connecting Anti-Mode 2.0 Dual Core.

✓The Anti-Mode 2.0 Dual Core as a 2-channel pre-amplifier and DAC

Connect your audio sources to the Anti-Mode 2.0 Dual Core's inputs. You can connect up to three sources using USB, Toslink (S/PDIF) and one of the analog inputs.

Connect the Dual Core to your amplifier using the XLR or RCA outputs.

✓Using with an Audio / Video Receiver or a Home Theater Amplifier

Use a DSP processing loop or tape monitor loop to connect the Dual Core.

If this is not possible, pick the audio source that most benefits from the room correction and insert the Dual Core between that source and the AVR.

✓Using with a 2.1 or 2.2 System

If you have a 2.1 or 2.2 system with an existing cross-over, connect the Dual Core before the cross-over. Your system is then seen as a 2.0 full-range system by the Dual Core and both your main speakers and subwoofer(s) get corrected.

✓Correcting Different Subwoofer Configurations

Select the appropriate configuration for your subwoofer set-up during the calibration process. If you connect more than two subwoofers, group them according to location and how symmetric their placement is. For example subwoofers in the front corners are more similar to each other than a subwoofer in a corner is to another subwoofer in the middle of a wall.

10.Troubleshooting

xAnti-Mode does not remember my settings after power down: to ensure optimal use of the built-in non-volatile memory, common settings are stored only when entering stand-by mode. In other words, if you want to store settings (such as default volume level, input volume trim etc), simply press the red power-on/off button on the remote controller. After that, Anti-Mode 2.0 will remember the stored settings even after power down.

xAmbiguous delay: this error may occur at the beginning of the calibration process if the distance / delay is not detected correctly. In most cases it happens either because of a bad placement of the microphone, or simply because of too low calibration volume compared to the ambient noise. The remedy is to move the microphone about 4 inches (10cm) away from the original place and making sure that there are no objects in the proximity of the microphone capsule. Also increasing the measurement volume is recommended unless the volume is already well beyond the "Volume OK" threshold. The distance / delay compensation values can also be inserted manually. The channel that is closer, should be delayed by $1000 * S / V$ [ms] where S is the difference in distances and V is the velocity of sound (340m/s or 1116 feet/s). For example, if the left speaker is 3 ft. closer than the right one, the compensation delay should be set to the left channels as $1000 * 3 / 1116 = 2.7\text{ms}$ and 0ms to the right channel.

x"Clipping Detected": this warning is displayed at the home screen in two cases.

1."Clipping Detected" is shown when Anti-Mode 2.0 Dual Core detects that the analog input signal has reached the maximum input voltage level, causing the input to be clipped.

Check the output settings of the device you have connected to the Anti-Mode 2.0 Dual Core analog inputs. If you can not reduce its output level, check that the input sensitivity is set to low in the "Audio Settings" menu. If this does not help, you can try using the RCA inputs instead of XLR inputs.

2. "Clipping Detected" is also shown when there is a possibility of clipping at the output stage, which could compromise audio quality.

Usually the reason for this warning is that too many boosting equalizers are being employed. To overcome, increase the digital headroom value (located at the "Settings" menu). Sometimes also decreasing the input gain may help.

xNo sound: check the connections. Check that you have selected the correct input (ANALOG, USB, or S/PDIF). Use the **INPUT** button on the remote controller to select the correct input. Check that mute is not active and that the volume level is not too low. Check that you have the correct output. Use the **OUTPUT** button on the remote controller to switch between the analog (RCA and XLR) and digital (S/PDIF Toslink) outputs. Note that you can use the room measurement's level check tool to generate noise and confirm that the output is functioning.

xNo sound / bad sound from S/PDIF digital input: In addition to the above, check the digital signal format of the audio source. Please note that currently Anti-Mode 2.0 does not accept 192 kHz sampling rate. Check that you have the latest firmware.

xNot enough bass after calibration: a rather common first impression after listening to years of resonance-infested bass. If the calibration system consisted of subwoofers, their volume can be adjusted after the calibration. With loudspeakers (as well as on subwoofers) you can use the House Curve or Parametric filter tool to increase the bass to match the preferred level and tone. Also you can try calibrating the advanced calibration using "Max" compensation setting.

xCustom filters get disabled: this can happen for two main reasons. First of all, when you start the calibration, all filters are disabled automatically. Even if you cancel the calibration before a new room correction is applied, the disabled filters are not activated with the exception of infrasonic and high/lowpass filters. The filter designs do not vanish, however. You can simply re-enable them after the calibration. Another possible reason is running out of computational resources. Even though Anti-Mode 2.0 is equipped with two very powerful processors, even their capacity is not infinite. If too many complex filter structures would exceed the CPU limit, the system automatically disables one or more of the least significant filters to not exceed the available resources.

xProblems with room correction are usually ultimately caused by a bad room calibration analysis. The single most important factor for successful room analysis is to make sure that the calibration volume is sufficiently high. Sometimes the automatic level detection may indicate "Volume OK" even when it is in fact too low. This can occur especially in noisy environments and/or with limited band subwoofers. When calibrating a subwoofer system, always disable subwoofer's integrated low-pass filters or set them to their maximum cut-off frequency. You can enable them after the calibration process, or utilize Dual Core's digital precision low-pass filters that are probably more adjustable and capable. It is also important to try to avoid all other cables running along the microphone cable, as they might induce errors through the microphone shielding. Make sure you do not open or close doors during calibration, as they tend to have much greater impact to the room response than expected. After these checks, recalibrate and verify that the before / after graphs are looking good. If the problems persists, contact DSPeaker support. It speeds up the troubleshooting process if you have a picture taken of the before / after graphs (with camera etc.) or exported the responses to a file using USB.

xVolume jumps each time USB is selected, because your PC sets the last volume setting it has previously used with the device. The PC is not aware of any manual adjustment performed by the remote control. Use the master volume control on the PC to set suitable volume. That setting is restored each time USB input mode is selected while the unit is attached. Be aware that volume controls in an audio or multimedia player program may use digital volume control, which causes the digital data to be scaled. So check that you use master volume. Using the master volume control does not cause any precision to be lost, because it controls the analog volume at the output.

11. Technical Specifications

11.1 Interfaces

- 2 x RCA inputs or alternatively 2 x XLR inputs
- 2 x RCA outputs and 2 x XLR outputs
- Toslink S/PDIF digital input (2-channel PCM only)
- Toslink S/PDIF digital output
- USB Audio (USB used also for measurement export and software update)
- DSPeaker Datalink
- Microphone jack
- 12VDC / 9VAC power connector

11.2 Analog Specifications

- Dynamic Range: > 108dB
- Total Harmonic Distortion (analog in, analog out, typical): 0.003%
- Volume Control Steps: 0.5dB
- Input Sensitivity XLR: 1.3 / 2.6 Vrms, RCA: 1.6 / 3.25 Vrms
- Output Voltage Unbalanced RCA: 1.65 V(RMS)
- Output Voltage Balanced XLR: 3.25 V(RMS)
- Power consumption (typical): active 2.6W, stand-by 0.6W

11.3 Firmware Specifications

Firmware specifications are subject to change between firmware versions.

- Digital Signal Processing: 40 bits, 2 channels
- Anti-Mode 2.0 Multi-Rate (FIR & IIR)
- House Curve, Linear-Phase Tilt, Parametric EQs, Adjustable Infrasonic Filter
- USB and Toslink S/PDIF inputs 44.1 kHz / 48 kHz 24-bit
- Toslink S/PDIF output 48 kHz 24-bit

12. Contact

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