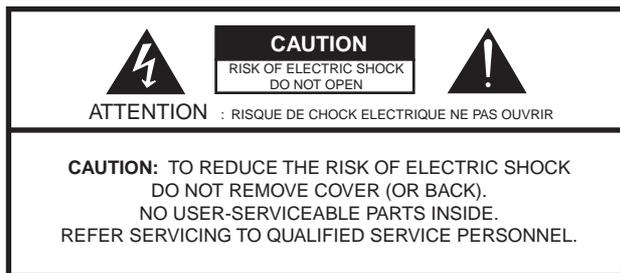


ADDA 2408

User manual



Digital Audio Denmark



The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.



The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.

IMPORTANT SAFETY AND INSTALLATION INSTRUCTION SAVE THESE INSTRUCTIONS

INSTRUCTIONS PERTAINING TO RISK OF FIRE, ELECTRIC SHOCK, OR INJURY TO PERSONS

WARNING – when using electric products, basic precautions should be followed, including the following:

1. Read all of the safety and installations instructions and explanation of graphic symbols before using the product.
2. The product must be grounded. If it should malfunction or breakdown, grounding provides a path of least resistance or electric current to reduce the risk of electric chock. This product is equipped with a power supply cord having an equipment-grounding conductor and a grounding plug. The plug must be plugged into an appropriate outlet which is properly installed and grounded in accordance with all local codes and ordinances.

DANGER – Improper connection of the equipment-grounding can result in a risk of electric shock. Do not modify the plug provided with the product – if it will not fit the outlet have a proper outlet installed by a qualified electrician. Do not use an adapter which defeats the function of the equipment-grounding conductor. If you are in doubt as to whether the product is properly grounded, check with a qualified serviceman or electrician.
3. Do not use this product near water – for example, near a bathtub, washbowl, kitchen sink, in a wet basement, or near a swimming pool, or the like.
4. This product should only be used with a stand or cart that is recommended by the manufacture.
5. This product, either alone or in combination with an amplifier and speakers or headphones, may be capable of producing sound levels that could cause permanent hearing loss. Do not operate at a high volume level or at a level that is uncomfortable. If you experience any hearing loss or ringing in the ears, you should consult an audiologist.
6. The product should be located so that its location or position does not interfere with its proper ventilation.
7. The product should be located away from heat sources such as radiators, heat registers, or other products that produce heat.
8. The product should be connected to a power supply only of the type described in the operating instructions or as marked on the product.
9. The power-supply cord of the product should be unplugged from the outlet when left unused for a long period of time. When unplugging the power supply, do not pull on the cord, but grasp it by the plug.
10. Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.
11. The product should be serviced by qualified service personnel when:
 - A. The power supply cord or plug has been damaged, or
 - B. Objects have fallen, or liquid has spilled into the product, or
 - C. The product has been exposed to rain, or
 - D. The product does not appear to be operating normally or exhibits a marked change in preformance, or
 - E. The product has been dropped, or the enclosure damaged.
12. Do not attempt to service the product beyond that described in the user maintenance instructions. All other servicing should be referred to qualified service personnel.
13. **WARNING** - Do not place objects on the power supply cord, or place the product in a position where anyone could trip over, walk on, or roll anything over cords of any type. Do not allow the product to rest on or be installed over cords of any type. Improper installations of this type create the possibility of a fire hazard and/or personal injury.

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1. Introduction

Welcome

Congratulations, and thank you for purchasing the Digital Audio Denmark ADDA2408 audio converter. It is an extremely powerful full duplex 8-channel audio converter for independent simultaneous analog-to-digital (A/D), and digital-to-analog (D/A) conversion or digital-to-digital (D/D) sample rate and format conversion.

The ADDA 2408 has modular structure enabling the use of up to 8 I/O modules.

Note: This user manual is a basic manual. For a more detailed description, please look at our web site at: <http://www.digitalaudio.dk>

Overview of ADDA 2408

The ADDA features everything you have ever wanted from an AD/DA converter, and more!

Key Features:

- 8-channel 24 bit, 96 kHz A/D converter and D/D sample-rate and format converter.
- A/D sampling with anti-AID filter for elimination of Alias Intermodulation Distortion.
- Low noise microphone preamplifier with 0.5 dB gain control from -18 to +72 dB, phantom power +48V, and low-cut filter.
- Line input level adjustable from +12 dBu to +30 dBu.
- Digital interfaces compatible with AES/EBU, S/PDIF, TDIF, ADAT® and ProTools™.
- Triple digital bus structure, DADBus.
- More digital interfaces to come (compatible with the ADDA 2408 DADBus).
- A/D and D/D with ± 180 degree phase inversion, and variable delay setting from 0 to 100ms (250 ms optionally).
- Psycho acoustic dither selectable for A/D and D/D inputs to 16, 18, and 20 bit.
- Built-in 8-channel digital mixer for monitoring of the A/D, D/D, and D/A input.
- Monitoring in L/R or M/S stereo, optional D/A outputs for 5.1 monitoring.
- Solo monitoring for all channels.
- 21 LED digital peak meters on all 8 channels for monitoring A/D, D/D, or D/A inputs.
- Sample-rates of 44.1kHz, 48kHz, 88.2kHz and 96 kHz. High-rates can be double-wire or double-speed.
- 8 slots for expansion cards.
- External PLL based synchronisation to word clock, super clock or Video PAL, NTSC, SECAM.
- 5 ppm internal oscillators, for precise sampling and master clock application.
- Alarm generation for detection of unlocked digital interfaces.

Key Specifications:

- AD converter dynamic range 117 dB(A)
- DA converter dynamic range 117 dB(A)
- Mic pre. Equivalent Noise(A), +24 dB analog gain <-144 dB(A)
- Frequency response 20-20.000 Hz, ± 0.1 dB @48 kHz
- Frequency response 20-38.000 Hz, ± 0.1 dB @96 kHz

The ADDA 2408 features very high-quality 8-channel, 24 bit, 96 kHz A/D and D/D converter with microphone preamplifier, and an 2-channel D/A converter with a monitor mixer in one compact 2U box. 8 interface slots are available for additional D/A converters and digital interfaces. The digital interfaces can be either single-wire or double-wire depending on the options available for the digital format.

It is easy and fast in use. The most common functions are accessed by pressing only one button.

Shortcuts are available for switching between AD-DD source and Monitor, Solo, Pan, Delay, Low-cut and Phase inversion.

The shortcuts are valid while the button is pressed. The control displays will reflect the previous settings when the button is released, the control displays will return to the previous menu.

The converter is able to do format and sample-rate conversion as well. If a digital input signal is selected at the AD-D source button, the signal is available on all digital outputs (expansion cards) transformed with the dither and sample-rate chosen at the Dither and Sample rate or Sync button.

Supreme microphone preamplifier with low-cut filter and phase inverse selection. The preamplifier has an equivalent input noise of 144 dB at 24 dB analog gain. The level control can set the gain in 0,5 dB steps from -18 to + 72 dB for precise control of the microphone input level.

The preamplifier provides a unique combination of analog and digital gain, thus providing a 108 dB gain range. The analog gain is set in 3 dB steps of (the resolution of the analog input amplifier) in order to preserve the input noise performance. In between the 3 dB steps, it is possible to adjust the input level digitally in 0,5 dB steps. This enables the user to scale the microphone level optimally to the 117dB dynamic range provided by the A/D converter.

The AD converter has better than ± 0.01 dB linearity over a dynamic range of more than 90 dB, and the analog preamplifier implements a analog gain control of additionally 42 dB. This gives the microphone preamplifier very superior harmonic and intermodulation distortion performance at all levels.

You have the option of adjustable delay on all AD-DD source inputs. The delay can be set to between 0 and 100.0 ms (250.0 ms optional). The delay can be set in meters, feet or ms. This function allows you to correct the phase for the relevant channels in a multi microphone set-up. It also enables the user to correct the audio delay time in a video editing system.

The Monitor mixer features Solo, Pan, and MS stereo control. The monitor mixer is able to mix all 8 input channels into a 2-channel output. The ADDA 2408 can therefore be used as an 8-channel sub mixer as well. The Stereo output is available on the headphones output on the front and on line level XLR connectors on the back or, if chosen, on a separate digital AES/EBU output.

The high-quality headphone amplifier with the ¼ inch jack connection on the front is very powerful, and is able to feed 8 sets of 600 Ω headphones at a reasonably audio level. The headphone amplifier can cause serious ear damage if not used properly. Always keep the amplifier at a safe audio level.

You can synchronise the ADDA 2408 to AES 11, Word Clock, Super clock, the digital input, or to video black-burst signal (PAL, NTSC, SECAM). The built-in PLL circuit is able to correct for minor variations (jitter) in the external sync clock signal, giving a very stable adaptation to the external synchronisation signal.

The ADDA 2408 also features a synchronisation alarm. If an 8-channel digital output is selected with the DA source button, and the digital input on this interface for some reason is out of synchronisation, the red Sync alarm LED will blink. This function is available to secure that the whole recording system is synchronised. The function can be turned off in the system menu.

The internal oscillators are very precise and have a stability of 5 ppm. Therefore, the ADDA 2408 can be used as a master studio clock generator for most applications.

Newly developed ultra low noise noise-shaped dithering is available for 16, 18 or 20 bit recording.

Security locks on the power button. The power standby circuit can only be turned off by pressing the power button on the front for 3 seconds.

4 LCD displays show the status of all stereo channels or the selected parameters.

Detailed 21 LED peak metering with the possibility of storing the maximum peak level can be set to show either the digital or the analog input level.

It is possible to store and name different set-ups.

2. Installing the ADDA 2408

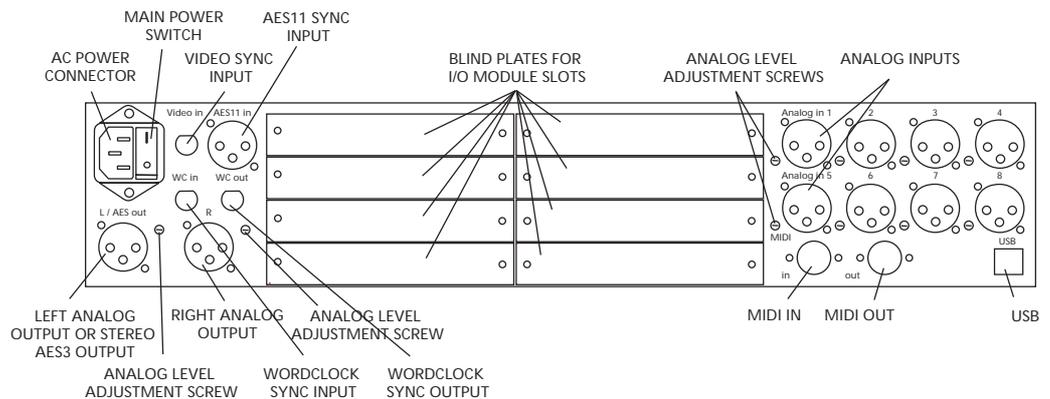
This is a quick description of the basics you need to cover in order to get started with your ADDA 2408.

Before You start

- Place your ADDA 2408 on a hard and dry surface or mount it onto a 19" rack, and leave plenty of room for ventilation.
- In order to meet the EMC requirements of directives 89/336/EEC and 93/68/EEC, and in order to obtain the high performance possible for the ADDA 2408, you must use correctly shielded cables of good quality for all external connections when installing the ADDA 2408. For the power connection, a normal unshielded power cable with a proper ground can be used.
- Make sure that your sound system is at a safe volume level.

Installation

This section will take you through installation of your ADDA 2408. We will describe how to mount the I/O modules and the power, audio and digital cable connections that can be accessed on the rear panel.



Installing I/O cards

If you need to mount an I/O module at a later stage, the following procedure is used.

Place the ADDA 2408 on a dry steady horizontal surface. Remove all cables (including the power cable). On the back of the ADDA 2408 there are 8 slots for I/O card mounting. On the drawing above, they are marked as Blind Plates for I/O module slots.

There's 2 different sizes of I/O modules: 1-slot modules and 2-slot modules. Decide where to place the module (When mounting a 2-slot I/O module make sure to have 2 free slots, one above the other, on the backplate of the ADDA 2408)

To remove blind plates for I/O module slots, remove the 2 screws on each side of the plate. Use a Posidrive screwdriver.

Only remove the number of blind plates necessary to fit the I/O module. If only one I/O slot is installed, remove only 1 blind plate. If 2 I/O slots are installed, remove 2 blind plates.

It is very important to insert the I/O card horizontally and carefully into the ADDA 2408. There are 3 guides inside the ADDA 2408 to guide the I/O card correctly into place. Do not in any way use force to insert the I/O card. This may damage the card.

When the cover plate of the I/O card covers the hole created by removing the blind plate, the 2 (1 slot) or 4 (2 slots) screws from the blind plate are mounted in the sides of the I/O card. Tighten the screws carefully and be careful not to damage the threads.

After inserting the I/O module, the ADDA 2408 might need to be initialised. If this is the case, please follow the instructions received with the I/O module.

The I/O card is now properly installed

Connecting the Power Cable

The ADDA 2408 runs on 85-240 V, 50-60 Hz AC voltage. Excessive voltages can seriously damage the ADDA 2408, so make sure that your AC power matches the voltage of your ADDA 2408.

When you connect the power use the cable you received together with your ADDA 2408 and plug it into a grounded outlet.

For safety and EMC reasons, and to prevent audio hum, the system must be properly grounded. If your power source does not have a standard three-hole socket, the system must be grounded in another appropriate manner.

Connecting the analog audio cables

Turn down the sound system volume, before rigging the ADDA 2408's analog audio cables. On the right side of the rear panel, you will find 8 balanced XLR input connectors marked 1 to 8. These analog inputs correspond to the ADDA 2408's 8 channels, and can be used to connect a balanced line input signal (+12 to +30 dBu) or a balanced microphone input signal. On the front panel, it is possible to set the type of input (line or microphone) for 2 channels at a time.

If line input is selected, the input level can be adjusted on the small adjustment screw just to the left of the XLR connector

On the left side of the rear panel, you will find two balanced XLR output connectors marked L/AES out and R.

In the System menu on the front panel, you can choose between analog L/R or digital AES output on these two outputs. If you have chosen analog output, you can connect two balanced XLR connector to Left and Right.

These analog outputs can be adjusted to a level of 0dBfs at +12 to +27 dBu on the small adjustment screw just to the left of the XLR connector.

If any of your I/O cards are DA modules, you should also connect the balanced XLR or Jack connectors at this point.

Connecting the digital audio cables

On the left side of the rear panel, you will find two balanced XLR output connectors marked L/AES out and R.

In the Monitor menu on the front panel, you can choose between analog L/R or digital AES output on these two outputs. If you have chosen digital output on the front panel, you can connect a XLR digital connector to L/AES out.

These AES signals carries the monitor stereo mix of the ADDA 2408. Many ADDA 2408 units can be linked together by connecting the AES3 output from one ADDA 2408 to the AES11 input on another ADDA 2408.

If one ore more I/O card is a digital audio interface, you should also connect the relevant cables to the connector.

Connecting the digital sync input/output

On the left side of the rear panel, you will find the Word Clock external synchronisation input and output connectors marked WC in and WC out, the AES 11 input connector marked AES11 in and the Video Black Burst input connector marked Video in.

If you are using a Word Clock studio master clock signal, you can connect this signal to the WC in connector. The Word Clock signal can be 44.1, 48, 88,2 or 96 kHz +/- 10%. If you wish to synchronise your ADDA 2408 to a WordClock signal, connect this signal to WC in through a BNC plug.

If you are using a Word Clock studio master clock signal, you can connect this signal to the WC in connector. The Word Clock signal can be 44.1, 48, 88,2 or 96 kHz +/- 10%. Word Clock output alternatively serves as SuperClock output. On the front plate in the System menu select which output to use the connection. If you wish to synchronise your ADDA 2408 to a WordClock or SuperClock signal, connect this signal to WC in through a BNC plug.

To use the ADDA 2408 as master studio clock for other units connect these using a BNC plug to WC out. The ADDA 2408 will generate a synchronising signal with the same frequency as used for the A/D or D/D conversion with a precision of 5 ppm.

There will always be a signal on WC out. Either a WordClock or a SuperClock signal. This means that the ADDA 2408 can convert a Video Black Burst or an AES11 signal to a WordClock or a SuperClock signal.

Wanting to use AES11 as external source for synchronising for the ADDA 2408 connect an AES11 signal to the XLR connection marked AES11 in. AES11 in can also be used to link multiple ADDA 2408 converters because an AES3 and an AES11 can be connected. Apart from synchronisation information, an AES3 signal can contain digital audio that is mixed with the stereo output from the monitor mixer of the ADDA 2408. To synchronise the ADDA 2408 to a Video Black Burst signal this is connected to a Video in with a BNC plug.

Connecting the USB 1,2 in/out and the MIDI in/out

The USB and MIDI are exclusively used to remote control the ADDA 2408. The USB in/out and MIDI in/out are found in the bottom right side corner of the the back plate of the ADDA 2408.

To remote control the ADDA 2408 from a PC or Mac via MIDI, connect the MIDI out from the computer with the MIDI in on the ADDA 2408 and the MIDI in on the computer and MIDI out on the ADDA 2408.

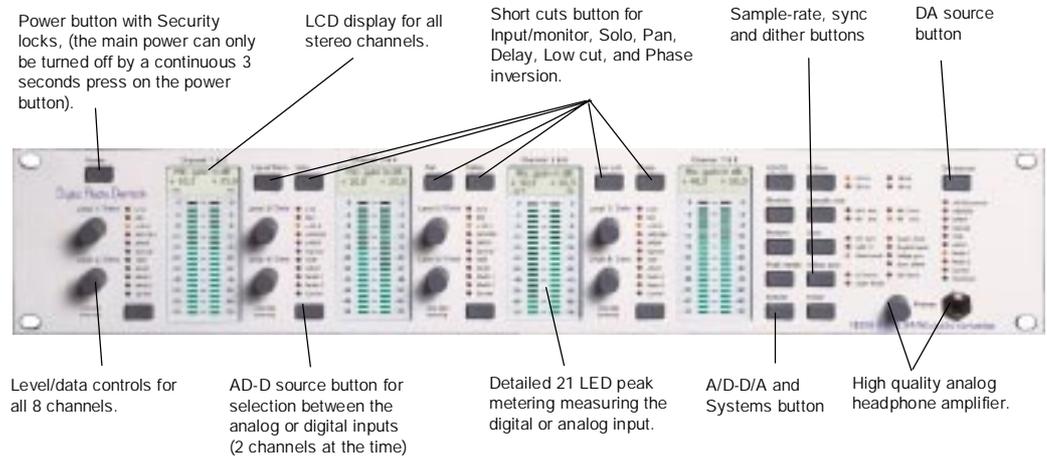
Multiple ADDA 2408's can be remote controlled by connecting the MIDI out from the computer with the MIDI in of the first ADDA 2408. From that same ADDA 2408 connect to the next ADDA 2408 from MIDI out to MIDI in and so on. MIDI out on the last ADDA 2408 is connected to MIDI in on the computer. In the System menu under Unit ID, separate Unit ID's are given to each ADDA 2408. It is important for the converters to have different ID's.

To remote control the ADDA 2408 from a PC via USB, connect the USB on the computer with the USB on the ADDA 2408.

Multiple ADDA 2408's can be remote controlled via USB by connecting the USB on the computer to a USB hub. The various USB outputs from the USB hub are connected to the USB on each ADDA 2408.

3. Operation

When you have connected the power and the audio cables, you have to activate the main switch on the rear panel and/or the power button on the front panel in order to turn on the ADDA 2408.



After start-up the ADDA 2408 has to calibrate for about 20 seconds. It is recommended to leave the ADDA 2408 on for approximately 5 minutes in order to warm up, and then activate the sample-rate button, (the ADDA 2408 has to be in internal sync. mode). This will cause the converter to recalibrate, and improve the dynamic range in a region of 1 to 2 dB.

The ADDA 2408 has a non-volatile memory system enabling the converter to remember the active settings. After a power up, the ADDA 2408 will restore the settings from before it was turned off.

The Power button

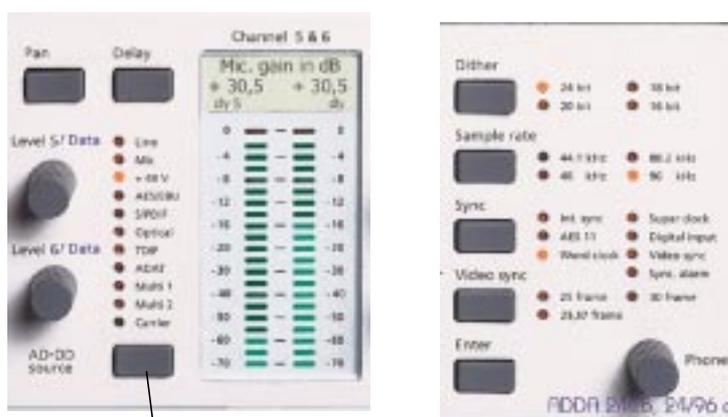
On the Power button on the left side of the front panel the converter can be set to a stand-by mode, where the power to the electronic circuits are disabled. For security reasons the Power button has to be activated for aprox 3 seconds in order to turn off the power.

By pressing the Power button in stand-by mode, the converter re-enters the operation mode, with all power restored, (if the main switch on the rear panel is switched off there will be no power on the ADDA 2408).

The AD-DD source button

The AD-DD button is used to decide which input to be A/D or D/D converted. Choose between Line, Mic, + 48V (Mic. with phantom power), AES/EBU, S/PDIF, Optical, TDIF, ADAT, Multi 1 or Multi 2. To Select a digital format the belonging I/O card must be mounted in the ADDA 2408. The digital or analogue input is chosen in pairs.

A constellation as an example could be to use channel 1 & 2 for conversion of a Phantom power microphone signal, AD conversion of a line signal on channel 3 & 4, format- and sample rate conversion of an ADAT signal on channel 5 & 6 and at last sample rate and format conversion of an AES/EBU signal on channel 7 & 8. This all at the same time.



AD-DD source button

Which type of signal selected for the 2 channels can be observed on the line of light diodes on top of the AD-DD button. If a digital interface is selected and the signal is on, the green light diode market Carrier lights.

The signals chosen on the AD/DD source button will be accessible in digital form on all digital outputs simultaneously with the sample rate and dithering chosen on the Sample-rate/Sync and Dither button. If several digital I/O's installed digital conversion between the formats can be accomplished.

Note that a digital multi channel I/O module uses the same sample frequency on digital in and digital out. If deciding to sample rate convert a signal from an 8-channel I/O no digital output on this module will be generated on the on the ADDA 2408.

Digital I/O's outside AES/EBU, S/PDIF, T/DIF or ADAT, for example ProTools will be installed as Multi 1 or Multi 2. The ADDA 2408 decides how to install a new digital I/O. Having 2 AES/EBU, S/PDIF, T/DIF or ADAT I/O's installed one will be installed as AES/EBU, S/PDIF, T/DIF or ADAT while the other as Multi 1 or Multi 2. (Watch section 2)

The Dither button

On the Dither button you can select the amount of dither added to the audio signal when sampled by the A/D or the D/D converter. The dither setting has to correspond to the bit resolution of the digital system to which the digital output of the converter is connected. If the audio signal is sampled in 24 bit resolution and connected to a 16 bit hard disk recording system, the noise behaviour of the recorded signal will not be optimal, and distortion in the sound will occur. Therefore it is important to set a correct bit (Dither) resolution.

The Sample Rate button

On the Sample rate button you can select the sample-rate of the A/D or D/D converter between 44.1 kHz, 48 kHz, 88.2 kHz and 96 kHz. When a sample-rate has been selected, the A/D converter will take about 3 seconds to calibrate before it is ready for operation. The sample-rate LEDs will always show the selected sample-rate. When an external synchronisation is selected via the Sync button, the sample-rate LEDs indicate the sample-rate of the received external synchronisation signal.

By pressing the Sample rate button in ext. sync mode at 44,1 or 48 kHz, a Word Clock, Super Clock or Video Black Burst signal can also be used to generate the double sample-rate i.e. 88,2 or 96 kHz. Pressing the Sample rate button will toggle the sample-rate between the rate of the synchronisation signal and the double rate. The selected sample rate is indicated by the sample-rate LED's.

The Sync button

On the Sync button you can select the synchronisation source for the sample-rate used by the A/D and D/D converter. The source can be internal synchronisation, which is the internal 5ppm precision oscillator, or external synchronisation, which can be AES 11, Word Clock, Super Clock (in case a multi channel ProTools compatible module is mounted), Digital input, or Video Black Burst reference signal connected to the AES11, WC in, Digital input, or Video in connector on the rear panel. Use the Sync button to choose the synchronisation source. The LED for the chosen synchronisation possibilities will light up.

At WC in and WC out, it is possible to connect a Word Clock signal. If a Word Clock signal is connected to WC in, you need to select Word clock with the Sync button. If a Super Clock signal is connected to WC in, you need to select Super clock with the Sync button.

If the Digital input is selected with the Sync button, it will be the Digital input chosen with the DA source button, which will be the synchronisation signal, used for the A/D or D/D converting.

If Video sync is selected with the Sync button it is necessary to select the frame rate of the video signal with the Video sync button. Choose between 25 frame, 29.97 frame and 30 frame.

If no ext. sync signal or digital input signal is connected to the ADDA 2408, the source can not be selected via the Sync button. If a selected external synchronisation source fails, the ADDA 2408 will automatically default to Int. sync mode. If the signal is re-established the ADDA 2408 will automatically switch to the selected source unless the synchronisation mode has been changed.

It is possible to detect the sample-rate of an incoming digital signal. Set the digital Sync button in Digital input mode and use the DA source button to select the digital input source you want to detect. The sample-rate LEDs will now indicate the incoming sample-rate.

The ADDA 2408 also functions as masterclock or sync format converter. The signal on WC out always matches the frequency used as source for the A/D or D/D conversion. If the Sync button is put as Int. Sync the internal oscillator functions as synchronising source at all times. These oscillators are very accurate (5 ppm) and the ADDA 2408 hereby guarantees a very accurate sync signal on WC out. The ADDA 2408 now functions as Master Clock generator.

If the Sync button is set as AES 11, Word clock, Super clock, Digital input or Video sync a PLL (Phase Loop Lock) will clean the incoming sync signal for slight variations (Jitter). The signal is here after accessible as a Word Clock or a Super Clock signal on WC out. The ADDA 2408 now functions as Sync signal converter. In the end of the system menu (watch System) the Word Clock signal or a Super Clock signal options on WC out can be adjusted.

Employing video sync. on 30 or 29,97 frame the sample rate can be changed (pull up/down) by setting the synchronising frequency on another frequency than the synchronising pulse the ADDA 2408 receives. Eventually set the video synchronising frequency to the frequency which the final format terminates being published. If, for example, on location a 30 frame video sync is used, knowing a 29.97 frame later is required, put the frequency on the ADDA 2408 to 29,97 even when using a 30 frame sync generator. This results in a higher sample rate frequency. (by 48 kHz sampling the frequency gets 48,048048. However, when finally playing the video by 29,97 frames the sample frequency returns to 48 kHz.) The hereby obtainable frequencies appear in the table below:

Ext. sync. signal	Choise of sample rate on 2408	Choise of video sync on 2408	Sample frequency	Function
30 frame	48 kHz	30 frame	48 kHz	Standard for video
30 frame	48 kHz	29.97 frame	48.048 kHz	NTSC to Film Pull-Up of 48 kHz
30 frame	44.1 kHz	30 frame	44.1 kHz	Standard rate for CD
30 frame	44.1 kHz	29.97 frame	44.144 kHz	NTSC to Film Pull-Up of 44.1 kHz
29.97 frame	48 kHz	30 frame	47.952 kHz	Film to NTSC Pull-Down of 48 kHz
29.97 frame	48 kHz	29.97 frame	48 kHz	Standard rate for video
29.97 frame	44.1 kHz	30 frame	44.056 kHz	Film to NTSC Pull-Down of 44.1 kHz
29.97 frame	44.1 kHz	29.97 frame	44.1 kHz	Standard rate for CD

The Sync Alarm

The ADDA 2408 can tell if several sample rates appear in your system. This for instance might happen if both a hard disk recording system and the ADDA 2408 are set as Int. Sync.

The different 8 channel interfaces detects if the incoming sample frequency is identical to the one leaving. If this is not the case the light diode Sync Alarm commences to light up. If using the ADDA 2408 for sample rate conversion the Sync Alarm is not active for the I/O module being the source to the sample rate conversion.

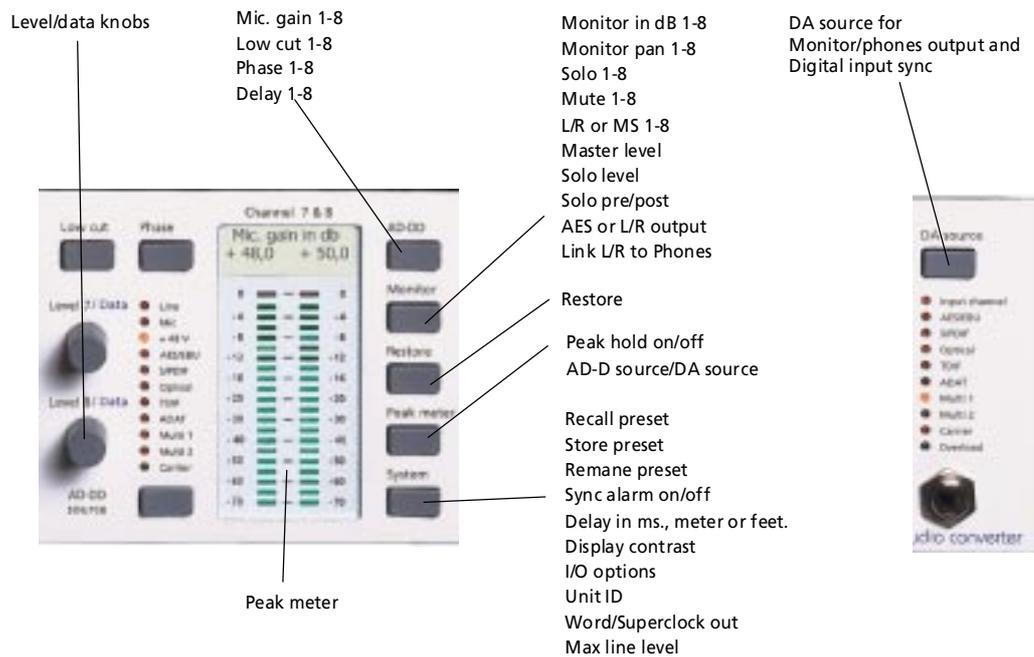
The Sync Alarm can be switched off in the system menu (Watch system menu)

If a chosen ex. sync signal disappear the light diode for Sync Alarm will light up and the light diode for sample-rate will blink.

The AD-DD, Monitor, Peak meter and System menu buttons in general

The buttons for AD-DD, Monitor, Peak meter and System contains sub menus partly operated by activating the buttons and partly by using the Level/Data knobs associated to each channel.

The menu points which on the drawing below has the values 1-8 represents the number of the individual channels. These are operated on the belonging Level/Data knob and the addressed values are exposed on the belonging LCD displays.



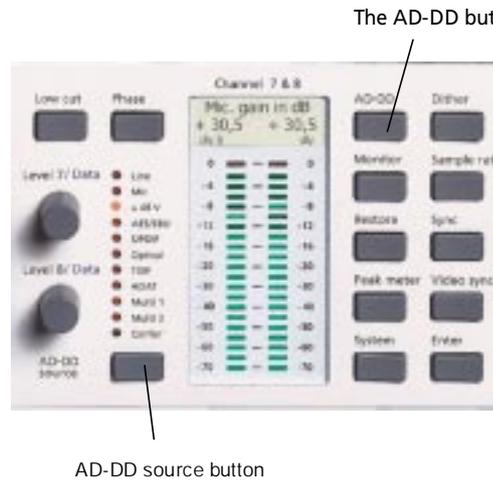
The additional menu points generally works for the entire ADDA 2408. These are operated by the Level/Data knob for channel 7 & 8 and the Enter button on the right side of the System button. The general values and settings are exposed in the LCD display for channel 7 & 8. When one of these menus are activated a text in the additional 3 LCD displays reads: General settings, use level 7 & 8/Data.

Each LCD display have 3 lines. The top line indicates the function adjustable using the belonging Level/Data knob. The next line indicates the value or values of the concerned function. The last line indicates the status of the channel. This line reads if a value for Low cut (L), Phase inversion (F), Delay (dly), mute (M), solo (S) or MS listening (MS) is stated.

The AD-DD button

The AD-DD button contains menu points for Mic. gain, Low cut, Phase inversion and Delay. The individual point are activated in the described order by pressing repeatedly on the AD-DD button.

In AD-DD mode the ADDA 2408 returns to Mic. Gain in dB mode when the Enter Button is activated.



Mic. Gain in dB (1-8):

By pressing the AD-DD button once the text Mic. Gain in dB appears in the LCD displays where Mic, or +48V are selected on the AD-DD source button.

The Mic Preamp itself has a very low equivalent input noise at - 144 dB at +24 dB gain setting. (The +24 dB of analog gain (in the display) correspond to an internal gain of +39 dB).

Set the Mic. Gain for the various channels using the belonging Level/Data Knob. Mic. Gain level stretches between -18 dB and +72 dB in 0,5 dB steps. The selected value for Mic. Gain in dB is exposed for every channel in the 2. line of the LCD display as long as the Mic. Gain in dB is selected.

Analog gain can be added in steps of 3 dB. The following analog gain steps are available: -18, - 12 , - 9, -6, -3, 0, + 3, + 6, + 9, + 12, + 15, + 18, + 21, and + 24 dB of analog gain, for a total of 42 dB of analog gain adjustment.

Low cut (1-8):

Pressing the AD-DD button once more the text Low cut appears in the top line of the LCD displays where Line, Mic or +48 V are selected on the AD-DD button. Low cut is an analogue high pass filter with cutting frequency at 80Hz. On the individual Level/Data knobs the value on or off can be set for the belonging channel. Set the value on by turning the Level/Data knob clock wise and off by turning counter clock wise.

The selected value for Low cut (on/off) can be read in the 2. line of each LCD display as long as the function Low cut is selected.

If Low cut is selected for an individual channel a **L** is shown in the 3. line of the LCD display of the belonging Channel.

Phase inversion (1-8):

By an additional press on the AD-DD button the text Phase Inv. appears in the top line of all LCD displays. Phase inversion turns the phase on the input signal. On the individual Level/Data knobs the value on or off can be set for the belonging channel. Set the value on by turning the Level/Data knob clock wise and off by turning counter clock wise.

The selected value for Phase inversion (on/off) can be read in the 2. line of each LCD display as long as the function Phase inversion is selected.

If Phase inversion is selected for an individual channel a **F** is shown in the 3. line of the LCD display of the belonging Channel.

Delay in ms, meters or feet (1-8):

By an additional press on the AD-DD button the text Delay in ms, meters or feet appears in the top line of all LCD displays. Which unit is used depends on the settings in the system menu.

A delay may be added on all formats (analogue or digital inputs) selected on the AD-DD source button.

The delay time can be adjusted between 0,0 – 100,0 ms (optional 250,0 ms). On the individual Level/Data knobs the value can be set for the belonging channel. The selected value for Delay (0,0 – 100,0) can be read in the 2. line of each LCD display as long as the function Delay is selected.

If Delay is selected for an individual channel a **Dly** is shown in the 3. line of the LCD display of the belonging Channel.

The Monitor button

The Monitor button serves to create a stereo mix of the 8 channel signal selected on the DA source button. The Mix will be accessible on L/AES3 out / R placed on the back plate of the ADDA 2408 as well as the Phones plug on the front plate.

Activating the Monitor button provokes menu points for Monitor in dB (1-8), Monitor pan (1-8), Solo (1-8), Mute (1-8), L/R or MS (1-8), Master level, Solo level, L/R or AES out og Link L/R to Pho.. The individual menu points are activated in that order by repeated pressing on the Monitor button.

In Monitor mode the ADDA 2408 returns to Monitor in dB (1-8) when the Enter button is activated.



Monitor Level in dB (1-8):

By pressing the Monitor button once the text Monitor in dB reads in the upper line of all LCD displays. Monitor level in dB for the various channels are adjusted on the belonging Level/Data knobs. Monitor level in dB can be adjusted between -120 – 0,0. The selected value for Monitor level in dB can be read in the 2. line of each LCD display as long as the function Monitor level in dB is selected.

Monitor pan (1-8):

By an additional press on the Monitor button the text Monitor pan appears in the top line of all LCD displays. This function accesses a pan between left and right of the stereo monitor output signal. Monitor pan can be adjusted between -50 L and +50 R on the belonging Level/Data knobs representing full pan between L and R. The selected value for Monitor pan can be read in the 2. line of each LCD display as long as the function Monitor pan is selected.

Solo (1-8):

By an additional press on the Monitor button the text Monitor Solo appears in the top line of all LCD displays. This function accomplishes to hear Monitor Solo on one or more channel.

On the individual Level/Data knobs the value on or off can be set for the belonging channel. Set the value on by turning the Level/Data knob clock wise and off by turning counter clock wise. The selected value for Monitor Solo (on/off) can be read in the 2. line of each LCD display as long as the function Monitor Solo is selected.

If Monitor Solo is selected for an individual channel an S is shown in the 3. line of the LCD display of the belonging Channel. The Monitor Solo Level is adjusted further down the Monitor menu as well as the option of listening Solo pre or Solo post monitor level can be adjusted in the System menu.

Mute (1-8):

If Mute is selected for an individual channel an M is shown in the 3. line of the LCD display of the belonging Channel.

L/R or MS stereo (1-8):

By an additional press on the Monitor button the text L/R or MS appears in the top line of all LCD displays. This function allows for 2 channel at a time to be decided as normal mono channels or to be able to decode a MS stereo signal.

On the individual Level/Data knobs the values L/R or MS can be set for the belonging channel. Set the value L/R by turning the Level/Data knob clock wise and MS by turning counter clock wise. The selected value for L/R or MS (on/off) can be read in the 2. line of every LCD display as long as the function L/R or MS is selected.

When MS stereo is selected for 2 or more channels the Enter button is required to be activated for returning to the monitor level in dB. The function for Monitor level in dB has now been changed allowing Level/Data knobs 1,3,5 and 7 to adjust the M stereo level, while the 2,4,6 and 8 Level/Data knobs adjusts the S level. When MS is selected S is reduced 10 dB compared to M.

If L/R or MS is selected for 2 or more channels an MS will be exposed in the 3. line of the belonging LCD displays.

Master output level :

By an additional press on the Monitor button the text Master level /dB appears in the top line of the LCD display for channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

On the Level 8/Data knob Master output level /dB is adjusted between -120 and 0,0. Value 0,0 refers to full scale of the analogue L/R output on the back plate. This output is adjusted between +12 and +27 dBu at 0 dBFS. (Watch section on adjustment of analogue outputs)

The selected value for Master output level in dB can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Master output level /dB is selected.

Solo level:

By an additional press on the Monitor button the text Solo level /dB appears in the top line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

On the Level 8/Data knob Solo level /dB is adjusted between -60,0 and +18,0 dB. The selected value for Solo level /dB can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Solo level /dB is selected.

Solo Pre/Post:

By an additional press on the Monitor button the text Solo Pre/Post appears in the top line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

The Solo Pre/Post is used to decide if the Solo function on the monitor mix is Solo Pre or Solo Post.

On the Level 8/Data knob the Solo Pre/Post is set to Pre or Post. Set the value Pre by turning the Level 8/Data knob clock wise and Post by turning counter clock wise. The selected value for Solo Pre/Post can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Solo Pre/Post is selected.

L/R or AES:

By an additional press on the Monitor button the text L/R or AES appears in the top line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

The L/R or AES is used to decide if the 2 XLR outputs on the back plate (L/AES output / R) are analogue L/R or digital AES. If AES is selected R will have no signal.

On the Level 8/Data knob the L/R or AES is set to L/R or AES. Set the value L/R by turning the Level 8/Data knob clock wise and AES by turning counter clock wise. The selected value for L/R or AES (L/R / AES) can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function L/R or AES is selected.

Link L/R to Phones:

By an additional press on the Monitor button the text Link L/R to Pho appears in the top line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

On the Level 8/Data knob the Link L/R to Phones is set to on or Off. Set the value on by turning the Level 8/Data knob clock wise and Off by turning counter clock wise.

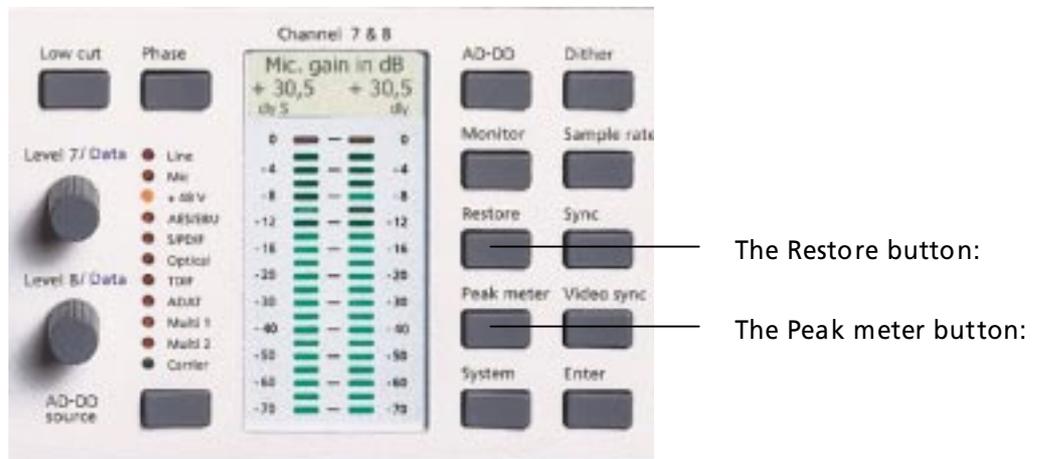
The selected value On or Off for Link L/R to Phones can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Link L/R to Phones is selected.

Link L/R to Phones sets the level on the 2 XLR outputs on the back plate (L/AES output / R) on the Phones knob on the front. The master output from the ADDA 2408 can hereby be adjusted on Phones knob on the front. The relation between L/AES output /R and Phones is remain unchanged. If relation between L/AES output /R and Phones is required the Link L/R to Phones must be Off.

The Restore button

The Restore button deletes Phase inversion, MS stereo, Low cut, Mute, Solo and Delay settings on all 8 Channels. Master level is set at 0,0 dB, Solo level at 0,0 dB, Monitor level at -30 dB on all channels and Mic. level at +15 dB on all channels.

When the Restore button is activated the text reads: Are you sure you want to restore? Wishing to restore press Enter. If not press any other button.



The Peak meter

The ADDA 2408 is equipped with very precise Fast Peak Programme Meters (PPM) for monitoring the level of the analog or digital input signal. The meter has a hold time of 1 second.

The peak meters reads out the fast peak value of the input signal in dB full-scale (dBFS), where reading is relative to the full-scale value of the digital signal. This means that if the XLR input analog level is adjusted to +18 dBu, which is 6.16 Volt RMS, this will give a 0dBFS reading. A red LED indicates 0dBFS. From 0 dBFS to -20dBFS the readout resolution is 2dB. From -20 dBFS to -70dBFS the readout resolution is 5dB. From -2 dBFS to -70dBFS the readout is indicated by green LEDs.

The peak meters can store the highest peak in a recording. It is recommended to adjust the input signal to a maximum, peak value of -12dBFS. This means that the meter normally reads out 'green' values. In this way a 12 dB headroom is preserved. The meter will read down to -70 dBFS, but still the A/D converter will have a dynamic range better than 117 dB (A). With a 12 dB headroom, the noise floor will thus be below 105 dBrel (A).

The Peak meter button

The Peak meter button sets the inputs of the peak-meters and makes it possible to store the highest value and to release the values stored on the peak-meters.

If set to store the highest peak value, the peak meter is cleared every time the peak meter is operated. When the peak meter button has been operated return to the previous menu by pressing Enter.

AD-DD or DA source:

By pressing the Peak meter button once the text AD-DD or DA reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

The AD-DD or DA source sets the peak meters whether to show the inputs selected on the AD-DD source or the input selected on the DA source button.

The AD-DD source or DA source is set using the Level 8/Data knobs. Set the value for AD-DD source turning the Level 8/Data knob clockwise and for DA source turning counter clockwise. The selected value AD-DD or DA can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function AD-DD or DA source is selected.

Peak store On/Off:

By pressing the Peak meter button once more the text Peak store reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

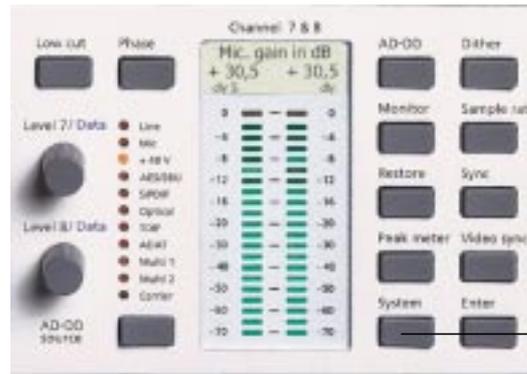
The Peak store sets the peak meters to show the highest value. (If the highest peak is stored the highest peak is cleared every time the peak meter is operated)

The Peak store On/Off is set using the Level 8/Data knobs. Set the value on turning the Level 8/Data knob clockwise and off turning counter clockwise. The selected value AD-DD or DA can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Peak store On/Off is selected.

The System button

The System menu allows to recall, save and rename all in all 8 presets (from preset no. 2 to preset no. 9), further to switch Sync Alarm on/off, to indicate the measurements of Delay times (ms, meters or feet), adjust contrast in the 4 LCD displays, set different values for the related I/O modules, set an unit ID for the ADDA 2408 and to set the ADDA 2408 to send a Word Clock or a Super Clock signal on WC out.

Having carried out adjustments in the system menu press the Enter button. From here the ADDA 2408 returns to origin as before the system menu was operated.



The System button:

Recall preset:

By pressing the System button once the text recall preset reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

The recall preset restores previous presets. The individual are structured from no. 1 to no. 9. Preset no. 1 is Default on which no preset can be stored. Default functions as a standard start up preset. Scroll through the stored presets in Recall preset mode using the Level 8/Data knobs. At the demanded preset, press the Enter button. The preset is hereby stored in the ADDA 2408 which returns to the menu prior to activating the System button. A preset holds all user settings on the ADDA 2408.

Store preset:

By pressing the System button once more the text Store preset reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

Store preset stores all user adjustments of the ADDA 2408 at the time of store. The individual presets are structured from no. 1 to no. 9. Preset no. 1 is Default on which no preset can be stored. Default functions as a standard start up preset. Scroll through the stored presets 2 to 9 in Store preset mode using the Level 8/Data knobs. At the demanded no. where a storage of the present setting is wanted to be placed hit the Enter button. The preset is now stored in the ADDA 2408 and can be recalled at any time. At the same time the ADDA 2408 returns to the menu prior to activating the System button.

Rename preset:

By pressing the System button once more the text Rename preset reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 7-8/Data knob.

Rename preset names or renames a preset no. preset in the ADDA 2408. A preset name consists of a number from 1 to 99 including up to 12 characters of own choice. In Rename preset using the Level 8/Data knob scroll through a variety of characters, changed using the Level 7/Data knob. Having named the preset press Enter. The preset name is now changed in the ADDA 2408 and can be recalled at any time. Having pressed the Enter button the ADDA 2408 returns to the menu prior to activating the System button.

To store a new setting of the ADDA 2408 with a new name commence by Store Preset under a preset no. 2 – 99 and name this preset.

Sync alarm on/off:

By pressing the System button once more the text Sync alarm reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

The Sync alarm on/off can be set on/off using the Level 8/Data knob. Set the value on by turning the Level 8/Data knob clock wise and Off by turning counter clock wise. The selected value Sync alarm on/off can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Sync alarm On/Off is selected.

Delay in ms, meter or feet:

By pressing the System button once more the text Delay in ms, meter or feet reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

Delay in ms, meter or feet can be set to ms, meter or feet using the Level 8/Data knob. Set the value ms, meter or feet by turning the Level 8/Data knob. The selected value Delay in ms, meter or feet can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Delay in ms, meter or feet is selected.

Display contrast:

By pressing the System button once more the text Display contrast reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

Display contrast can be set between 0 and 10 using the Level 8/Data knob. The selected value Display contrast can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Display contrast is selected.

I/O Options:

By pressing the System button once more the text I/O Options reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

Scroll through the various installed I/O cards using the Level 8/Data knob and choose the options using the Level 7/Data knob. For instance double wire/double speed. The selected value I/O Options can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function I/O Options is selected.

Unit ID:

By pressing the System button once more the text Unit ID reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

Unit ID can be set between 1 and 16 using the Level 8/Data knob. The Unit ID is used when remote controlling the ADDA 2408 via the attached software program. If 2 or more ADDA 2408's are remote controlled make sure to have separate Unit ID's. The selected value Unit ID can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Unit ID is selected.

Word/SuperClock out:

By pressing the System button once more the text Word/SuperClock out reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

Word/SuperClock out sets the which clock signal to be WC out on the back plate of the ADDA 2408.

Word/SuperClock out can be set as WordClock out or SuperClock out using the Level 8/Data knob. Set the value WordClock by turning the Level 8/Data knob clock wise and SuperClock by turning counter clock wise. The selected value Word/SuperClock can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Word/SuperClock is selected.

Max line level in:

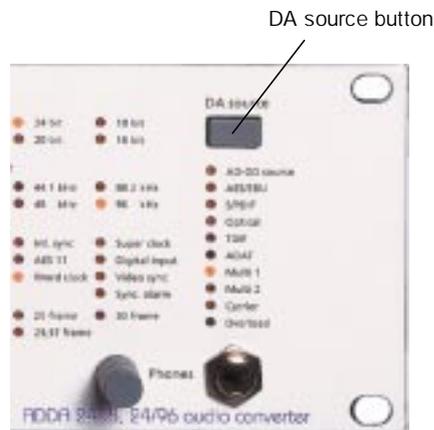
By pressing the System button once more the text Max line level in out reads in the upper line of the LCD display belonging to channel 7 & 8. The other 3 displays reads General settings, use Level 8/Data knob.

Max line level in is adjusting the input line of the 8 analogue input channels.

On the Level 8/Data Knob Max line level in can be set to +21 dBu or +30 dBu. To set the value +21 dBu turn the Level 8/Data Knob clockwise and for +30 dBu turn counter clockwise. The selected value Max line level in can be read in the 2. line of the LCD display belonging to the channel 7 & 8 as long as the function Max line level in is selected. If +21 dBu is selected the ADDA 2408 can be adjusted to a line input level between +12 dBu and +21 dBu. If a +30 dBu is selected the ADDA 2408 can be adjusted to a line input level between +21 dBu and +30 dBu. To adjust line levels see appendix on Adjustment of line levels.

The DA source button

The DA Source button is used to select which signal to be employed on the Monitor mixer and in D/A converted shape be accessible on Phones and L/AES3 / R outputs.



Select from AD-DD source, AES/EBU, S/PDIF, Optical, TDIF, ADAT, Multi 1 or Multi 2. To select a digital format the belonging I/O card must be installed.

To see which signal selected watch the line of light diodes below the DA source button. In case of signal on the selected digital interface the green light diode market Carrier lights.

Selecting 8 channels with the DA source button and mixing these down to 2 digital channels overloads might occur. In this case the red light diode market Overload lights. To avoid overloads turn down one or more channels in the Monitor menu.

Digital I/O's outside AES/EBU, S/PDIF, T/DIF or ADAT for instance ProTools are installed as Multi 1 or Multi 2.

If 2 AES/EBU, S/PDIF, T/DIF or ADAT I/O' cards installed the one is installed as AES/EBU, S/PDIF, T/DIF or ADAT and the other as Multi 1 or Multi 2. (Watch section 2)

The Phones knob

The exit level on the Phones Jack on the front plate is adjusted by the Phones knob. The ADDA 2408 holds a powerful Head phones amplifier capable of operating 6 sets of 300 ohms head phones simultaneously at a reasonable sound pressure. Very intense sound may damage ears thus the importance to keep sound level on the Phones amplifier at a safe level at all times.

The Phones knob may also serve to control the exit levels on the XLR connections L/AES / R on the back plate. For more information on this watch section on Monitor button.



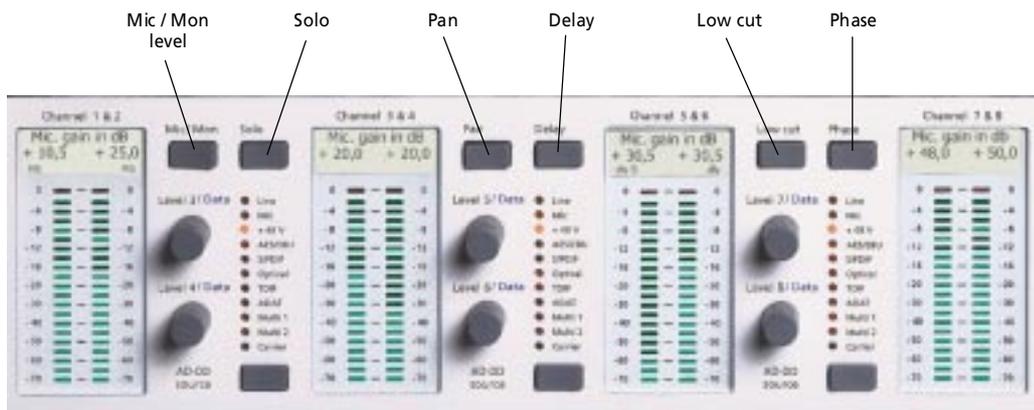
Phones knob

Shortcuts

On top between the 4 LCD displays 6 buttons Mic/Mon., Solo, Pan, Delay, Low cut and Phase are found. These buttons are shortcuts to some of the functions belonging to the AD-DD and Monitor menus.

The shortcuts are only active when the shortcut button is pressed. When letting go of a shortcut button the ADDA 2408 returns to the function prior to pressing a shortcut. If for instance the Mic. gain in dB is on and the shortcut is activated the belonging function may be adjusted using the belonging Level/Data knob. Letting go of the shortcut button returns the function to Mic. gain in dB. Identifying the location is therefore easy at all times. It is recommended for the ADDA 2408 to be in Mic. gain in dB or Master level mode while recording.

The Mic./Mon button shifts between Mic. gain in dB and Monitor in dB. Being in Mic. gain in dB or one of the AD-DD buttons sub menus, pressing the Mic./Mon shifts to Monitor in dB while pressed. Being in Monitor in dB or one of the Monitor buttons sub menus, pressing the Mic./Mon shifts to Mic. gain in dB while pressed. If either in peak meter or System menu the Mic./Mon button results the ADDA 2408 to return to Mic.gain in dB or Monitor in dB depending on the which of the 2 conditions latest operated. A continuously pressed Mic./Mon button returns the system to the latest operated function. Letting go of the button results for the system to return to the previous condition.

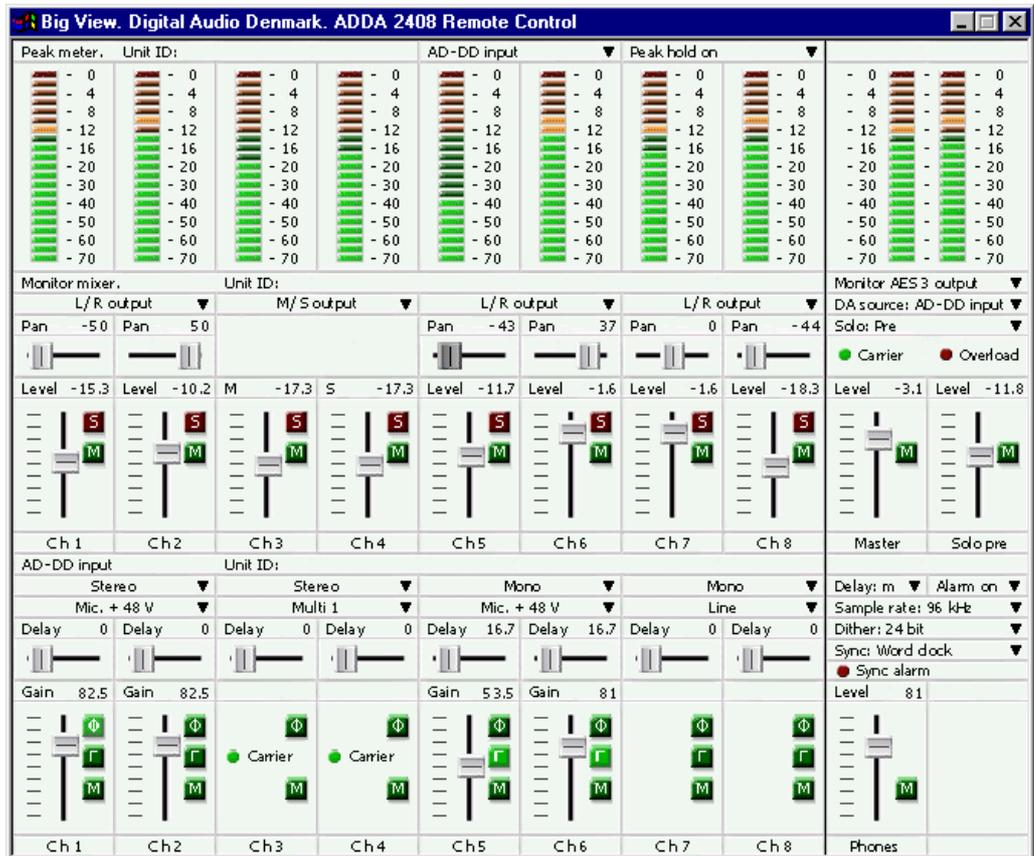


The functions off the buttons Delay, Low cut and Phase watch the related functions in AD-DD button section.

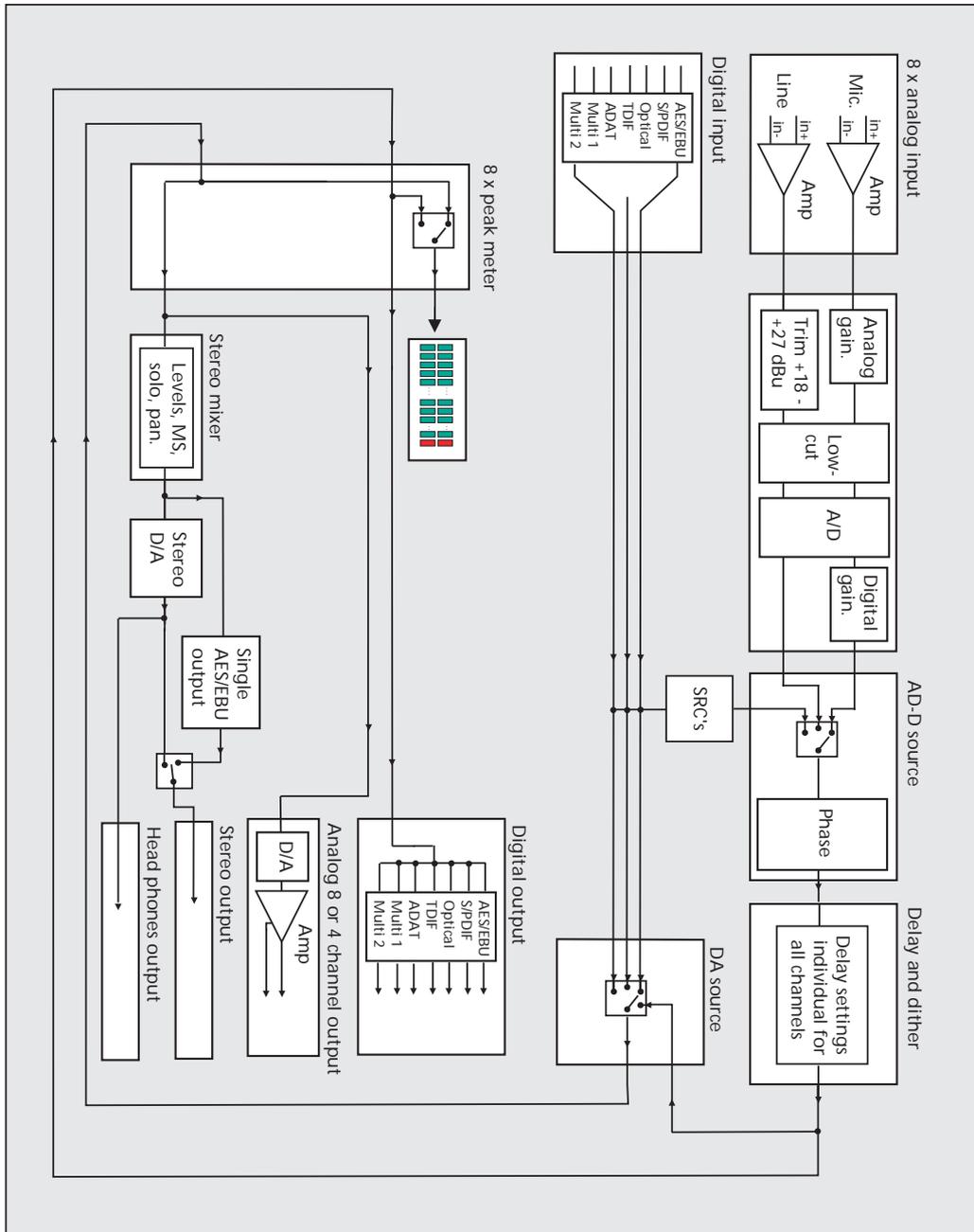
The functions off the buttons Solo and Pan watch the related functions in The Monitor button.

PC program

All functions on the ADDA2408 can be remote controlled with a comprehensible PC program via MIDI or USB. The newest version of the program can be downloaded at <http://www.digitalaudio.dk/downloads.htm>



Block Diagram ADDA 2408



5. Specifications:

Analog to Digital Audio Conversion, Line input

Cross-talk:

Cross-talk at 997 Hz, -3 db Fs < 115 dB

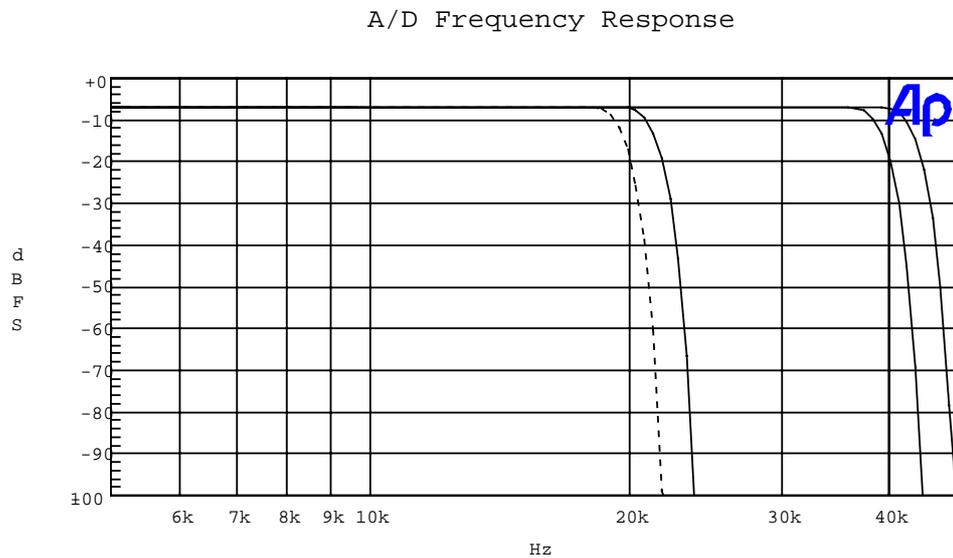
Processing delay:

Processing delay. < 1,0 ms

Frequency response

- Fs 44,1kHz, 18 Hz-18,3 kHz:±0.1dB
- Fs 48kHz, 18 Hz-20,0 kHz:±0.1dB
- Fs 88,2kHz, 18 Hz-36,0 kHz:±0.1dB
- Fs 96kHz, 18 Hz-39,0 kHz:±0.1dB

The frequency response has a Stop-band attenuation of 117 dB at half the sampling frequency, thus eliminating high-frequency aliasing

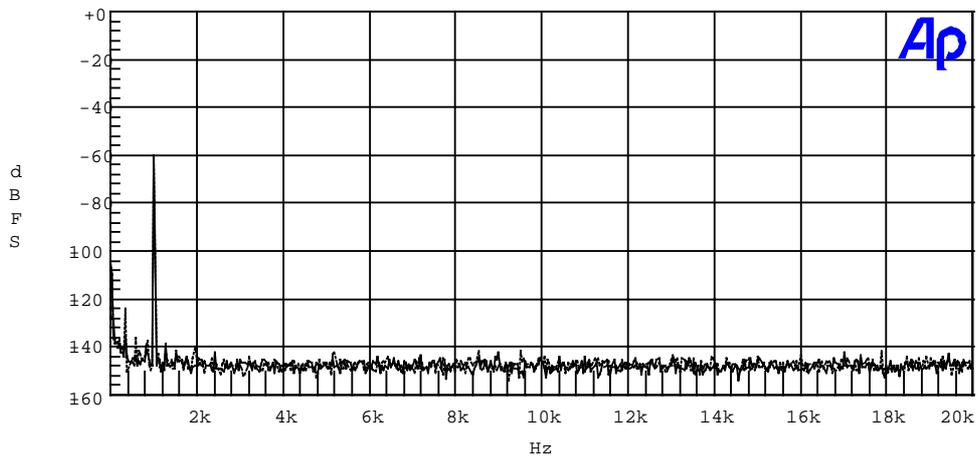


The frequency response is measured at 44.1 kHz, 48 kHz, 88.2 kHz and 96 kHz.

Dynamic range

The Dynamic Range is better than 117 dB (A-weighted) at 44,1 kHz sampling. The dynamic range with a -60 dB FS input signal is better than 140 dB measured as a FFT.

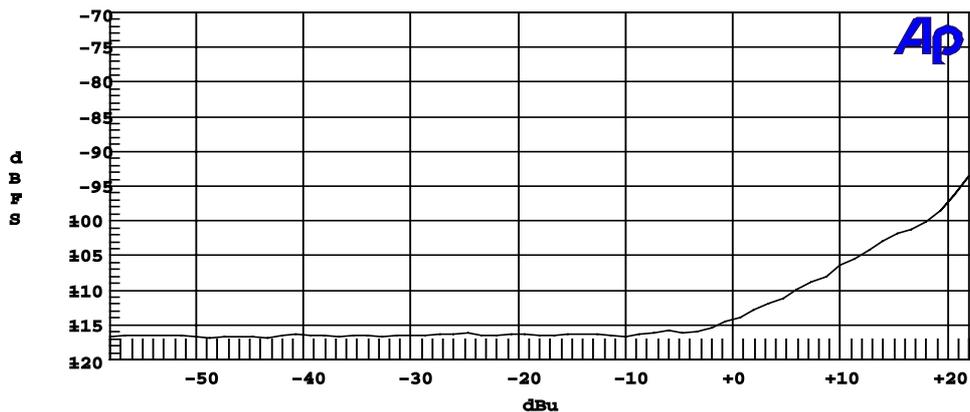
A/D. Dynamic range. FFT @48 kHz



THD+Noise

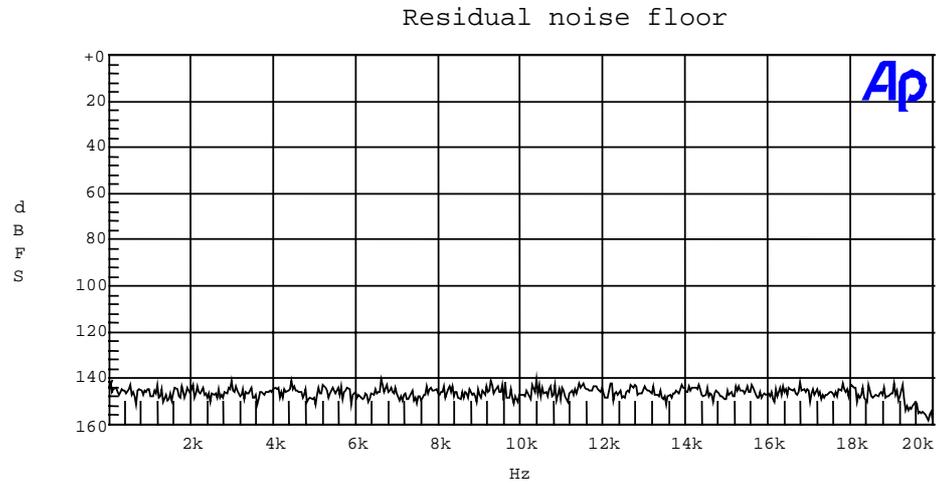
The THD+N vs. input amplitude is shown in the figure below. At a nominal input level of 12 dBu with the input adjusted to 0 dBfs at +22 dBu, the THD+N is below -104 dBfs.

A/D THD+N vs Ampl at +22 dBu max



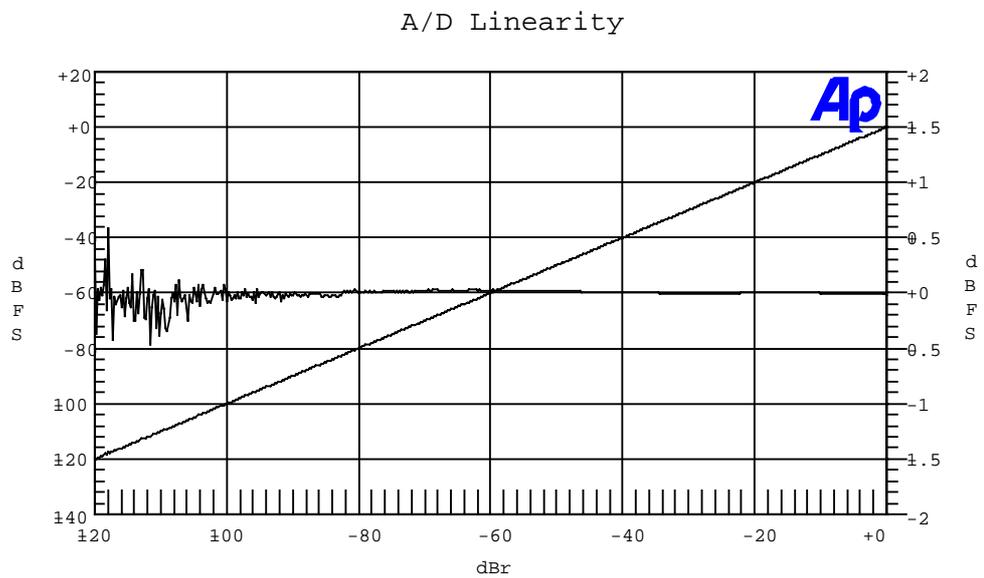
Residual noise

The residual noise floor is better than -140 dBfs.



Linearity

Linearity better than $\pm 0,1$ dB down-to -100 dB, and $\pm 0,5$ dB down to -110 dBfs



Analog to Digital Audio Conversion, Mic input

Analog Gain steps

Analog gain can be added in steps of 3 dB. The following analog gain steps are available: -18, -12, -9, -6, -3, 0, +3, +6, +9, +12, +15, +18, +21, and +24 dB of analog gain, for a total of 42 dB of analog gain adjustment.

Mic. pre equivalent noise

- -12 dB gain: < -118 dB (A)
 - +3 dB gain: < -133 dB (A)
 - +24 dB gain: < -144 dB (A)
- (At input shorted with a 40 Ohm resistor).

Cross-talk:

Cross-talk at 997 Hz, -3 db Fs, + 7 dBu < 120 dB

Processing delay:

Processing delay. < 1,0

Frequency response

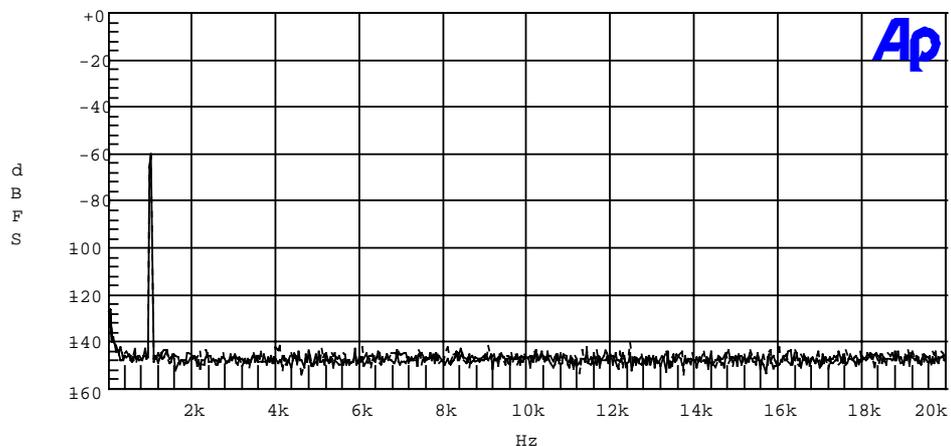
- Fs 44,1kHz, 18 Hz-18,3 kHz:±0.1dB
- Fs 48kHz, 18 Hz-20,0 kHz:±0.1dB
- Fs 88,2kHz, 18 Hz-36,0 kHz:±0.1dB
- Fs 96kHz, 18 Hz-39,0 kHz:±0.1dB

The frequency response has a Stop-band attenuation of 117 dB at half the sampling frequency, thus eliminating high-frequency aliasing.

Dynamic range

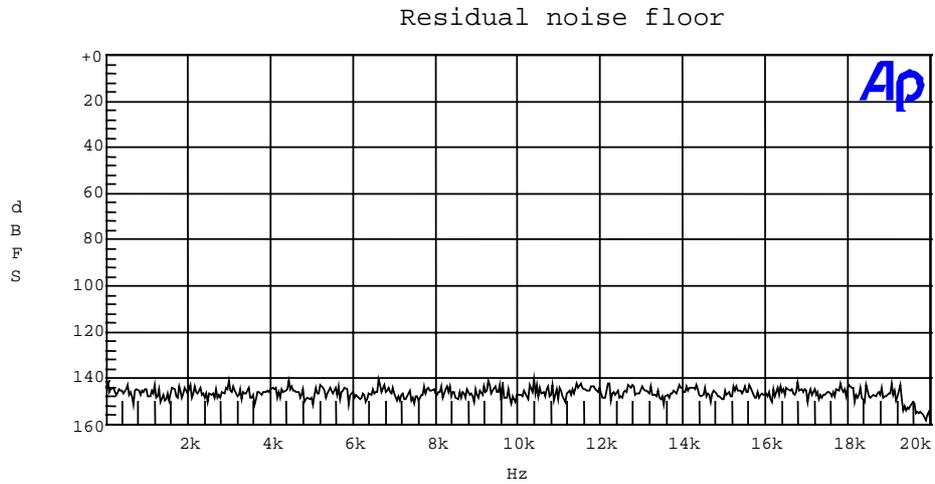
The Dynamic range is better than 117 dB (A-weighted) at 48 kHz sampling. The dynamic range with a -60 dB FS input signal is better than 140 dB measured as a FFT.

Mic. A/D Dynamic range. FFT @48 kHz



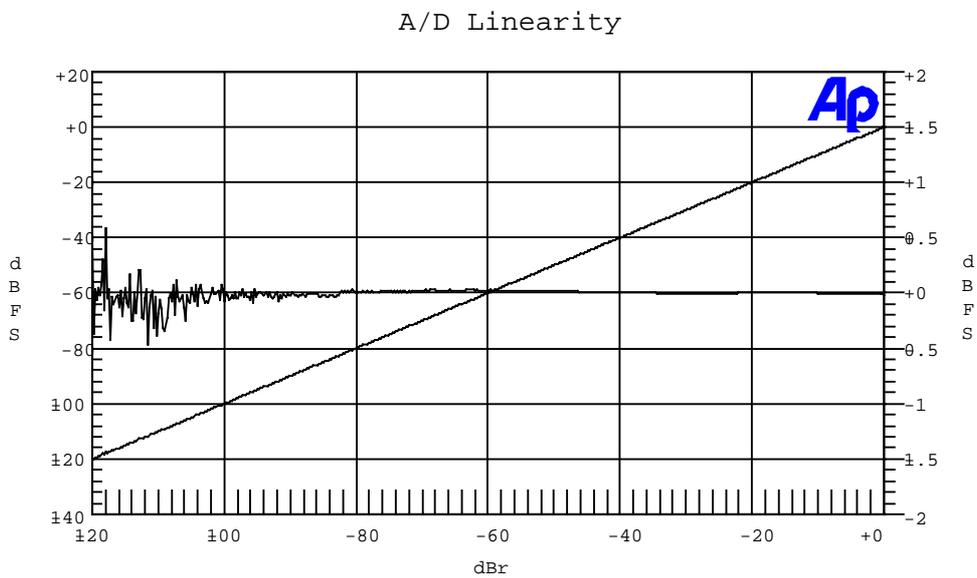
Residual noise

The residual noise floor is better than -140 dBfs.



Linearity

Linearity better than $\pm 0,1$ dB down to -100 dB, and $\pm 0,5$ dB down to -110 dBfs



Digital to Analog Audio Conversion

Cross-talk:

Cross-talk at 997 Hz, -0 db Fs, < 120 dB

Processing delay:

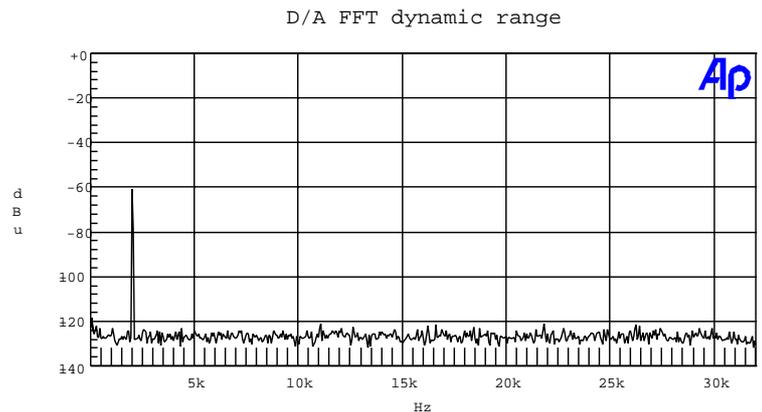
Processing delay. < 1,0 ms

Frequency response

- Fs 44,1kHz, 18 Hz-20.5 kHz:±0.1dB
- Fs 48kHz, 18 Hz-22.7 kHz:±0.1dB
- Fs 88,2kHz, 18 Hz-30.5 kHz:±0.1dB
- Fs 96kHz, 18 Hz-30.5 kHz:±0.1dB

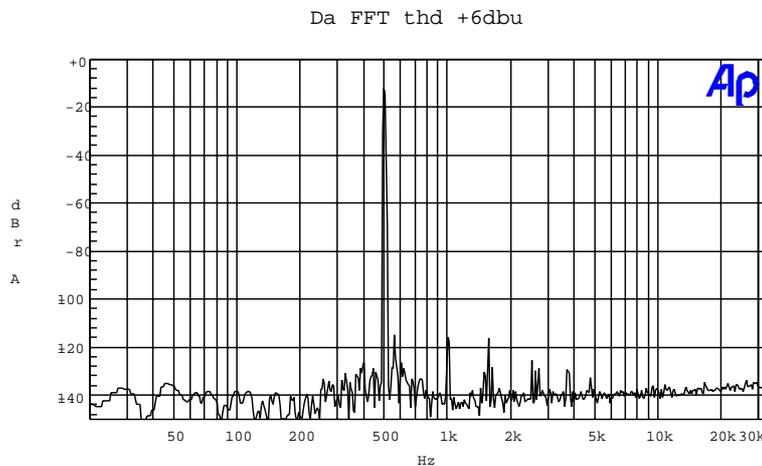
Dynamic range

The Dynamic range is better than 117 dB (A-weighted) at 48 kHz sampling.



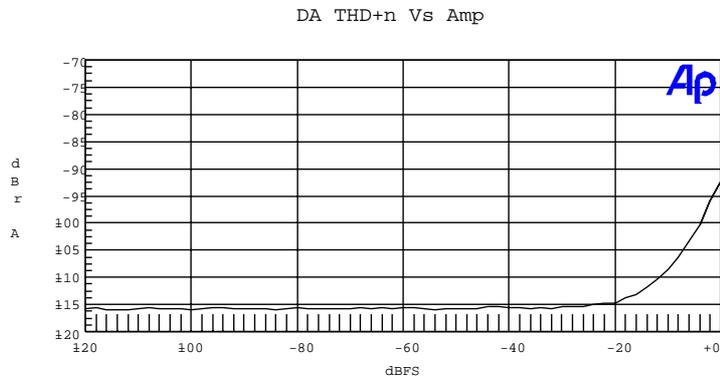
THD+Noise

The THD+N vs. input amplitude is shown in the figure below. At a nominal input level of 6 dBu, the THD+N is below -93 dB.



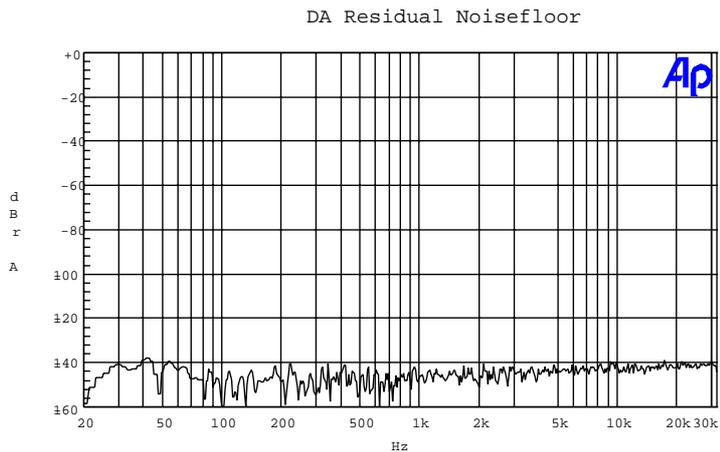
THD+Noise v/amplitude

The THD+N vs. input v/amplitude is shown in the figure below.
The THD+N is below -93 dBr.



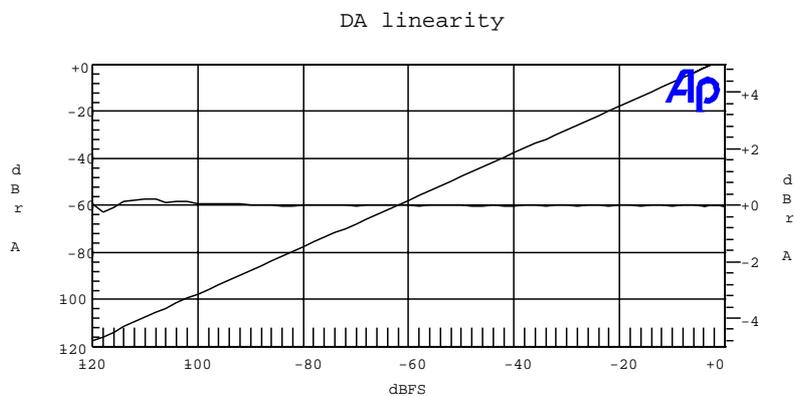
Residual noise

The residual noise floor is better than -140 dB.



Linearity

Linearity better than +/- 0,5 dB down-to -105 dB



Digital to Digital Audio Conversion (D/D)

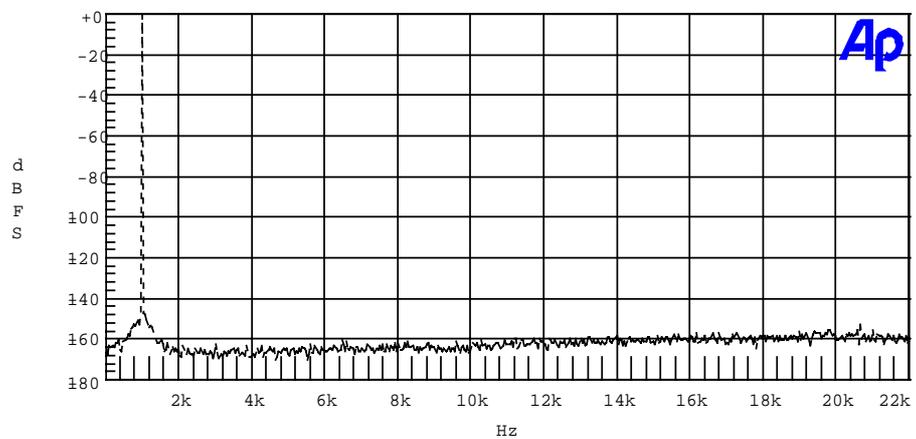
Processing delay:

Processing delay. $< 1,0$

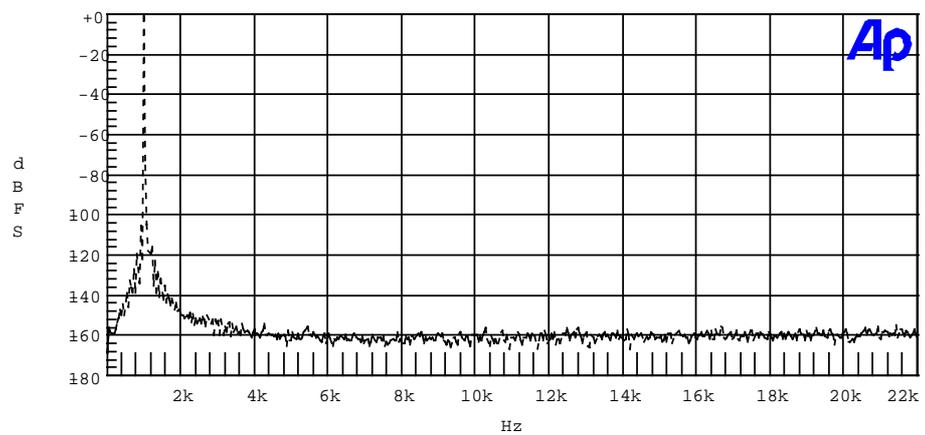
Dynamic range

The Dynamic range is better than 133 dB (A-weighted) at sample-rate conversion between all sample-rates. .

D/D FFT. 96 to 44.1 kHz



D/D FFT. 44.1 to 88.2 kHz



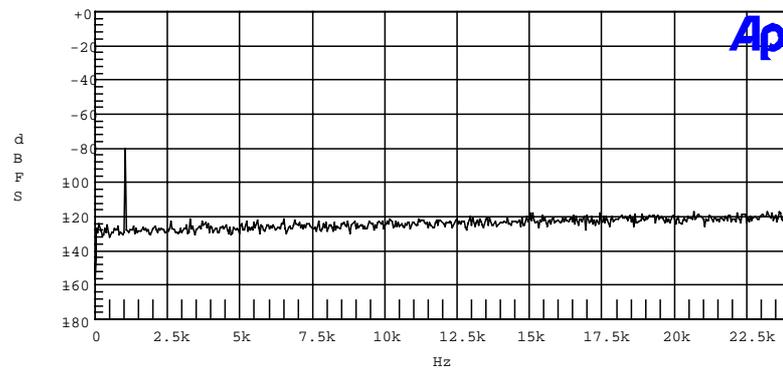
Dither

Dynamic range

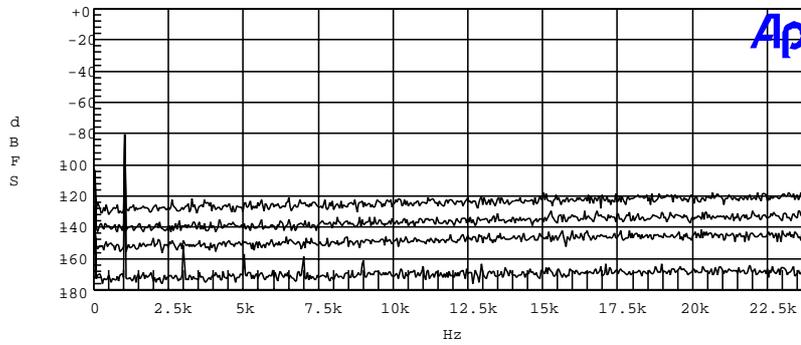
- 16 bit 95 dB (A)
- 18 bit 107 dB (A)
- 20 bit 119 dB (A)
- 24 bit 139 dB (A)

The measuring is done D/D only adding dither to the signal at -80 dBfs

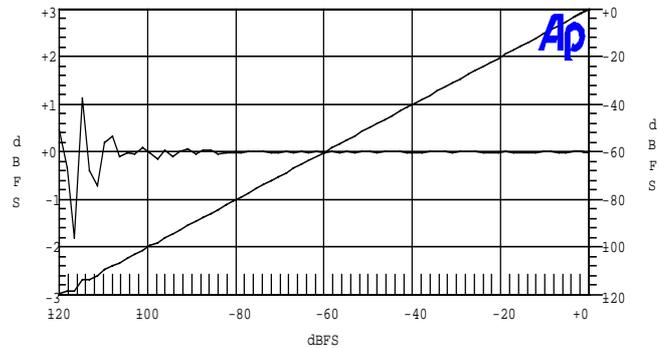
Dither FFT 16bit



Dither FFT, 16, 18, 20 and 24 bit



Dither Linearity 16 bit



General specifications

A/D CONVERSION, LINE INPUT

Resolution	24 bit
Sample-rates	44.1, 48, 88.2, 96 kHz
Dither bit	Psycho acoustic, 16,18, 20
Dynamic range (A)	> 117 dB
THD+N (A) at +12 dBu, 22 dBu max	< -104 dBfs
Cross talk	< -115 dB
Input impedance	> 15 k Ω
Max. input level	+12 to 30 dBu, adjustable
Processing delay	< 1.0 ms
Connectors	XLR (pin 2 hot)

A/D CONVERSION, MIC. INPUT

Resolution	24 bit
Sample-rates	44.1, 48, 88.2, 96 kHz
Dither bit	Psycho acoustic, 16,18, 20
Dynamic range (A)	> 117 dB
THD+N (A) at +12 dBu, 18 dBu max	< -104 dBfs
Mic pre Equivalent Noise(A), +24 dB analog gain	< -144 dB
Cross talk	< -120 dB
Input Impedance	> 15 k ohm
Max. Input Level Mic (Balanced)	+ 24 dBu
Gain	-18 to +90 dB
Freq. response deviation	< 0,05 dB
Processing delay	< 1.0 ms
Connectors	XLR (pin 2 hot)

D/A CONVERSION, LINE OUTPUT

Resolution	24 bit
Sample-rates	44.1 to 100 kHz
Dynamic range (A)	> 117 dB
THD+N (A)	< - 93 dB
Cross talk	< -120 dB
Input Impedance	< 40 ohm
Max. output Level Mic (Balanced)	+12 to+27 dBu, adjustable
Processing delay	< 1.0 ms
Connectors	XLR (pin 2 hot)

D/D CONVERSION

Resolution	24 bit
Sample-rates	44.1 to 96 kHz
Dynamic range (A)	> 133 dB
Processing delay	< 1.8 ms

FREQUENCY RESPONSE A/D

Frequency response A/D. Stop-band attenuation 117 dB at half of the sampling freq.

Fs 44.1kHz, 18-18.3 kHz:	±0.1dB
Fs 48kHz, 18-20 kHz:	±0.1dB
Fs 88.2kHz, 18-36 kHz:	±0.1dB
Fs 96kHz, 18-39 kHz:	±0.1dB

FREQUENCY RESPONSE D/A

Fs 44.1kHz, 18-20.5 kHz:	±0.1dB
Fs 48kHz, 18-22.7 kHz:	±0.1dB
Fs 88.2kHz, 18-30.5 kHz:	±0.1dB
Fs 96kHz, 18-30.5 kHz:	±0.1dB

DIGITAL INPUTS AND OUTPUTS

Formats:	AES/EBU, S/PDIF, Optical S/PDIF, TDIF, ADAT, ProTools TDM
Ext. Sync input:	AES11, WordClock, VideoClock
Ext. Sync output	WordClock, SuperClock

HEADPHONE AMPLIFIER

RMS output power (600W)	50 mW
Connectors	¼ " stereo phone jack

GENERAL

EMC complies with:	EN 50082, and EN 50022
Operating Temperature:	+5 to 45° C
Dimensions (w,h,d):	19", 2U, 275 mm
Weight:	5.8 kg
Mains Voltage:	95 to 240VAC
Power Consumption:	20 Watts

Due to our policy of continuous improvement of our products, Digital Audio Denmark reserves the right to make feature and specifications changes without notice.

6. References

- AES3 AES3-1992: "AES Recommended Practice for Digital Audio Engineering – Serial transmission for two-channel linearly represented digital audio data" Published by Audio Engineering Society, Inc.
AES3 is a specification of the AES/EBU digital audio format. AES3 specifies the frame format for the audio data in both professional and consumer mode, as well as the electrical interface for the professional balanced XLR connection. AES3 is however primarily a professional standard. The detailed specification for the consumer interface is IEC 60958.
- IEC958 IEC 60958 1989: "Digital Audio Interface". Published by the International Electrotechnical Commission.
IEC958 is a specification of the consumer AES/EBU digital audio format also referred to as S/PDIF. IEC958 specifies the frame format for the audio data in consumer mode, as well as the electrical interface for the unbalanced data connection.
- AES11 AE11-1991: "AES Recommended Practice for Digital Audio Engineering – Synchronization of Digital Audio Equipment in Studio Operations". Published by Audio Engineering Society, Inc.
AES11 is a recommendation for synchronization of digital audio signals. Recommendations are made concerning the accuracy of sample clocks as embodied in the digital interface signal, and concerning the use of this as a convenient reference signal.
- EN50082 EN 50082-1: "Electromagnetic compatibility – generic immunity standard, part 2: Residential, commercial and light industry". Published by CENELEC 1992. Note that the standard also covers EN 50081-1 (part 1).
EN 50082-1 is the standard specifying the requirements for CE compliance concerning Immunity following the EEC provisions of directives 92/31/EEC.
- EN500022 EN 500022-1: "Limits and methods of measurement of radio interference characteristics of immunity" Published by CENELEC 1987.
EN 500022-1 is the standard specifying the requirements for CE compliance concerning Emission following the EEC provisions of directives 89/336/EEC.
- IEC 268-10 IEC 60268-10: edition 2.0 "Sound system equipment – part 10. Peak programme level meters" Published by the International Electrotechnical Commission 1991-03.
EN 60268-10 is the standard specifying the requirements of audio-frequency peak programme level meters, for use in equipment for broadcasting, sound reinforcement, sound recording, and household entertainment.
- DITHER The dither of the ADDA 2402 is implemented by using a psychoacoustic noise shaping filter for truncating the 24 bit audio word to 16, 18 or 20 bits, while the 24 bit sound quality is preserved. A detailed description of the dither algorithm can be found in the AES paper: "Psychoacoustically Optimal Noise Shaping" by Robert A. Wannamaker, Journal of the Audio Engineering Society, Vol 40, No 7/8, 1992 July/August.

7. Appendix. Analog Level Settings

At the back panel of the ADDA 2408 the input clipping level, and the maximum output level (I/O levels) can be set to different values. It has to be noted that the setting controls are small and delicate and has to be operated with care. Only trained personnel should do this work. Note that the warranty of the unit does not cover if some of these controls have been damaged.



Adjust the line input.

The Input line level can be adjusted between +12 and +30 dBu. In the system menu at the front panel the max line level can be set to 12 to 21 dBu or 21 to 30 dBu.

Use the adjustment screw at the back panel in order to adjust the max input level between +12 and +21 dBu, or +21 and +30 dBu.

Turn the adjustment screw clock wise in order to get a higher analog output level or turn the adjustment screw counter clock wise in order to get a lower analog output level.

Adjust the line output

The analog line output can be adjusted between +12 and +27 dBu.

Use the adjustment screw at the back panel in order to adjust the max output level.

Turn the adjustment screw clock wise in order to get a higher analog output level or turn the adjustment screw counter clock wise in order to get a lower analog output level.

Distortion Effects from Aliasing in Digital Audio

Mikael Vest and Peter Scheelke
Digital Audio Denmark
www.digitalaudio.dk

ABSTRACT

When converting audio between the analog and digital audio domains, the well-known effects from alias signals have to be taken into consideration. Generally, audio converters do not have adequate filtering to eliminate this kind of spectral distortion. During recent works it has been concluded that the sound quality of digital recordings can be drastically improved when applying efficient stop-band filtering.

INTRODUCTION

When converting an analog signal to a digital signal, the analog signal is sampled with a frequency of twice the bandwidth that is required for the digital representation. To get an optimal sampling, the bandwidth of the analog signal must not exceed half the sampling frequency, also referred to as the Nyquist frequency (NF). If frequency contents are present above the NF, a mirror of the frequencies is generated at frequencies symmetrical to the NF and new frequencies are thus generated with no harmonic relationship to the tonal contents of the original signal. This phenomenon is referred to as aliasing distortion. If the distortion is only present at high frequencies, it is not likely that it can be heard, and therefore it will not degrade the signal quality. However, when the digital signal has to be reproduced using D/A converters, amplifiers and indeed analog transducers such as loudspeakers will introduce intermodulation distortion (IMD) to the signal. IMD means that the frequencies of the signal mix together and generate new frequencies, which can be audible if the distortion exceeds certain levels. The problem is now that when new a-tonal frequencies are added to the signal due to aliasing distortion, such signals will mix very badly with the rest of the program and thus be audible.

The important fact of this problem is that distortion happens when the signal has both aliasing distortion and IMD above a certain level. Of course, it depends on the quality of the sampling and the quality of the reproduction equipment, and in particular the loudspeakers. This effect is referred to as Aliasing Intermodulation Distortion (AID).

Over the years discussions and papers on sound quality of digital recorded material have been many. However, at the 106th AES Convention in Munich a new thesis was presented by Mr. Richard Black [1] stating the problems of this combined distortion phenomenon.

This paper has the purpose of explaining the basis of this distortion type, which he had recently identified.

Today high quality audio A/D converters are typically based on the delta-sigma conversion principle with 64/128 times over sampling and internal digital filtering. According to the manufacturers of the converter chipset, this removes the need for an external anti-alias filter. However the filters normally implemented are so called 0.45/0.55 times the sample frequency (FS) filters. This means that the pass band goes to 0.45xFS, and the stop-band starts from 0.55xFS. In Figure 1 the frequency response is shown with 44.1 kHz and 48 kHz sample frequency.

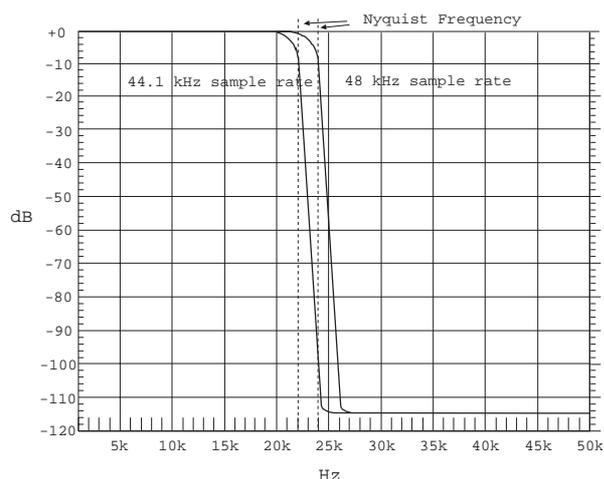


Figure 1, Frequency response of 0.45/0.55xFS filter

In the case of FS = 44,1 kHz, the stop-band will only be effective at 24,26 kHz, and the attenuation at the NF is approx. 8.5 dB. In Figure 2 a close-up of the transition band frequency response is shown.

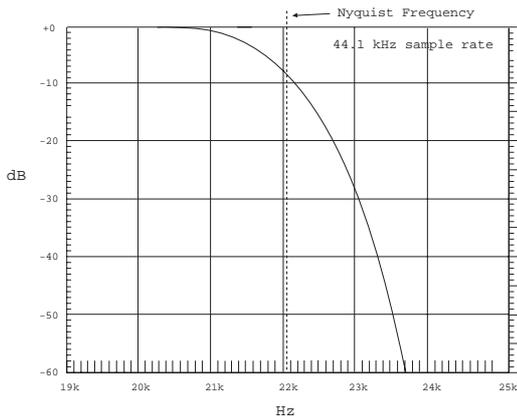


Figure 2, Transition band of 0.45/0.55xFS filter

According to the classic theory on sampling of analog signals, the spectrum of the signal will be mirrored round the NF and repeated for each multiple of the NF, as shown in Figure 3.

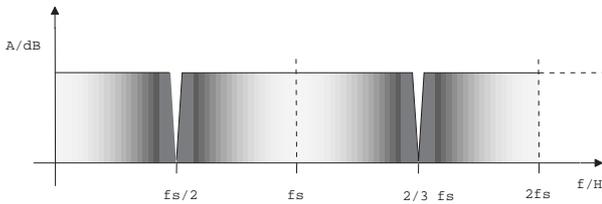


Figure 3, Spectrum of a sampled signal

If signal contents are available in the signal above the NF a mirror of the signal will fold down below the NF as an alias signal, and thus generate aliasing distortion. If a broadband signal is sampled with a 0.45/0.55 anti-alias filter, the spectrum can act as shown in the example in Figure 4.

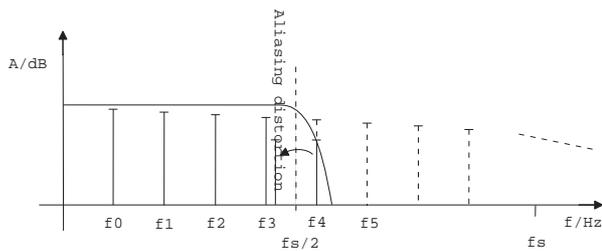


Figure 4, Aliasing distortion example

The example shows a signal with a fundamental frequency of f_0 and a number of harmonics. The 4th harmonic is just above the NF, but still in the transition band of the anti-alias filter of the A/D converter. When the signal is sampled with an A/D converter having an anti-alias filter, with stop-band attenuation at the NF, no aliasing distortion appears.

INTERMODULATION DISTORTION

Intermodulation distortion is a type of distortion where non-linearities in the signal path generate modulation between all the frequency components of the signal. The primary source for IM distortion in the audio signal path is the loudspeaker. This is due to the non-linear behaviour of the speaker cone, which of course depends on the loudspeaker type and quality.

Modulation means that new frequencies f_{im} are generated from all positive combinations of two existing frequencies.

$$f_{im} = | \pm f_1 \pm f_2 |$$

If two frequencies are denominated f_1 and f_2 , where $f_2 > f_1$, and assuming $f_2 < 2f_1$, the modulation frequencies generated are as shown in Table 1.

	f_2	$2f_2$	$3f_2$
f_1	$f_2 - f_1$	$2f_2 - 2f_1$	$3f_2 - f_1$
$2f_1$	$2f_1 - f_2$	$2f_2 - 2f_1$	$3f_2 - 2f_1$
$3f_1$	$3f_1 - f_2$	$3f_2 - 2f_1$	$3f_2 - 3f_1$

Table 1, IM frequencies

IM distortion will appear up to a very high order, but third order IM distortion is the highest order that has any significance in this application. The magnitude of the modulation products is reduced when the modulation order rises. The non-shaded areas in the table are the modulation products with the highest relevance.

IM distortion will generate a-harmonic signals when aliasing distortion is present, since the alias signal will modulate with the harmonics of the original signal. Depending on the frequency contents of the signal spectrum, some modulation frequencies will be more audible than others. If an IM frequency is near the fundamental frequency for the tonal spectrum, the effects will be quite annoying, unless the IM frequency is within the masking area, in which case it will not be heard. An example of IM distortion is shown in Figure 5.

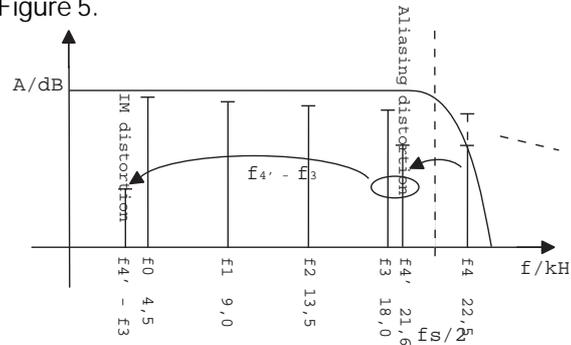


Figure 5, IM distortion example

In this case the 4th overtone of a fundamental tone of 4.5 kHz will generate aliasing distortion at 21.6 kHz. This frequency will modulate with the other tones in the signal spectrum, and first order modulation with the 3rd overtone will then generate a modulation frequency of 3.6 kHz, which is likely to be audible and make interference with the original signal. IM distortion coming from modulation between the harmonics of the signal will not generate a-harmonic distortion signals, since the IM products will have frequencies equal to the harmonic of the signal, and will therefore not be audible.

DVD audio

Recently the new sampling 96 kHz format has been defined. This format is meant to be used in the music production environments, and as the DVD linear audio format. Generally, there is an acceptance that this higher sample-rate gives a better sound quality due to the extended signal bandwidth from 48 kHz to 96 kHz. However, sampling with 96 kHz having 0.45/0.55 filters – which is currently the case for most converters – the aliasing distortion will move up in a frequency area where the IM distortion has very little effect since the harmonic contents of audio signals are insignificant above 40 kHz. An interesting test is to compare the 96 kHz sampled signals with the 44,1 kHz sampled signals having NF stop-band filtering. At 96 kHz sampling the ultra-sonic harmonics will be present in the signal giving a better quality than when sampled with 44.1/48 kHz even with a NF stop-band filter applied. However the primary issue when considering the different sample-rates for high quality sampling is to avoid AID, more than having a higher sample-rate.

SUMMARY

When converting audio from the analog to the digital domain, care has to be taken that stop-band filtering is applied in order to avoid aliasing distortion in the digital audio signal. Once the signal is converted to digital, the aliasing distortion products can not be removed with-out reducing the bandwidth accordingly. The aliasing distortion will, when the signal is reproduced on a set of loudspeakers having some IM distortion, generate AID. Of course good loudspeakers have lower distortion, but most hi-fi speakers will have IM distortion thus giving audible AID.

The only way to eliminate the problem with AID is to apply stop-band filtering on the A/D conversion. If this is not done AID will cause different problems depending on the application for the sampled digital audio signal.

When using 44.1/32 kHz sample-rates, the NF is close to/within the audible frequency band. This is a problem for the Compact Disc (CD). Since CDs are mastered in 44.1 kHz AID is a problem, if the A/D converters are used without proper filters. If recording is done with 48 kHz sampling, the Aliasing distortion will be at frequencies above 20 kHz. This will be a problem when working with the audio signals in the sound studio, where the monitored signals will have AID.

If digital sample-rate conversion is used to generate the 44.1 kHz master version from 48 kHz source material, AID will not occur since the alias products are filtered by the sample rate converter alias filter assuming of course that a good quality sample-rate converter is used.

Not many A/D converters are available with NF stop-band filtering. As mentioned in the abstract, the majority of converters have 0.45/0.55 anti-alias filters, which will result in AID. One converter chip is however available today with NF stop-band filtering, and that is the CS5397 from Crystal Semiconductors.

This topic of AID and the influence of sampling bandwidth on the sound quality is something that needs to be investigated further, since to our knowledge documented test results are not available. This is however a subject in which Digital Audio Denmark will conduct more research in the future.

Digital Audio Denmark A/D converters

The ADDA 2402 A/D, D/A and D/D converter from Digital Audio Denmark has implemented NF stop-band filtering for eliminating the risk of AID. This means that digital recordings can be made without aliasing distortion. However, a trade-off has to be made concerning the bandwidth of the sampled signal. When sampling with 44.1 kHz the transition band starts at 18.1 kHz, and the attenuation at 20 kHz is 12 dB. This means that the signal 3 dB bandwidth is reduced to 19.3 kHz.

References

- [1] R. Blake: "Anti-alias filters: the invisible distortion mechanism in digital audio". Preprint 4966, 106th AES Convention, Munich 1999.