Digital Audio Processor for two-, three, or four-way Speaker Systems

Digital Audio Processor AC-DAP

- **2 x 7 VAC or 2 x 10 VDC Supply Voltage**
- **Extremely Low Distortion and Noise:**
  - 100 dB THDN
- **Two symmetrical Audio Inputs with Selectable Sensitivity (Studio & Consumer)**
- **S/PDIF optical and coaxial input (option)**
- **Input for digital MEMS microphone (V3)**
- **I²S connector**
- **Eight independent single-ended outputs or**
- **Four symmetrical (balanced) outputs**
- **On-board Muting/Standby-Sequencer with Control Inputs and Outputs and Audio Signal Presence Detection (ASD)**
- **Programmable via Analog Devices SigmaStudio® Graphical User Interface**
- **No DSP Programming Knowledge Required**
- **On-board self-boot function or A/B Testing with SigmaStudio®**
- **On-board 8-Bit AVM Microprocessor for general control purpose and display support via I²C**
- **Various user configurable elements (LEDs, Potentiometers and DIP-Switches)**
- **RoHS compliant, IPC-A-600 Class 2 and IPC-A-610 Class 2**

Figure 1: Digital Audio Processor AC-DAP for two-, three-, or four-way speaker systems with or without Sub-Woofer

Description / Novel Features

The purpose of the compact, high-end, audio processor AC-DAP is to generate the appropriate driving signals for a Doppler compensated two-way, three-way, or four-way speaker system. The AC-DAP shows an excellent linearity and extremely low distortion and noise (better -100 dB THDN), combined with a perfect flat audio frequency response.

The block diagram in Figure 2 visualizes the main building blocks.

The AC-DAP receives the audio input as single ended or symmetric signal. In order to maximize the dynamic range of the ADC the input sensitivity of each input can be selected for studio or consumer audio levels. A band pass filter in the input circuitry determines the noise bandwidth between 2 Hz and 120 kHz and rejects out of band spurious signals. Differential amplifiers with high common mode rejection ratio deliver the symmetrical audio signals to differential input 24-bit Analog Digital Converters. The anti-aliasing filtering is mainly done with digital filters at a high oversampling rate and an upper corner frequency close to half of the sampling rate.

For the description of the Core Digital Audio Processor see below.

Eight 24-bit Digital to Analog Converters feed their signals to eight output filters and driver stages with a bandwidth of 2 Hz to 120 kHz in order to avoid linear distortions.

Eight single-ended, or four symmetrical outputs can be configured via software. These driver stages are perfectly suited to feed the AudioChiemgau Power Modules (AC-PAZ75 adapted impedance power module and/or AC-

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PAR75 power module with on-board motion feedback control loop).

The internal outputs carry also MUTE and STBY signals for the AC-PAZ75 or AC-PAR75 amplifiers.

An optical TosLink (S/PDIF) input is provided which interfaces trough a synchronization circuitry directly with the digital processor core.

An integrated digital controller is responsible for ensuring an un-audible ON/OFF switching sequence controlled by the on-board audio signal presence detector, or by external signals. Temperature monitoring of the module is implemented as well as a power saving stand by function.

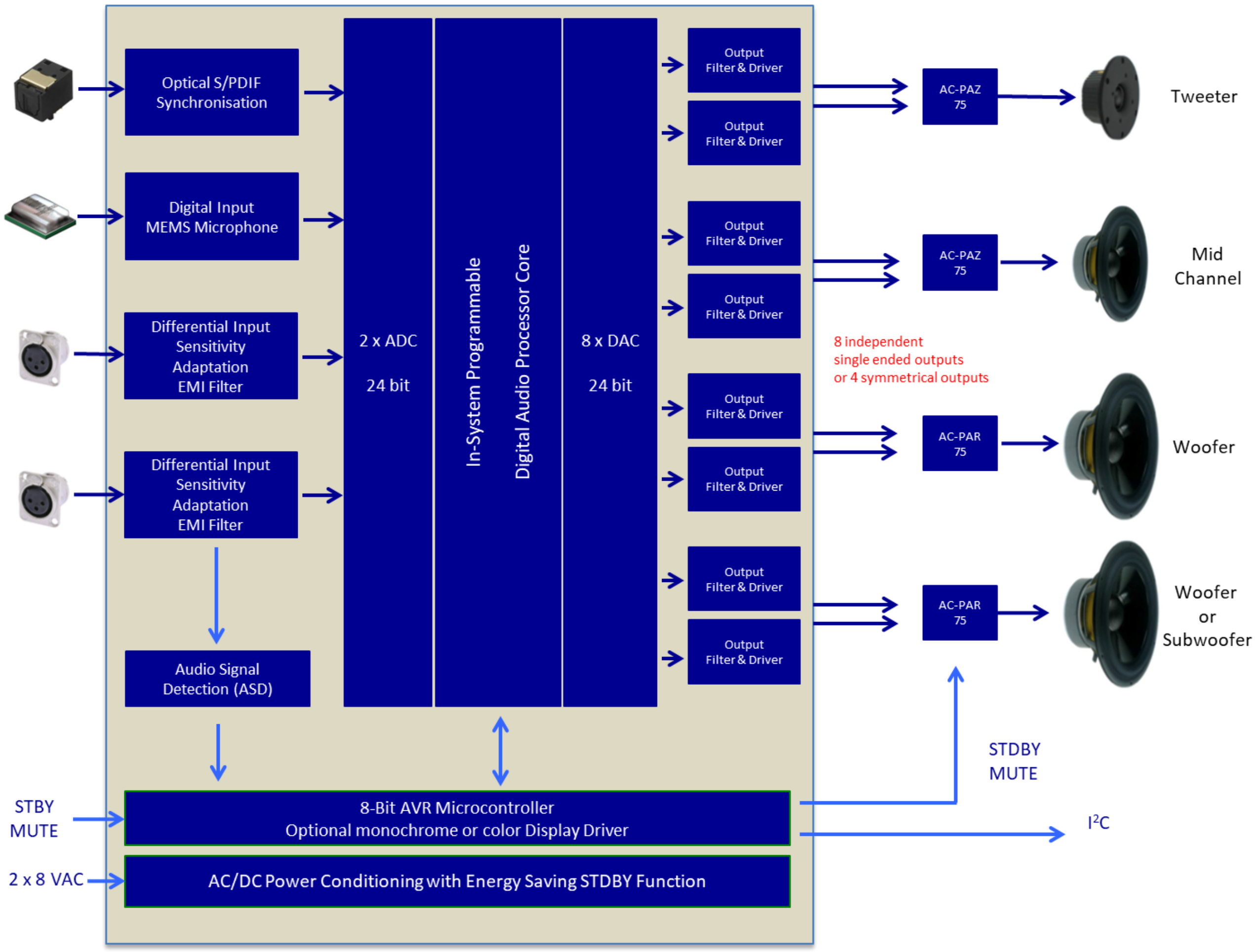

Figure 2: Block Diagram of the Digital Audio Processor AC-DAP 1)

Figure 3 shows examples of functions, which can be easily implemented with the AC-DAP. The shown configuration would be used for a 4-way audio system or for 3-way audio system with sub-woofer.

[1]) MEMS-Microphone implemented with Version 3

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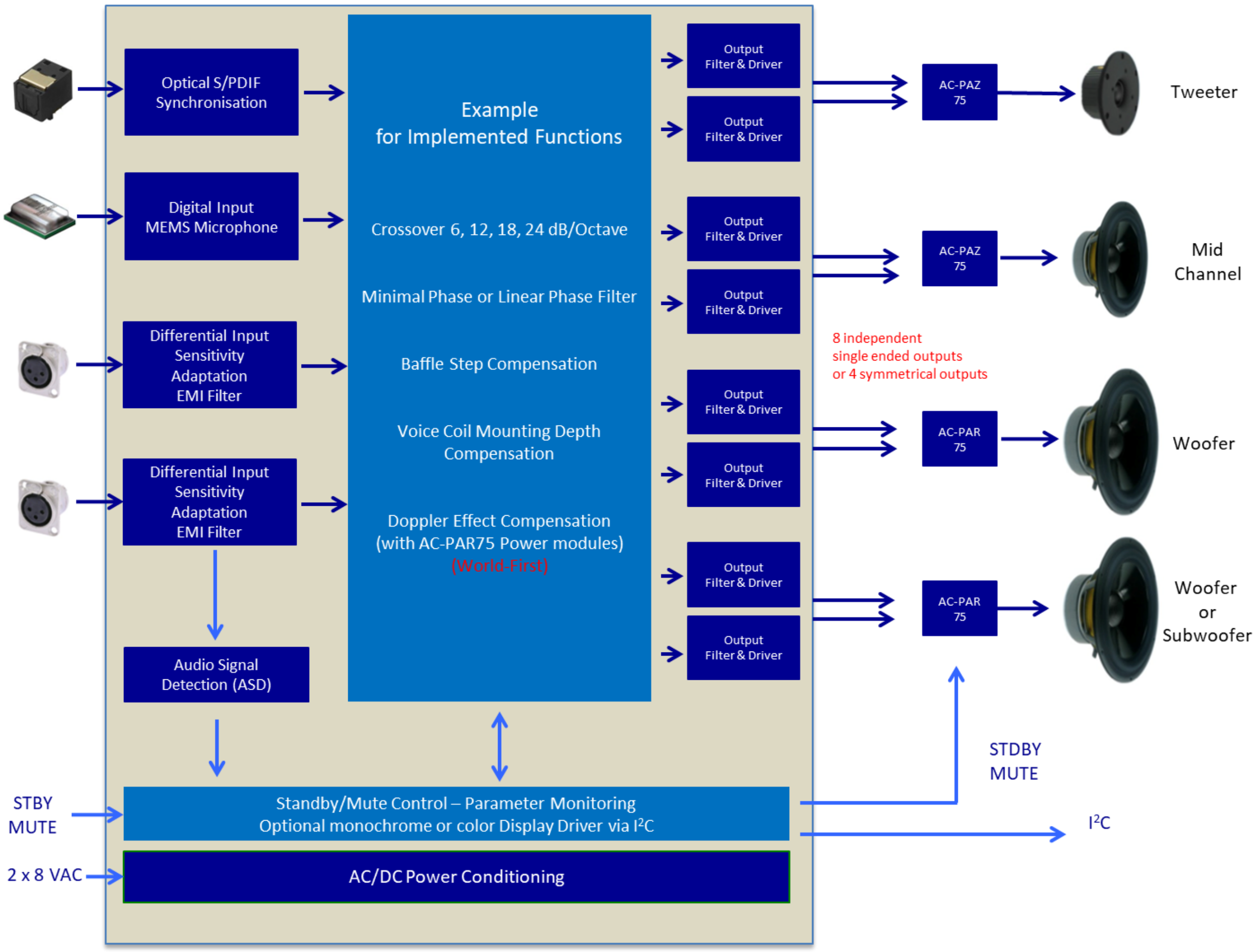

Figure 3: Examples of implementable functions

Figure 4 shows the easy programming of the AC-DAP through the Analog Devices SigmaStudio® graphical user interface. The example shows the implementation of a 3-way speaker system with symmetrical (balanced) outputs for the power modules.

The user is free to route signals arbitrarily between all three inputs and the eight outputs.

SigmaStudio® offers a wealth of useful function blocks.

It is easy to implement even sophisticated processing steps, as the compensation of the Doppler effect, which any moving speaker membrane generates - a world first innovation from AudioChiemgau.

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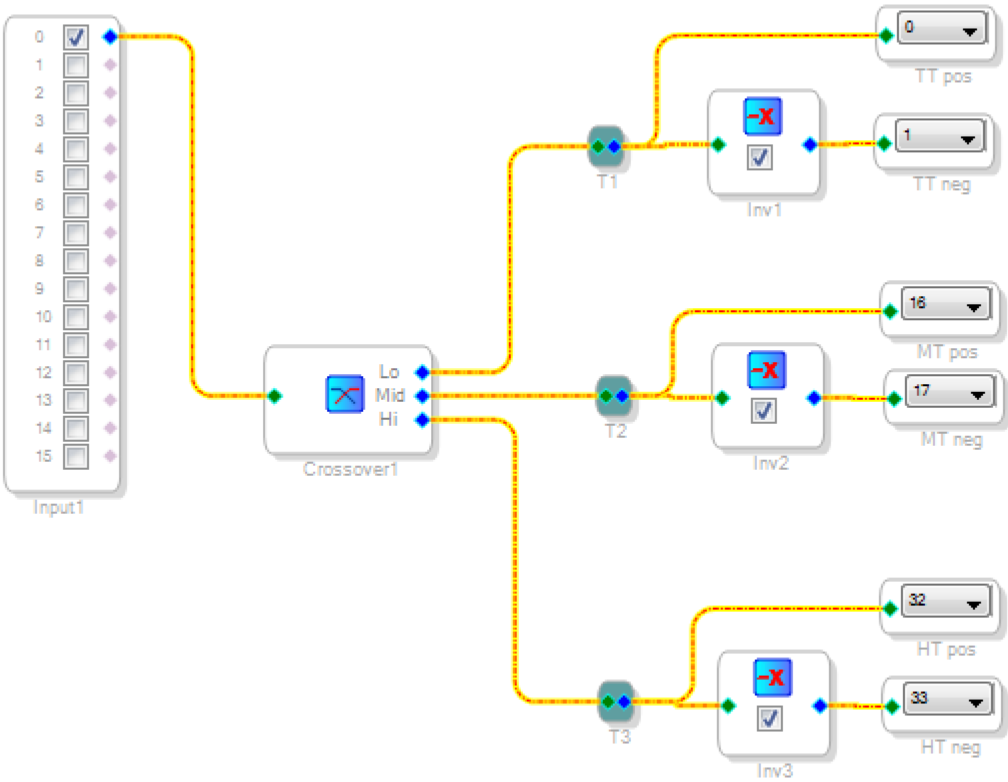


Figure 4: Example of programming the AC-DAP through the SigmaStudio® Graphical User Interface

Figure 5 shows the corresponding frequency response. The sum of the filter outputs is for all frequencies constant (constant voltage filter).

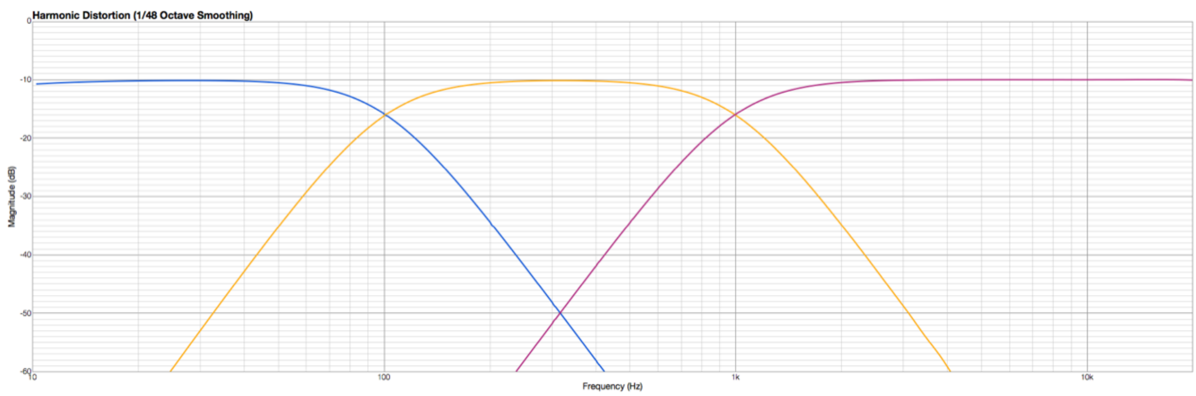


Figure 5: Frequency response of the example filter implementation

Detailed description and Application Information

The AC-DAP is operated typically with two equal 8 VAC voltages. Two separate windings of an AC transformer, or a middle tapped secondary winding can be used.

The analog audio input signal is feed to a differential input amplifier stage with high common mode signal rejection in order to avoid ground loops. A connection between the driving source ground and the AC-DAP ground (e.g. through pin 1 of the XLR connector) is therefore necessary in order to establish a common GND reference.

In order to simplify building a complete loudspeaker box with e.g. one tweeter and one or two mid-range speakers and one or several woofers, a dedicated star-grounding scheme is implemented for an optimum result. The AC-DAP board offers a star ground, which is realized by the input and output connectors in combination with the dedicated 4-port connector X5. For detailed information see the application note AC-AN-001 for recommended Grounding implementation.

The AC-DAP is optimized for the combination with the amplifier modules AC-PAZ75 for the tweeter, one or several AC-PAR75/AC-PAZ75 for the mid-range speakers and one, or several AC-PAR75 the woofer(s). For these differential input power modules, the 8 output channels of the AC-DAP should be configured as symmetrical output pairs, as shown in the example in Figure 4.

A digital optical S/PDIF interface is provided via a TosLink connector. Version with coaxial (SMA) connector is available. A dedicated synchronization unit feeds the digital signals to the digital processor core for further processing.

One connector for I^2S interface is available with PCB-Version 3.3 or higher.

On-Board controller

The additional on-board microprocessor serves inter alia as sequencer for the handling of the STBY and MUTE function as well as for the signaling of operation mode and status. The AC-DAP features an auto on function through the detection of the input signal and an auto off function after 10 Minutes without input signal. This function is dubbed Audio Signal Detection (ASD).

As default the ASD function is enabled and may be disabled with an installed jumper X101.

Independent from the MUTE-signal during power-on, the system will be activated and connected equipment will be switched on in a defined sequence.

With active ASD the system will stay in ON as long as an analog audio signal above the S/W implemented threshold will be detected. In case the signal is eight minutes under the threshold the controller changes the status to STBY and after additional two minutes the system switches to OFF.

An active TosLink-Input (S/PDIF) will also be detected as audio signal. In case such a digital signal is present on the optical receiver and the systems locks to this signal, the system selects this input instead the analog input [2]).

Furthermore, the system could be controlled by the MUTE-Signal independent from the audio input signal. In case the MUTE signal is pulled down for more than 10 minutes, the controller will switch the system to OFF.

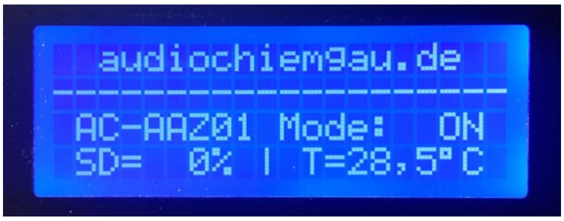
All implemented durations and thresholds for ON and OFF could be changed easily (values stored in the EEPROM of the on-board controller).

Mute (MUTE), standby (STBY) and OFF will be indicated by different signals of the connected two-color LED (see table on page 15).

[2]) Implementation in DSP firmware necessary to support this function (see example in Figure 22)

Support for optional Displays

An I^2C on-board interface allows connecting various display types showing the operating mode, the temperature, the audio level and a customer defined logo. See the following example for a possible implementation and display content of such a display.



This example shows the simplest solution with a four-line alphanumeric display.

Supported are also graphical displays with monochrome as well as color displays including touch function to control the different modes of the Audio Processor. All displays are available with different resolutions and different sizes.

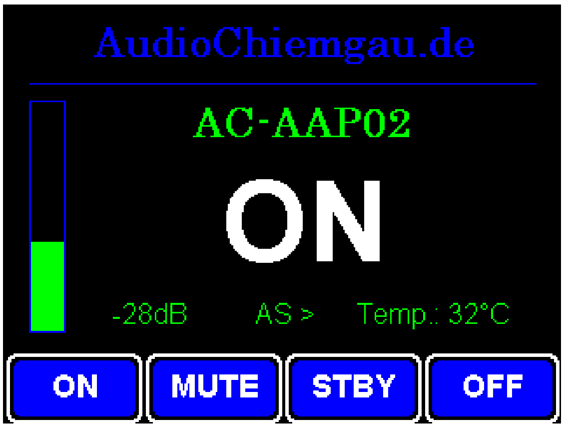


Figure 6: 3.2" Color Display with four touch buttons

Please contact AudioChiemgau for available display types, display colors or content changes.

Temperature Monitoring

An over temperature monitoring of the module is implemented. In that case the module as well as external connected equipment will be switched OFF via the remote-control lines (see below). After cooling down, the system will be reactivated. Over temp condition will be indicated by the status (LED2) or the optional display.

Remote ON/OFF control outputs

Two remote control lines (ON/OFF) for external equipment are also available. The outputs are under control of the on-board processor, are short circuit protected and designed to drive 12 V relays directly with up to 40 mA load current (35 mA constant current).

In OFF-mode the internal supply voltage for the module itself is switched off in order to reduce power consumption to an absolute minimum.

The background illumination of an optionally connected display will also be switched off 15 seconds after the system switches to OFF.

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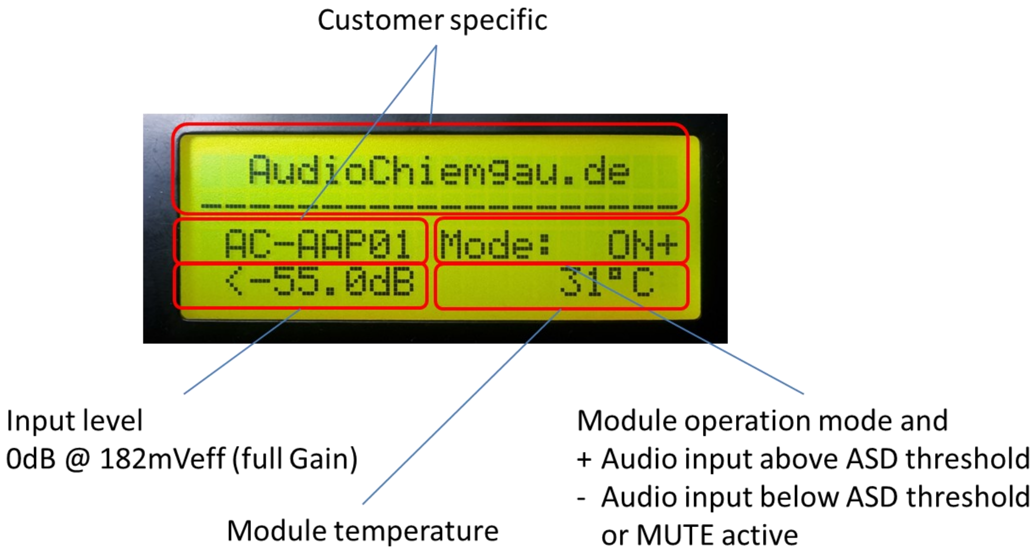

Figure 7: Example for Tiny Display

<u>Customer / User Adjustments by hardware:</u>

1. Jumpers for sensitivity selection of the input differential amplifiers
2. Up to three independent trimmers located on-board.

   Up to two (four in PCB-Version 4) connectable to external potentiometers as input to the DSP. These Signals may be used for any user defined DSP-Function
3. Four DIP-Switches available for selectable user defined DSP-Functions

   (one reserved for the activation of the Self-boot option (PCB version 1.0)

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Absolute Maximum Ratings (T_{amb} = 25°C; unless otherwise specified)

Symbol	Parameter	Value	Unit
Vs	AC supply voltage (two symmetrical transformer windings)	12	Vrms
T_{op}	Operating ambient temperature range	0 to +50	°C
V_{OD}	Open drain voltage in high state (MUTE/STBY)	35	V
T_{stg}, T_j	Storage temperature	+ 80	°C

Stresses beyond those listed under "Absolute Maximum Ratings" may cause permanent damage to the module.

Electrical Characteristic (T_{amb} = 25°C; f = 1kHz; unless otherwise specified)

Symbol	Parameter	Min	Typ	Max	Unit
Vs	AC supply voltage range Two identical transformer windings, or mid tapped secondary winding	6	8	10	Vrms
Is_{OFF}	Supply current in OFF (idle)		20	35	mA
$Is_{operating}$	Supply current in nominal operation [3])	200	300	500	mA
P_S	Required AC power per winding			10	VA
Analog Audio Input Channel(s) [4])					
R_{id}	Differential input resistance (AC)		100		kΩ
R_{i0}	Input resistance to GND (AC)		50		kΩ
V_{CM}	Input common mode range		±4		V
V_{IS}	Input voltage range, selectable by jumpers (commercial and studio level)		1.20 4.00	1.27 4.06	V_P
V_{ASD}	ASD Sensitivity/Threshold of V_{IS} (adjustable by firmware)		0,1		mV
Standby & MUTE Function (open Drain driver with pull-up)					
I_{OL}	Low-level sink current capability			20	mA
V_{OH}	Output voltage in high-state [5])	4,3	4,7	13	V

[3]) Without external loads on the remote-control driver(s) and without display. Version with linear supply regulation. Version with low-power DC/DC converters are available. Max value depends on DSP activity.

[4]) Second input channel optional

[5]) See interface description for more details

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Symbol	Parameter	Min	Typ	Max	Unit
External Error and Status Indicator					
I_{LED1}	LED operating current [6])		1,5		mA
Remote Control Output Driver (up to PCB-Version 3.1)					
I_{NOM}	Nominal driver capability		40 [7])		mA
V_{out}	Output Voltage Driver active With typical supply voltage Vs	14	16	18	V
I_{max}	Max output current before switch off		65		mA
I_0	Short circuit output current (fold back)	15			mA
V_{OFF}	Output Voltage Driver OFF	-0,5	-0,2	+0,5	V
Remote Control Output Driver (PCB-Version 3.2 and later)					
I_{NOM}	Nominal driver capability (constant current)		35 [8])		mA
V_{out}	Output Voltage Driver capability With typical supply voltage Vs			18	V
Over Temperature Detection and Turn ON/OFF					
T_{OFF}	Switch OFF temperature [9])	+75	+80	+85	°C
T_{ONHY}	Switch ON Hysteresis		4		K

[6]) Adjustable by R101 and R106

[7]) Also deliverable with higher drive capability

[8]) Also deliverable with higher drive capability. Value optimized for 12 V relay with coil resistance of 340 Ω

[9]) Changed from +70°C to +80°C with F/W Version 5D04

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PCB Layout and Mechanical Support

The complete circuit is realized on a multi-layer PCB with the dimensions of 150 mm x 80 mm. The PCB provides five mounting holes as shown in the figure below (one for grounding connection).

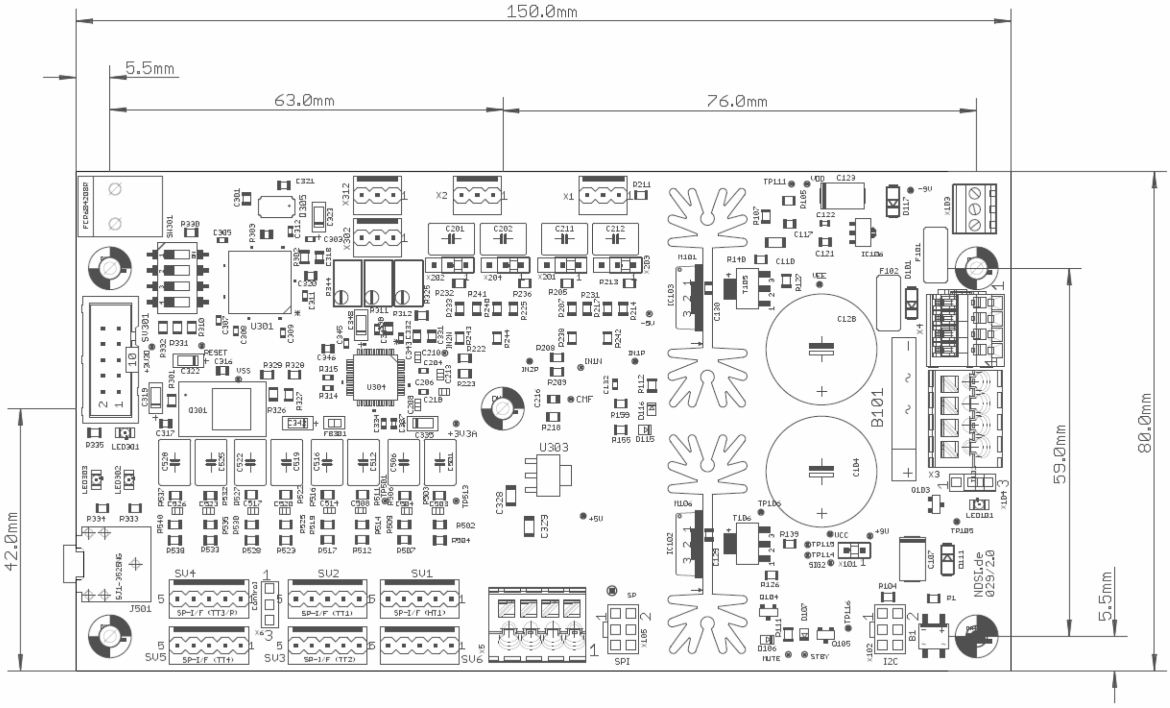

Figure 8: Layout of the PCB, top view and populated as AC-DAP (V2)

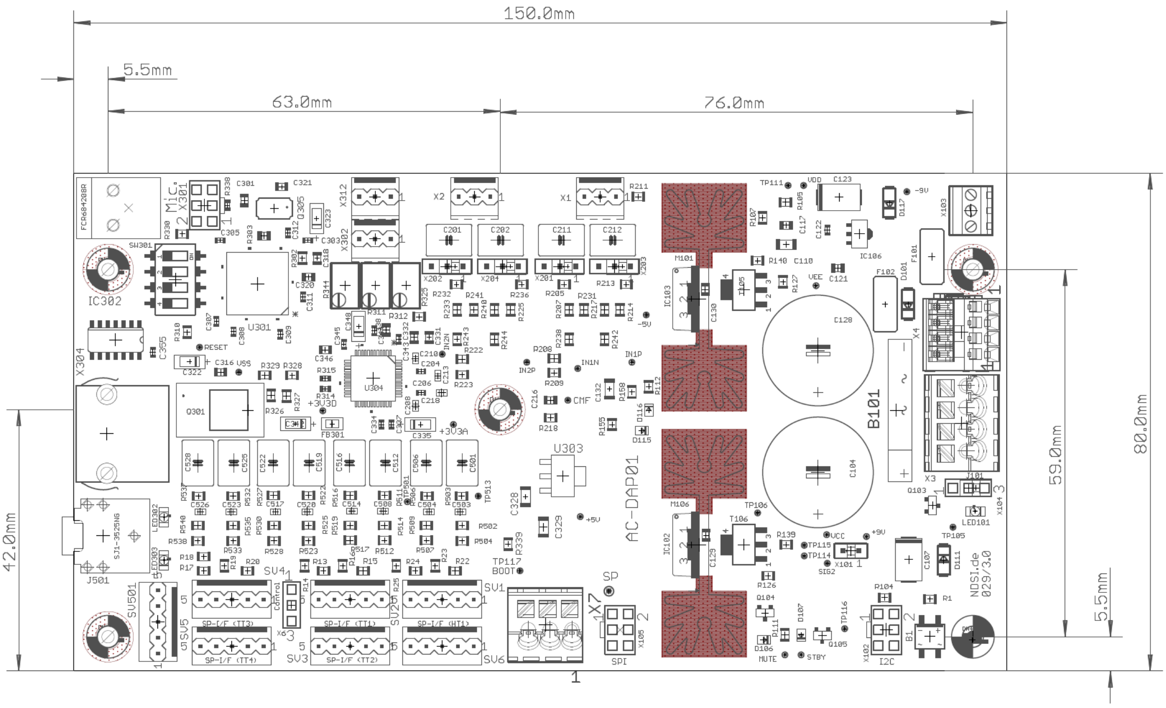

Figure 9: Layout of the PCB, top view and populated as AC-DAP (V3)

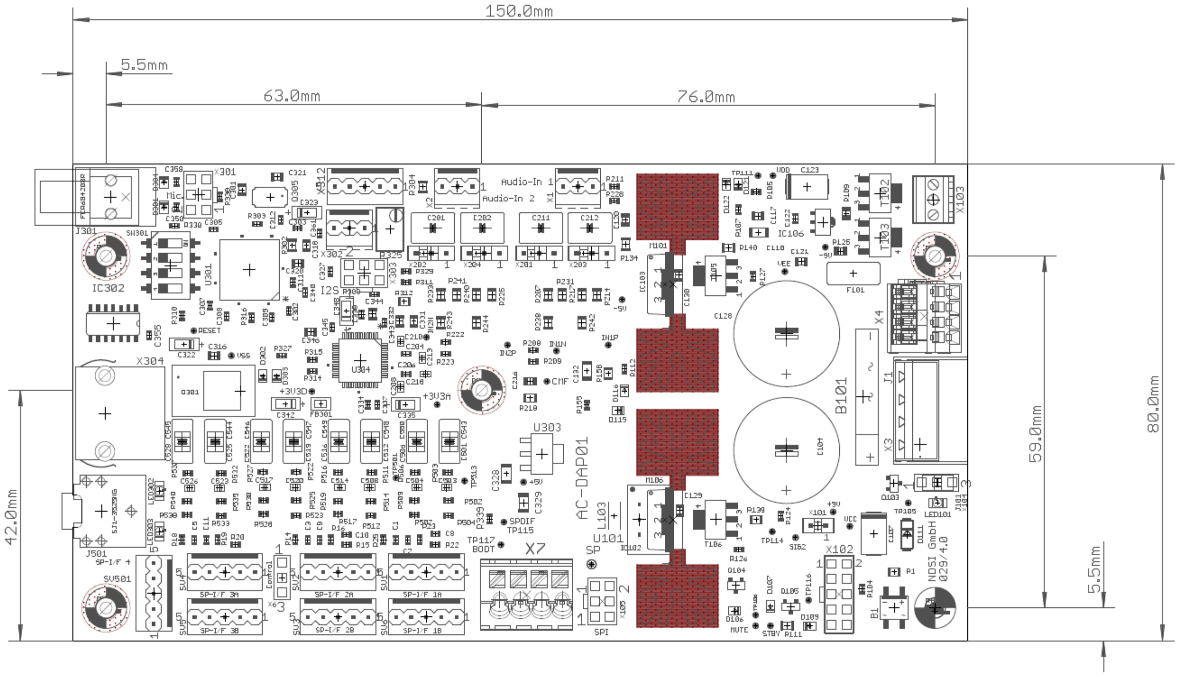

Figure 10: Layout of the PCB, top view and populated as AC-DAP (V4)

Figure 11 shows an optional flange for vertical mounting of the module.

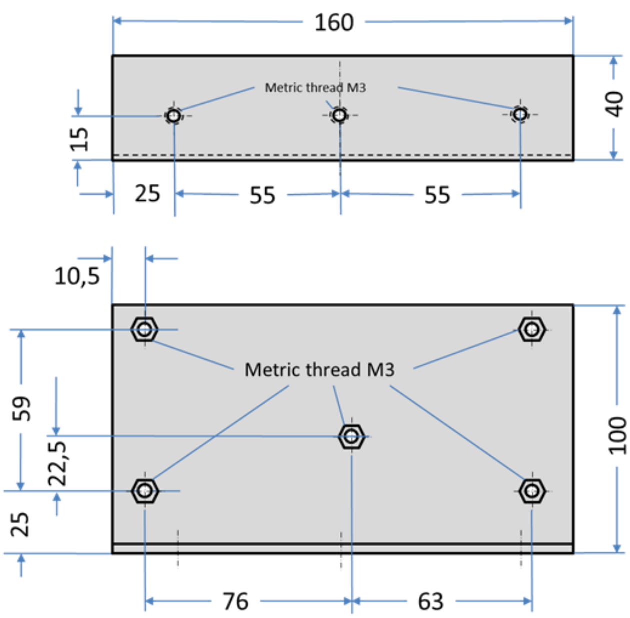

Figure 11: Flange for vertical mounting of the module

The mounting flange provides on the small side (upper sketch) three metrical threads M3.

Both sides have flat surfaces; there are no protruding elements.

The flange is not necessary for any cooling purpose.

The AC-DAP will be delivered mounted on such a flange.

Table 1: Mechanical Characteristic (AC-DAP PCB without mounting flange)

Symbol	Parameter	Min	Typ	Max	Unit
X	Module Dimension X	149	150	151	mm
Y	Module Dimension Y	79	80	81	mm
H	Module Height H		45		mm
W	Module Weight (including mounting flange)		327		g
T	Mounting Flange Material Thickness		3		mm
M	Flange Material	AlMgSi0,5/EN AW-6060			

WARNINGS

- Any external power supply used with this module shall comply with relevant regulations and standards applicable in the country of intended use.
- This product should be operated in a well-ventilated environment, and if used inside a case, the case should not be covered or should provide enough room for proper ventilation/cooling. Alternative the module could be mounted on a stable, flat, conductive surface for additional cooling purpose.
- The connection of incompatible devices to the module may affect compliance, result in damage to the unit, and invalidate the warranty.
- All peripherals used with this product should comply with relevant standards for the country of use and be marked accordingly to ensure that safety and performance requirements are met. These articles include but are not limited to power supply, amplifiers, programming adapters and other external elements.
- The cables and connectors of all peripherals used with this product must have adequate insulation so that relevant safety requirements are met.

SAFETY INSTRUCTIONS

To avoid malfunction or damage to this product, please observe the following:

- Handle only in an environment which is protected against Electro Static Discharge (ESD) to avoid possible damage of the sensitive parts.
- Do not expose to water or moisture, or place on a conductive surface whilst in operation.
- Do not expose to heat from any source; this module is designed for reliable operation at normal ambient temperatures.
- Take care whilst handling to avoid mechanical or electrical damage to the printed circuit board and connectors.
- Whilst it is powered, avoid handling the printed circuit board, or only handle it by the mounting flange to minimize the risk of electrostatic discharge damage.

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Electrical Interfaces and Used Connector Types

Table 2: Connector Types, Jumpers and Interface Description

Connector	Parameter/Signal	Typ	Wire Size
X1 X2	Audio input interface for left respectively right channel	Molex Pin - KK - 1x3-pol - 22-27-2031 (straight)	Shielded twisted pair recommended
X3	AC Power Supply	WAGO 250-204 (PCB-Ver. up to 3.x)	0,2 – 1,5mm² AWG 24-15
X3	AC Power Supply	AMPHENOL ANYTEK 20020107-D041A01LF (PCB-Ver. 4.x)	0,2 – 1,5mm² AWG 24-15
X4	Remote control output	WAGO 233-504	0,2 – 0,5mm² AWG 24-20
X5	GND connection (Star Point)	WAGO 250-204	0,2 – 1,5mm² AWG 24-15
X201 X202 X203 X204	Selection of audio input sensitivity	Pin Header 1x3 RM 2,54mm	Jumper only
X6	Connector for external control (MUTE/STBY)	Pin Header 1x3 RM 2,54mm	-
X101	Enable/Disable ASD-Function Installed: ASD disabled	Pin Header 1x2 RM 2,54mm	Jumper only
X102	I²C-Interface (reserved)	Pin Header 2x3 (2x5) RM 2,54mm	-
X104	Connector for external two-color LED (option)	Pin Header 1x3 RM 2,54mm	-
X105	SPI-Interface (reserved)	Pin Header 2x3 RM 2,54mm	-
X301	Interface Connector for digital MEMS microphone	Pin Header 2x3 RM 2,54mm	-
X312	Interface for external potentiometer (PCB Ver. up to 3)	Molex Pin - KK - 1x3-pol - 22-27-2031 (straight)	Shielded twisted pair recommended
X312	Interface for external potentiometer (PCB Ver. 4)	Molex Pin - KK - 1x5-pol - 22-27-2051 (straight)	
X302	Interface for external potentiometer	Molex Pin - KK - 1x3-pol - 22-27-2031 (straight)	
X303	I²S-Interface Connector	Pin Header 2x3 RM 2,54mm	-
SV1 – SV6 SV501	Audio out, with MUTE and Standby to Power Amplifiers	Molex Pin - KK - 1x5-pol - 22-27-2051 (straight)	See Table 4 for deliverable hardware configuration
J501	Audio signal Monitor-Out	Jack plug 3,5mm CUI SJ1-3525NG	-
LW301	S/PDIF digital audio input	S/PDIF / TOSLINK optical cable	-

Connector	Parameter/Signal	Typ	Wire Size
SV301	DSP Control Port Header [10])	2x5 pin four-wall header (PCB-Version 2)	Connection to EVAL-ADUSB2EBZ
X304	DSP Control Port Header	RJ45 (PCB-Version 3 and later)	See Table 20

Table 3: Interface: AC/DC Power Supply and Connector Pinout (X3)

X3 Pin	Parameter/Signal	Remark
L1	AC Input A (or positive DC supply)	L2 and L3 could be connected to use transformer with center tap For proper power drop detection take care of polarity in case of DC supply.
L2	Return A internally connected to L3	
L3	Return B internally connected to L2	
L4	AC Input B (or negative DC supply)	

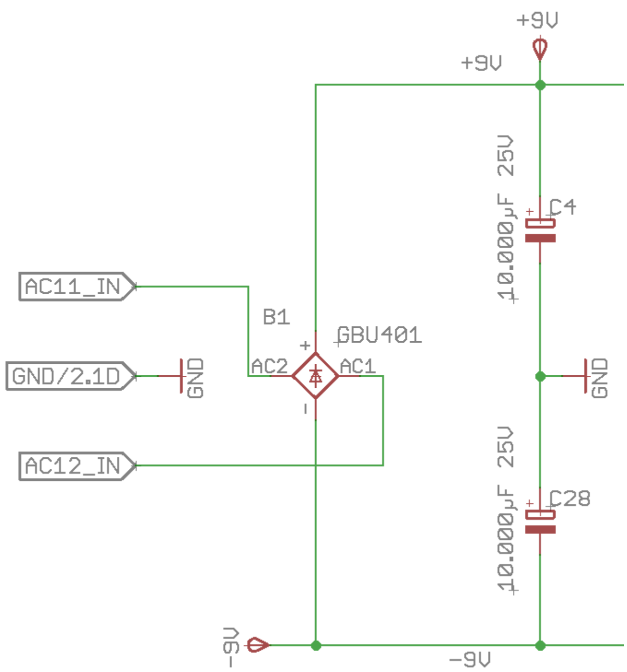

Figure 12: AC Input Circuit Diagram

Remark: Recommended connector for PCB-Version 4.x and later:
Multicomp EBK-04-B (straight) or Multicomp MC000114 (bent).

One of the recommended connectors will be delivered together with the module.

[10]) If not connected DSP will change to self-boot mode (PCB Version 2.0 and later)

Interface: Optional External Status Indicator LED Connector Pinout (X104)

The module offers a status indication output for an external LED (e.g. white LED) and a second LED interface for fault condition indication (e.g. red LED). If not used, short the connector of the white in order to activate the on-board LED, or let the connector open if no indication is required. Both LEDs could be realized with a two-color LED with common anode (e.g. OptoSupply, Part Number OSRMMA7K91B; one LED is will be delivered together with the module).

X104 Pin	Parameter/Signal	Remark
X104-1	Cathode of indicator LED2 (red)	If no second (red) LED is implemented the LED1 will flash in case of failure condition
X104-2	Common contact (anode) for both LEDs	Open: on-board indicator disabled Closed: on-board indicator active LED: Connect to additional LED for external indication
X104-3	Cathode of indicator LED1 (white)	

The AC-DAP provides a status indication by the internal LED101 and an optional external LED supported by a second or two-color LED. Both will be used for signaling of the different states of the audio processor and the connected equipment. In nominal operation the white LED is permanently on and they will flash for signaling of certain nominal operating states or during failure condition:

Flashing	Status	Remark
1 (white)	System in MUTE	System in nominal operation (no failure condition)
2 (white)	System in Standby	
3 (white)	System waiting for DSP Boot	System switched ON but not ready – waiting for (self-)boot of DSP
4 (red) [11])	Over Temperature (>80°C)	Over temperature detected - module switched to OFF for its own protection
2 Sec. (red)	Analog input signal above acceptable limits (clipping of input ADC or maximum coil travel reached)	The input signal is above the acceptable ADC conversion level (full scale). (These functions must be implemented and supported by DSP firmware)
5 (red)	DSP Supply Voltage(s) fails	DSP Supply Voltage(s) out of specified range PCB version 1.0 and 2.0: +3V3 only PCB version 3.0 and later: +3V3 and +1V2

OFF: The LED101 (white) will "breathe" every 10 seconds to indicate a powered system in idle mode.

During system power down (switch off) the LED flushes one time in red for 200ms showing the proper detection of power loss.

[11]) In case no red LED is connected the white LED will flash four times

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Interface: Audio Out and Control Connector Pinout (SV1, SV2, SV3, AV4, SV5, SV6 and J501)

The AC-DAP provides up to seven interface connectors for connecting of independent amplifiers like the AC-PAZ75 or AC-PAR75.

Table 4: Deliverable connector and output driver configurations for SV1 to SV6 and SV501

Option	Configuration	Connector Type
-1	4 symmetrical output ports	MOLEX 22272051 (straight)
-2	3x2 symmetrical output ports + 1x1 symmetrical output port	
-3	6 unsymmetrical output ports	

Remark: An additional 3.5 mm jack-plug (J501) carries two output signals in parallel of SV501 (e.g. for headphones).

MOLEX 22-01-2057 with Inserts 2759 or 4809 are recommended for SV1 to SV6 and SV501.

Table 5: The interfaces are connected in parallel for MUTE and STBY signals.

SV1-6 Pin	Parameter/Signal	Remark
1	MUTE Output	Could be used to control external power amplifiers like AC-PAZ75 and AC-PAR75 (see chapter below for more details)
2	STBY Output	
3	GND	Directly connected to the Star Point of the Module
4	NF (Audio) Output negative	
5	NF (Audio) Output positive	

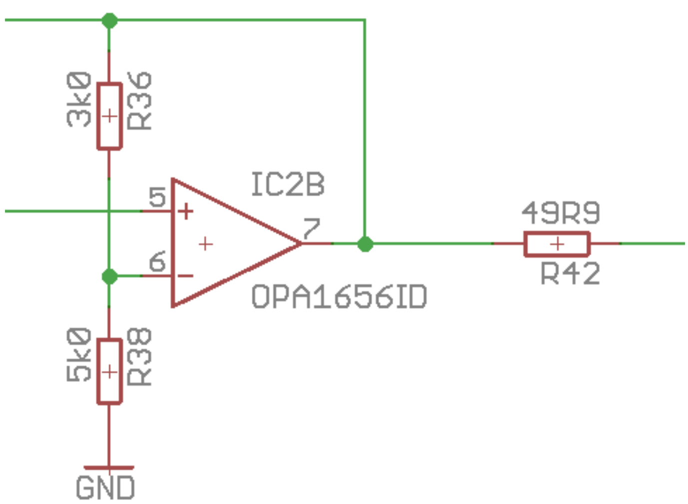


Figure 13: General Audio Output Interface

Standby (STBY) and Muting (MUTE)

The module offers two independent input/output lines for standby (STBY) and muting (MUTE). Both outputs are realized as Open Drain outputs with implemented pull-up resistors and are active low. They can serve as outputs in order to synchronize external power amplifiers like AC-PAZ75 and/or AC-PAR75. It is also possible to use both lines as input signals using open collector drivers. This information is available on a DSP GPIO for dedicated handling within the DSP firmware (e.g. audio signal turn off).

The circuit dedicated to the switching on and off of the amplifier has been carefully optimized to avoid any kind of uncontrolled audible transient at the output during settling of the internal control loops, especially for the amplifiers AC-PAZ75 and AC-PAR75.

If not used, both control input/outputs may be left open. For a controlled on and off sequence without any noise the correct implementation is recommended.

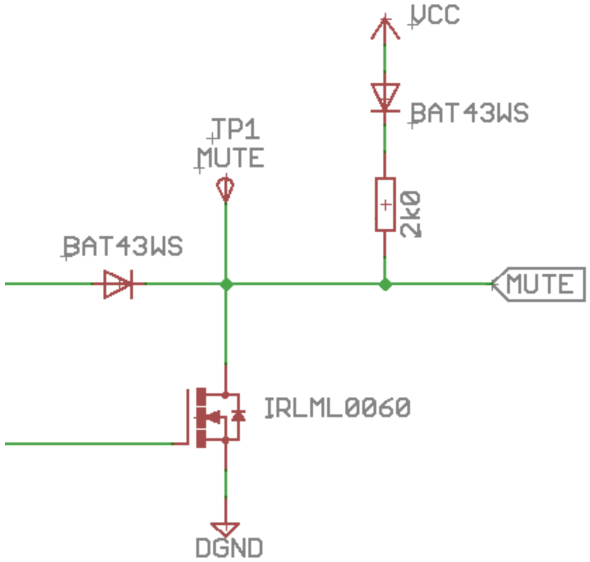


Figure 14: MUTE Input/Output Interface (Standby identical if populated)

Normally the AC-DAP delivers 5 V in high-state but external pull-up resistors can be used to handle external receivers with higher input voltages.

System with MUTE-Switch or Audio Signal Detection (ASD):

Optionally the system may be controlled by an external (manual) MUTE and/or STBY switch, which may be connected to X6.

After power on the equipment will turn to ON immediately and the Audio Signal Detection (ASD) will be enabled. This will initiate a sequence to turn on the whole equipment including external equipment under control of the remote-control output(s) on X4. Is no audio signal detected above the threshold or the external MUTE is active for 8 Minutes the system will change to STBY and after further two minutes the on-board controller of the AC-DAP will turn OFF the system.

Table 6: Interface: External MUTE/STBY (X6)

X6 Pin	Parameter/Signal	Remark
1	MUTE	Connect MUTE (Pin1) and/or STBY (Pin3) to GND to activate the function
2	GND	
3	STBY	Could be used to connect external switch(es)

Table 7: Interface: Remote Control Output (X4)

X4 Pin	Parameter/Signal	Remark
1	Remote Control Out Channel 1 (N)	E.G., to be used for power switching of the power amplifier(s) transformer
2	Remote Control Out Channel 1 (P)	
3	Remote Control Out Channel 2 (P)	
4	Remote Control Out Channel 2 (N)	

The module offers two drivers for control (switch ON/OFF) of internal or external equipment. The output is under control of the on-board processor and its firmware. The output is capable to drive relays directly and is short circuit proof with fold-back characteristic.

With PCB-Version 3.2 changed to constant current output.

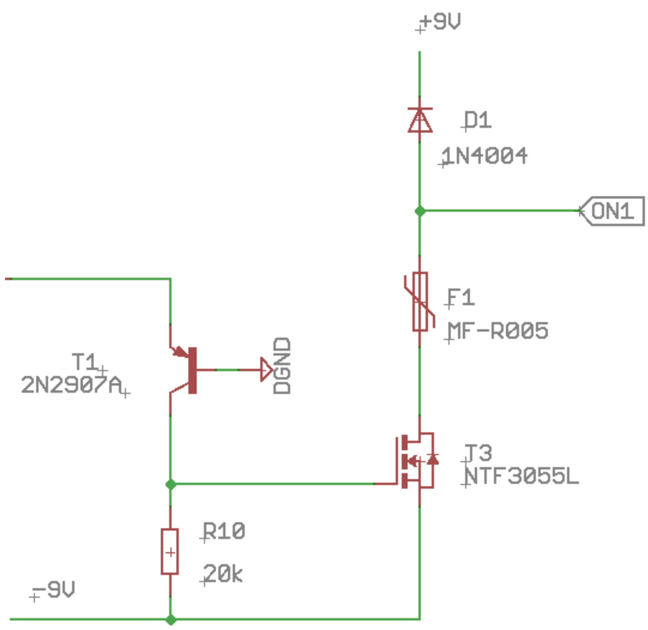


Figure 15: Relay driver stage for 2-channel ON/OFF remote control [12])

The connection of any output lines to system ground should be avoided. The load (normally any relay or solid-state switch) should be ground free.

[12]) F1 = MF-R010 on PCB-Version 2

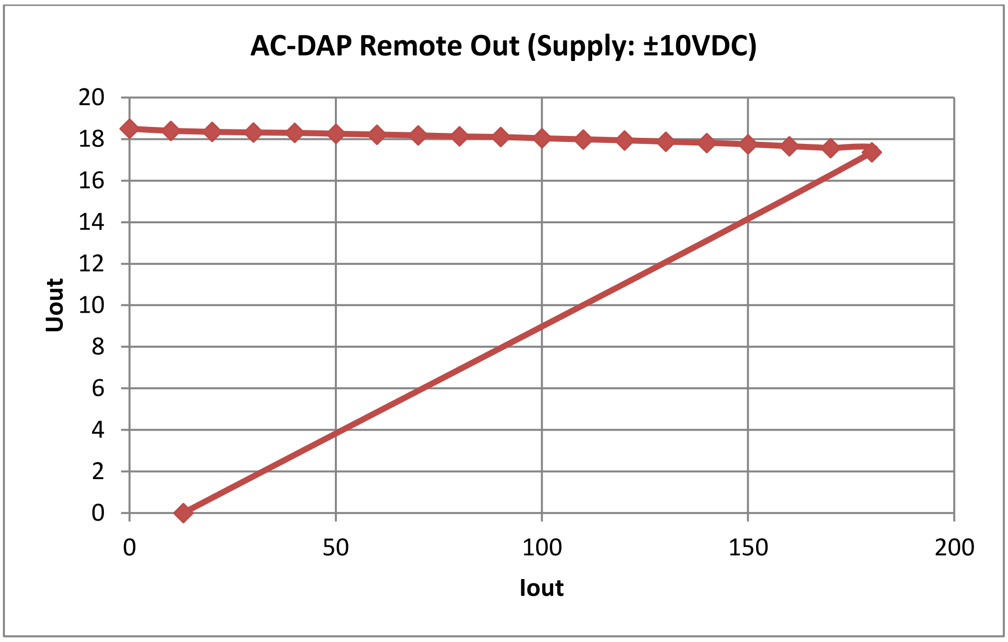


Figure 16: Remote Output driver capability and over current protection (V2/MF-R010)

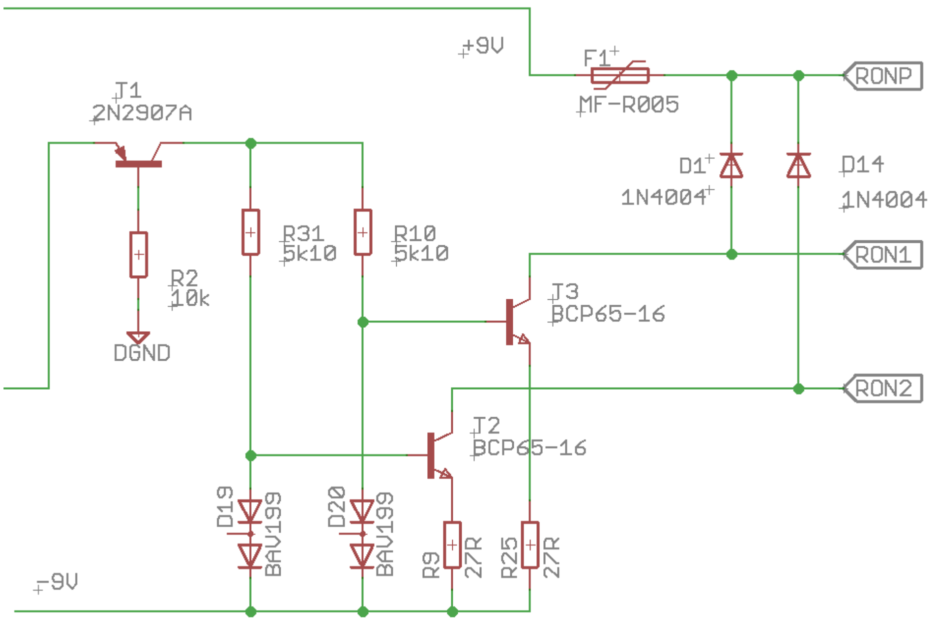


Figure 17: Relay driver stage for 2-channel ON/OFF remote control (constant current driver) [13])

[13]) Implementation for PCB-Version 3.2 and later

Digital Audio Processor for two-, three, or four-way Speaker Systems

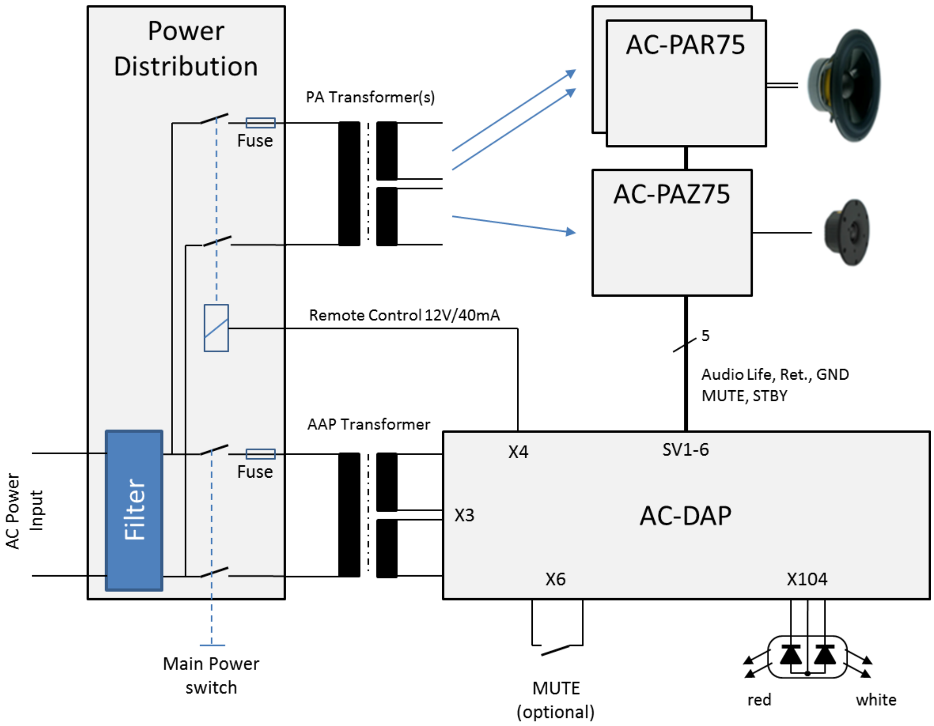


Figure 18: Recommended power distribution and control configuration

Table 8: Grounding/Star Point (X5)

X5 Pin	Parameter/Signal	Remark
1	System Ground Star Point (AGND)	This connector could be used as a Star Point (SP) in the system
2		
3		
4		

The GND of both audio input connectors are separately routed to this SP.

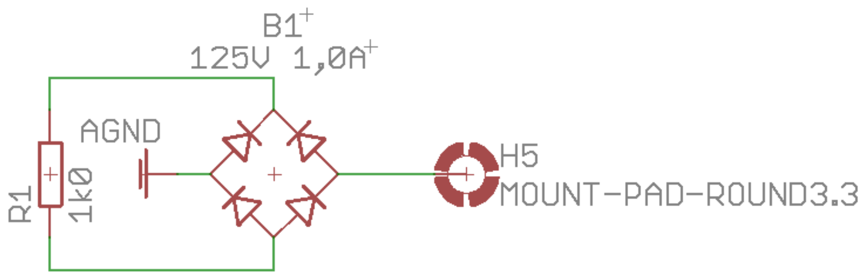


Figure 19: Connection from AGND (system ground) to the mounting flange

Table 9: Interface: Audio Input Connector Pinout (X1/X2)

X1/X2 Pin	Parameter/Signal	AUDIO CONN AT-XLR-F Cliff CP30070	Remark
1	Audio Input positive	2	
2	GND	1	On PCB internally connected to the Module Star Point X5
3	Audio Input negative	3	

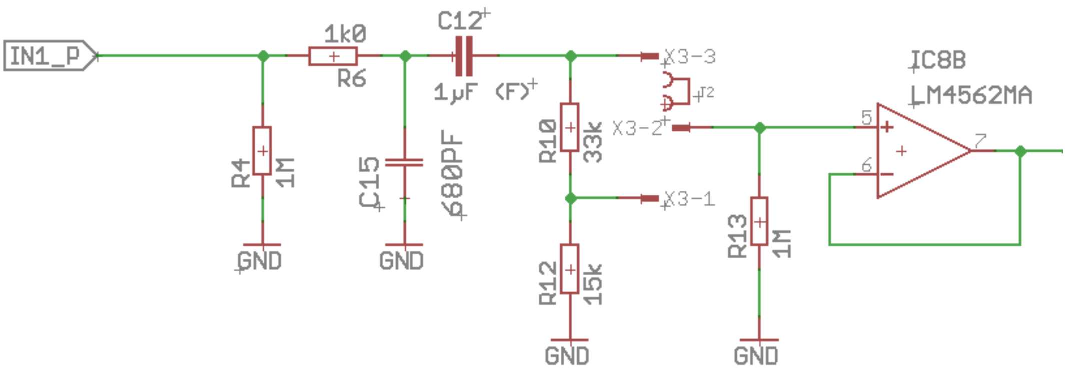


Figure 20: One Channel Audio Input Interface

The positive audio input of Audio-In 1 is taped for the Audio Signal Detection (ASD) circuit.

The input sensitivity could be selected by X201/X202 for Input X1 and X203/X204 for Input X2.

Position 2-3 for commercial level (default) and 2-1 for studio level.

X201/X202 as well as X203/X204 should have the same position.

Table 10: Interface: External Potentiometers (X302 and X312)

X302 X312 Pin	Parameter/Signal	Remark
1	+3V3A (positive DSP Supply)	Recommended value for external potentiometer 1kΩ until 10kΩ
2	Potentiometer center tap	
3	Signal Ground (GND)	

Remark: If used by external units the internal potentiometers R311 and R322 should be not populated

Table 11: Interface: External Potentiometers (X312) for PCB-Version 4 and later

X312 Pin	Parameter/Signal	Remark
1	Input AUX_ADC5	Recommended value for external potentiometer 1kΩ until 10kΩ or other voltage source with max +3V3
2	Input AUX_ADC4	
3	+3V3A (positive DSP Supply)	
4	Input AUX_ADC1	
5	Signal Ground (GND)	

Table 12: Interface: DIP-Switch SW301

The DIP-Switch SW301 is located on the module board and may be used if necessary.

SW301	Parameter/Signal	Remark
1	Activation Self-Boot General Purpose Switch	**PCB-Version 1: Self-Boot is activated if switch is open (OFF)** PCB Version 2 and later: Connected to DSP GPIO9. May be used as control signal to Microcontroller and X301/3 (Microphone)
2	General Purpose Switch	Connected to DSP GPIO6 and X301/4 (Microphone)
3	General Purpose Switch	Connected to DSP GPIO7 and LED303
4	General Purpose Switch	Connected to DSP GPIO8

Remark: The switch operates with internal pull-up resistors (closed/ON = low signal)
Remark: Switch 3 is connected in parallel with the LED303 (if LED should be used active by the DSP the switch shall remain in open (OFF) position)

Table 13: Interface: Available DSP GPIOs and their assigned function (PCB versions 1.0 and 2.0)

GPIO	Parameter/Signal	Remark
MP5	Out: CTRL0 (DSP Ready)	Connected to ATmega PD0
MP6	In: DIP-Switch SW301/2	Low Signal if switch is “ON”
MP7	Out: LED303 In: DIP-Switch SW301/3	Low Signal if switch is “ON” and connected to LED303 on AC-DAP (H = LED ON)
MP8	In: DIP-Switch SW301/4	Low Signal if switch is “ON”
MP9	Out: CTRL1	Connected to ATmega PD2 (not used)
MP11	Out: LED302	Connected to LED302 on AC-DAP (H = LEDON)
MP12	In: STBY	Status-Information “STBY” (Low when system is in STBY)
MP13	In: MUTE	Status-Information “MUTE” (Low when system is in MUTE)

Table 14: Interface: Available DSP GPIOs and their assigned function (PCB versions 3.0 and later)

GPIO	Parameter/Signal	Remark
MP5	Out: CTRL0 (DSP Ready)	Connected to ATmega PD0
MP6	In: MEMS_DATA	Used for digital Microphone
MP7	Out: LED303 In: DIP-Switch SW301/3	Low Signal if switch is “ON” and connected to LED303 on AC-DAP (H = LED ON)
MP8	In: DIP-Switch SW301/4	Low Signal if switch is “ON”
MP9	In: DIP-Switch SW301/1 Out: CTRL1	Connected to ATmega PD2 and X301/3 (not used)
MP11	Out: LED302 and external LED on X104-1	Connected to LED302 on AC-DAP (H = LED ON)
MP12	In: STBY	Status-Information “STBY” (Low when system is in STBY)
MP13	In: MUTE	Status-Information “MUTE” (Low when system is in MUTE)

Table 15: Interface: Available DSP GPIOs and their assigned function (PCB versions 4.0 and later)

GPIO	Parameter/Signal	Remark
MP5	Out: CTRL0 (DSP Ready)	Connected to ATmega PD0
MP6	In: MEMS_DATA (X301/4) SW301/2 X102/2 (with R329)	Used for digital Microphone
MP7	Out: LED303 In: DIP-Switch SW301/3	Low Signal if switch is "ON" and connected to LED303 on AC-DAP (H = LED ON)
MP8	In: DIP-Switch SW301/4 X303/3 and X102/1 (with R344)	Low Signal if switch is "ON"
MP9	In: DIP-Switch SW301/1 Out: CTRL1	Connected to ATmega PD2 and X301/3 (to be used for S/PDIF detection)
MP11	Out: LED302 and external LED on X104-1	Connected to LED302 on AC-DAP (H = LED ON)
MP12	X303/2	Optional available on X102/2 (with R311)
MP13	In: MUTE/STBY	Status-Information "MUTE" & "STBY" (low for system in MUTE/STBY)

Remark: For individual use of the GPIOs please contact AudioChiemgau. Different operation could require a change of PCB population.

Table 16: Interface: Linear/Potentiometer Assignments to DSP ADC inputs

AUXADC	Potentiometer/Connector	Remark
0	R311 / X302	See table above for connector pin assignment
1	R344 / X312	
2	R325	
3	Not used	Not available for external access
4	X312	For PCB-Version 4 and later only
5	X312	

MP5 shall be driven by the following configuration:

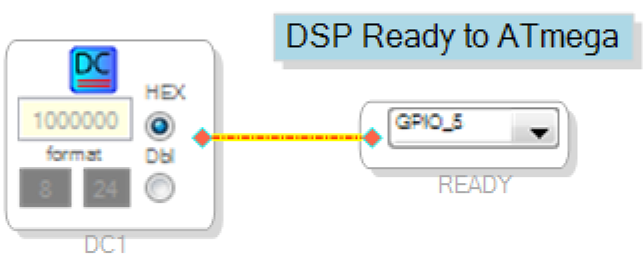
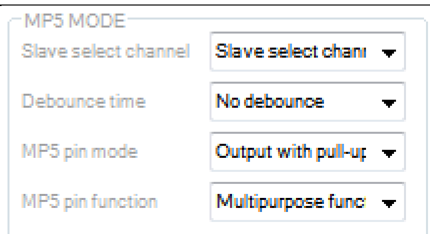
SigmaStudio® Configuration	Remark
	DSP ready signaling after boot process
	Register Control / Multipurpose

Table 17: Support-LEDs: LED301, LED302 and LED303:

Three LEDs are provided on the module board.

LED	Parameter/Signal	Remark
LED301	Control Port Header (USBi-Interface) connected	Not available on PCB V3 and later
LED302	General Purpose LED	Connected to DSP GPIO11 (ON if DSP output is high)
LED303	General Purpose LED	Connected to DSP GPIO7 (ON if DSP output is high)

Remark: LED303 is connected in parallel with the DIP-Switch SW301/3 (if LED should be used by the DSP the switch shall remain open (OFF) position)

Remark: Output for LED302 provides an additional driver to activate LED101(red) [14])

Table 18: Interface Connector for digital MEMS Microphone (X301):

Available with PCB Version 3.0 and later. This connector may be used for a digital microphone (MEMS based).

X301 Pin	Parameter/Signal	Remark
1	GND	DSP Ground
2	+3.3VD (digital)	Supply voltage for Microphone
3	CTRL1	Optional signal, not necessary for microphone
4	MEMS_DATA	Data-I/F from Microphone
5	MEMS_CLK	Clock-I/F to Microphone
6	GND	DSP Ground

Table 19: Interface Connector for digital I^2S Interface (X303):

Available with PCB Version 3.3 and later. This connector may be used for a digital input channel (I^2S).

X303 Pin	Parameter/Signal	Remark
1	SDATA_IN2	
2	LRCLK_IN2	
3	LRCLK_OUT2	Connected to DIP-Switch SW301/4 Must be open to use I^2S
4	BCLK_OUT2	
5	BCLK_IN2	
6	GND	DSP Ground

Remark: All I^2S signal lines are directly connected to the DSP core. Any possible ESD should be avoided to preserve the DSP from damage!

[14]) Implemented in PCB-Version 3.1 and later

Table 20: Interface: USBi-Connector X304

X304 Pin	Parameter/Signal	Remark
1	USBi-Power (+5V)	Self-boot if not connected
2	+3V3D from Target	Via 2k0 resistor
3	GND	System Ground
4	USB Reset	Reset Command from programming interface
5	SS	Target Select
6	SCLK	Serial Clock from programming interface
7	MOSI	Master Out Slave In
8	MISO	Master In Slave Out

Recommendation:

Connect X304 to a Cat.6A waterproof bulkhead coupler as panel or wall feed-through. Available from LogiLink as part number NP0083. This plug could be closed with the included dust cap and is then air tight as well.

For programming via standard LAN cable an adapter to the USBi-Programmer EVAL-ADUSB2EBZ (or similar) could be provided. Please contact AudioChiemgau for information.

Table 21: Interface: I^{2}C-Connector X102

X102 Pin	Parameter/Signal	Remark
1	I^{2}C SDA	
2	+5V (VCC)	Controller power supply
3	D0	Reserved
4	Spare 0	
5	I^{2}C SCL	
6	GND	

Remark: Controller power supply, provided on pin 2 could be used for external low-power equipment like the later mentioned health monitoring display including background illumination. Any overload should be avoided. 50mA are acceptable.

Table 22: Interface: I^{2}C-Connector X102 (on PCB-Version 4 and later)

X102 Pin	Parameter/Signal	Remark
1	DSP GPIO8	Connected to DSP GPIO 8
2	DSP GPIOxx	Connected to DSP GPIO 6 (default; optional GPIO 12)
3	I^{2}C SDA	I^{2}C serial data
4	+5V (VCC)	Controller power supply
5	D0	Reserved
6	OFF request	L on this input initiates a power-down sequence
7	I^{2}C SCL	I^{2}C serial clock
8	GND	Digital signal return
9	MUTE	

X102 Pin	Parameter/Signal	Remark
10	STBY	Could be used as status information or active driven by external equipment (open drain)

Remark: Controller power supply, provided on pin 2 could be used for external low-power equipment like the later mentioned health monitoring display including background illumination. Any overload should be avoided. 50mA are acceptable.

General Issues for DSP-Programming in Analog Device SigmaStudio® Environment

The following input port settings should be used:

Input	Parameter/Signal	Remark
0	Input X1	Input ADC1L on AD1938
1	Not used	-
16	Input X2	Input ADC2L on AD1938
17	Not used	-

The following output port settings should be used in **differential output configuration**:

Output	Parameter/Signal	Remark
0	Output Connector SV1/5 (SV6/5)	Positive Audio Signal Ch1
1	Output Connector SV1/4 (SV6/4)	Negative Audio Signal Ch1
16	Output Connector SV2/5 (SV3/5)	Positive Audio Signal Ch2
17	Output Connector SV2/4 (SV3/4)	Negative Audio Signal Ch2
32	Output Connector SV4/5 (SV5/5)	Positive Audio Signal Ch3
33	Output Connector SV4/4 (SV5/4)	Negative Audio Signal Ch3
40	Output Connector J501 tip and Output Connector SV501/5	Positive Audio Signal Ch4
41	Output Connector J501 ring and Output Connector SV501/4	Negative Audio Signal Ch5

The following output port settings should be used in single-ended output configuration:

Output	Parameter/Signal	Remark
0	Output Connector SV1/5	Positive Audio Signal Ch1 (SV1/4 GND)
1	Output Connector SV6/5	Positive Audio Signal Ch2 (SV6/4 GND)
16	Output Connector SV2/5	Positive Audio Signal Ch3 (SV2/4 GND)
17	Output Connector SV3/5	Positive Audio Signal Ch4 (SV3/4 GND)
32	Output Connector SV4/5	Positive Audio Signal Ch5 (SV4/4 GND)
33	Output Connector SV5/5	Positive Audio Signal Ch6 (SV5/4 GND)
40	Output Connector J501 tip and Output Connector SV501/5	Positive Audio Signal Ch7
41	Output Connector J501 ring and Output Connector SV501/4	Negative Audio Signal Ch8

For further details of the programming with SigmaStudio® see separate manual (currently attached to this document).

Digital Audio Processor for two-, three, or four-way Speaker Systems

Health-Monitoring and Operating Parameters:

The on-board Control Processor logs in the internal nonvolatile memory section (EEPROM) some operating parameters. This includes the number of switch-on cycles, the number of operating hours and the maximum temperature of the module during operation.

Additionally, some parameters are also stored within this memory section which will be used during operation. Most of the parameters are defined during initial test but some of them could be adjusted if required or necessary.

Supported Display with Status and Error Information:

An optional display could be connected to X102 to show any error or status information of the module.

The module uses an I^2C-Bus on X102 to support a four-line display with 20 characters based on the PCF8574T chip. The display should be listening on address 0x27.

Please contact AudioChiemgau for further information if necessary. The optional display is supported in PCB version 3 and F/W 4D20 or higher only.

Figure 21: Default screen for nominal operating PAR75 (V6 only)

In nominal operation the display content will be changed every five seconds. The following parameters are shown:

Display	Value	EEPROM Position	Remark
T	Temperature in °C	-	Current module temperature in °C
TM	Max Temperature in °C	0x06	Max temperature of the module in °C
ASD	Detected ASD level	-	Current detected input level used for ASD in Digital Numbers (DN) Max Value = 1023
LEV	Detected audio Signal level	-	Current detected input level in DN. Max Value = 1023
OASD	Offset ASD	0x16	Offset value used for ASD level calculation
OLV	Offset Level	0x18	Offset value used for audio level calculation
ON	Counted ON cycles	0x00	Accumulated number of counted power ON cycles Max Value = 65535
OP	Operating time	0x02	Accumulated operating hours in hours and minutes Max Value for hours = 65535

Display	Value	EEPROM Position	Remark
F/W	Version Firmware	0x3fC	Version of the operating control processor firmware
M	MUTE input	-	Current status of the MUTE input L = MUTE active; H = no MUTE condition
S	STBY Input	-	Current status of the STBY input L = STBY active; H = no STBY condition
VDD	DSP Supply voltage	-	Permanently observed DSP supply voltage +3,3V
VSS	DSP Supply voltage	-	Permanently observed DSP supply voltage +1,2V
ASD ON	ASD threshold for ON	0x12	Threshold value for ASD to switch on A signal level above this value turns the system on Default= 80
OFF	ASD threshold for OFF	0x14	Threshold value for ASD to switch off A signal level continuous below this value turns the system off Default = 100
T-Cor	Offset value for temperature	0x10	Offset value which will be used for temperature sensor calibration

In case of a failure mode or an error condition the following messages are possible:

Message	Status information and relief
FUSES not OK *Red LED dimmed on*	Programmed Fuses not in recommended default condition • Contact manufacturer
DC polarity wrong *Red LED flushes with 2 Hz*	In case the module is supplied with DC, the polarity of the supply is wrong • Turn positive and negative supply connected on X1

Continuous status information message (F/W 5D00 and later)

In parallel to the I²C output addressed to the display the on-board controller sends periodically a predefined data block with status information in raw format. This block/array with currently 16 Bytes is addressed to 0x50 and contains the following information:

Position/Index	Content
0	Module (Controller) Temperature
1	Module (Controller) max Temperature
2	Number of switch-on cycles
3	
4	Number of operating time (hours)
5	
6	Status of TOSLINK (0 = off | 1 = active)
7	Status of ASD. Above (1) or below (0) threshold level
8	Number of operating time (minutes)
9	Spare (not relevant/don't care)
10	F/W Version of the controller (string with four ASCII characters)
11	
12	
13	
14	Current Status of the system (ON|OFF|MUTE|STBY)
15	Reserved (don't care)

Remark: This content is the current status with controller firmware 5. Changes are possible without further notice.

Firmware update:

If necessary, the firmware of the on-board controller may be easily updated via the SPI interface provided on the connector X105. Please contact manufacturer for further information.

Used and implemented type of microcontrollers:

PCB Version	Serial Numbers	Implemented Microcontroller	Remark
V1	S/N #01 - #36	ATmega328-AU	In-System-Programmable via SPI (X105)
V2			
V3			
V3.2	S/N #37 - #48	ATmega328PB-AU	
V3.3	S/N #49 - #68	ATmega328-AU	
V4.0	S/N #69 and more	ATmega328-AU	

Digital Audio Processor for two-, three, or four-way Speaker Systems

As described in the following chapters the on-board DSP is in-system-programmable using an SPI.

Used and implemented type of DSP:

PCB Version	Serial Numbers	Implemented DSP	Remark
V1	S/N #01 - #48	ADAU1452WBCPZ-300	In-System-Programmable via SPI (SV301)
V2			
V3			In-System-Programmable via SPI (X304)
V3.3	S/N #49 - #56 S/N #59 - #68	ADAU1462WBCPZ-300	
V3.3	S/N #57 & #58	ADAU1466WBCPZ-300	
V4.0	S/N #69 and more	ADAU1466WBCPZ-300	

Major changes in hardware, relevant for programming:

During ongoing development process some hardware changes are realized. The following table of changes are relevant for programming of the DSP.

PCB Version	Serial Numbers	Major Changes
V1		
V2		
V3		
V3.2	S/N #37 - #48	DSP GPIO13 used for MUTE DSP GPIO12 used for STBY
V3.3	S/N #49 - #68	DSP GPIO13 used for MUTE <u>and</u> STBY
V4.0	S/N #69 and more	DSP GPIO12 available for I^{2}S support on X303

Further details may be provided on request.

Configuration and self-boot memory content as factory setting

The schematic in Figure 22 shows the basic function, which is implemented in DSP as factory setting. This content will be loaded in self-boot mode at startup or after reset cycle.

Remark: Take care of this content. There is no frequency separation. All outputs providing the input signal. This content is for test purpose only.

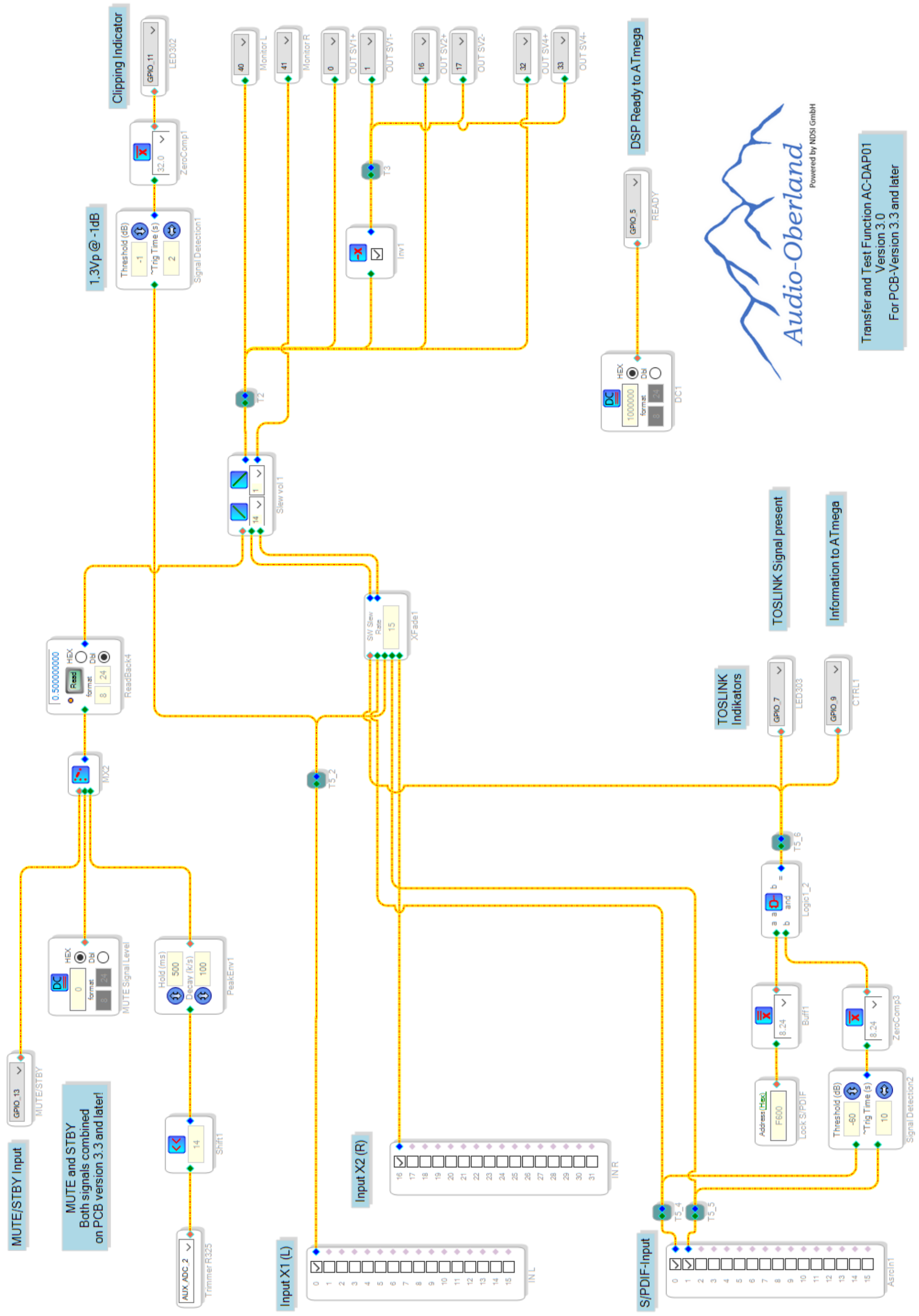


Figure 22: Self-boot content at time of delivery (Version 2.0)

Digital Audio Processor for two-, three, or four-way Speaker Systems

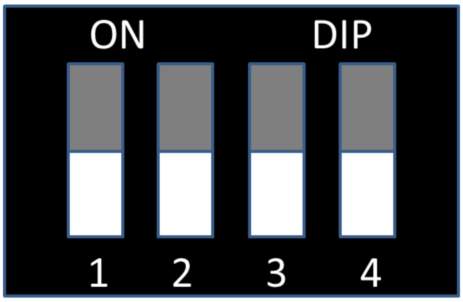
Basic Function after self-boot in factory configuration

In the basic configuration only the monitor output (3,5 mm jack plug) will deliver a signal from both inputs X1 and X2 (stereo).

In differential configuration SV1, SV2 and SV4 will deliver the signal present on X1.

Depending on the settings on DIP-Switch SW301 the DSP will deliver the following output signals:

DIP-Switch Settings:



1: PCB Version 1.0: Self-Boot (OFF: Enabled)

PCB Version 2.0 and later: Use for any purpose

2: Test LED (ON: LED303 ON)

3: Not used (don´t change)

LED302 is used as clipping indicator

4: Not used (don´t change)

The Output-Volume could be adjusted by R325.

With activated MUTE and/or STBY the DSP will not deliver any output signal.

LED Function:

LED302: Signal Clipping

LED303: Test purpose. To be activated by DIP-Switch SW301/2

Ordering Guide

The following combinations are available:

Model	Analog Inputs	Digital Input (S/PDIF)	Power Regulation (+5V Supply)
AC-DAP01-1OL	1	Optical	Linear
AC-DAP01-1OD			DC/DC
AC-DAP01-1CL	1	Coaxial (SMA)	Linear
AC-DAP01-1KD			DC/DC
AC-DAP01-2OL	2	Optical	Linear
AC-DAP01-2OD			DC/DC
AC-DAP01-2CL	2	Coaxial (SMA)	Linear
AC-DAP01-2CD			DC/DC

AC-DAP01-2OD is the standard configuration. Other combinations on request and with extended lead times. Default configuration for outputs: four channels balanced.

Digital Audio Processor for two-, three, or four-way Speaker Systems

Document Change History:

Version 1.1 to 1.3 initial Issues

Version 1.4: Changes for PCB Version 3.0 implemented

Version 1.5: Further Changes for PCB 3.0 implemented

Version 1.6: Further Changes for PCB 3.2 implemented
Remote driver stage changed from voltage to current
Update of Figure 22
Minor corrections

Version 1.7: Ordering Guide and I^2S Interface added
Update of **Figure 1** and Figure 22 (PCB-Version 3.3)
Minor corrections

Version 2.0: Changes necessary for new PCB layout 4.0 with some minor development steps
Further clarifications and additional information

PCB Change History (major issues):

Version 1.0: Initial Configuration, partly upgraded to Version 2.0 (see documentation for V1.0)

Version 2.0: Self-boot activation via USBi (Self-boot if SV302/X304 is not connected)

Version 3.0: Interface Connector X301 for digital MEMS microphone added
DSP supply monitoring extended for +1.2V
Connector SV501 added and connected parallel to J501 (Monitor)
SV301 changed to X304 (RJ45)
Buffer (IC302) for Programming interface implemented

Version 3.2: Remote output stage changed from voltage to current driver.

Version 3.3: Optional digital (S/PDIF) coaxial input deliverable
Connector for I^2S input implemented

Version 4.0: Additional access to AUXADC4 and AUXADC5
Necessary implementation for remote control

Evaluating the AudioChiemgau Digital Audio Processor AC-DAP

GENERAL DESCRIPTION [15])

This user guide describes the setup and operation of the AC-DAP board. This board operates with the software for the ADAU1452 SigmaDSP® processors.

The AC-DAP board provides access to the digital serial audio ports of the ADAU1452 and some of the general-purpose input/outputs (GPIOs). An analog inputs and outputs are provided by the AD1938 codec.

The ADAU1452 core is programmed using the Analog Devices, Inc., SigmaStudio® software, which interfaces to the board via the universal serial bus interface (USBi). The on-board electronically erasable programmable read-only memory (EEPROM) can be programmed for self-boot mode.

The integrated oscillator circuit and the on-board, 12.288 MHz passive crystal (Q305) provides the master clock.

For full details, see the ADAU1452 and AD1938 data sheets, which should be used in conjunction with this user guide when using the AC-DAP board.

Remark: An additional Application Note (AN) is available on request from AudioChiemgau for a detailed description of necessary or recommended operating blocks (in German only).

EQUIPMENT REQUIRED

- EVAL-ADUSB2EBZ communications adapter (USBi)
- USB cable with mini-B plug
- SigmaStudio Application (Analog Devices)

SETTING UP THE AC-DAP BOARD

Using the AC-DAP board requires a PC running a Windows® 7 or later operating system with a USB interface, the USBi, and an internet connection.

The PC communicates with the board via the USBi.

The software tool used with the ADAU1452 is SigmaStudio, a fully graphical user interface (GUI)-based programming environment. No digital signal processing (DSP) programming is required. A full version of SigmaStudio, which includes a library of DSP building blocks and the required USBi drivers, can be downloaded from the SigmaStudio software page on the Analog Devices website at www.analog.com/SigmaStudio.

INSTALLING THE SIGMASTUDIO SOFTWARE

To download the latest version of Sigma-Studio, take the following steps:

1. Go to the SigmaStudio software page on the Analog Devices website and select the latest version of the SigmaStudio software from the Downloads and Related Software section.
2. Determine whether the software must be installed on a 32-bit or 64-bit version of Windows and locate the latest, corresponding release version of SigmaStudio.
3. Download the installer and run the executable file. Follow the prompts in the program and accept the license agreement to install the software.

[15]) All issues addressed to ADAU1452 are applicable for all possible implemented compatible DSPs like ADAU1462 or ADAU1466

INSTALLING THE AC-DAP (USBI) DRIVERS

To use the USBi, install SigmaStudio first (see the Installing the SigmaStudio Software section). After installing the SigmaStudio, take the following steps:

1. Connect the USBi to an available USB 2.0 port using a USB cable. The USBi does not function properly with a USB 3.0 port.
2. Install the driver software as described in the following chapter.

USING WINDOWS 7 OR LATER

After connecting the USBi to the USB 2.0 port, Windows 7 or later recognizes the device and installs the drivers automatically. After the installation is complete, leave the USBi connected to the PC.

To confirm that the USBi drivers are properly installed, take the following steps:

1. With the USBi still connected to the computer, check that both the yellow I^{2}C LED and the red power indicator (D4) LED on the USBi interface board are illuminated.
2. In the Windows Device Manager under the Universal Serial Bus controllers section, check that Analog Devices USBi (programmed) is displayed.

DISABLING THE SELF BOOT SWITCH

When setting up the AC-DAP board, ensure that the first switch of the four-position, dual, inline package (DIP) switches, SW301/1, is in the **ON** position.

In this case the ADAU1452 is prevented from executing a self-boot operation at power-up. When the switch is in the **OFF** position Self-boot causes the ADAU1452 to attempt to load code from an EEPROM when the device powers up or comes out of reset.

<u>With PCB version 2.0 or later self-boot is active with disconnected SV301 (USBi connector). Don’t care the SW301/1.</u>

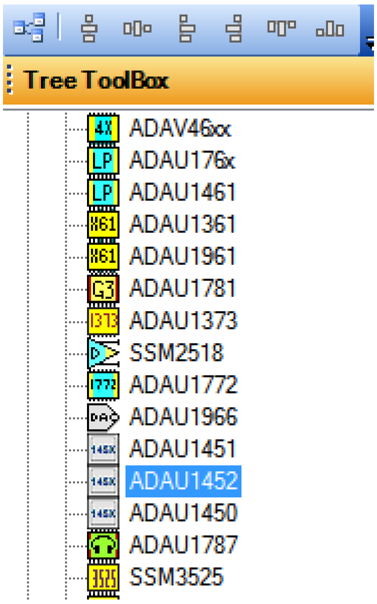
SETTING UP COMMUNICATIONS IN SIGMASTUDIO

To set up communications with the AC-DAP board in SigmaStudio, take the following steps:

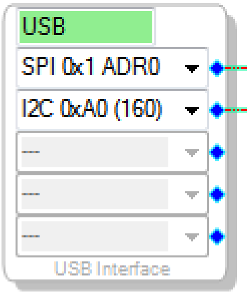
1. Start the SigmaStudio software either by double-clicking the desktop shortcut or by finding and running the executable file in File Explorer.
2. To create a new SigmaStudio project, select New Project from the File menu or by pressing the Ctrl + N keys. The Hardware Configuration tab is the default view of the new project.
3. In the Hardware Configuration tab, add the appropriate components to the project space by clicking and dragging them from the left Tree ToolBox panel to the empty white project space on the right of the window. The user can change the names of the component blocks as desired.
4. Add a USB Interface block (the USBi) to the project by clicking USBi from the Communication Channels subsection of the Tree ToolBox (see Figure), and then dragging it to the project space.

5. Add an IC 1 block to the project by clicking the ADAU1452 component from the Processors (ICs/DSPs) subsection and then dragging it into the project space [16]).

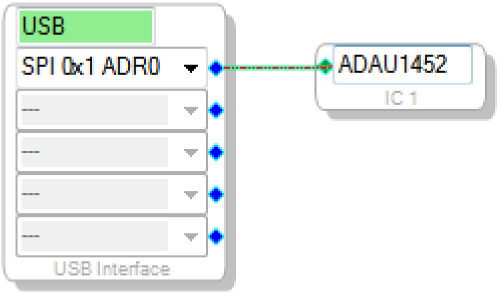
[16]) It is recommended to select the ADAU1452 also for compatible DSPs like ADAU1462 or ADAU1466



6. Ensure that SigmaStudio can detect the USBi on the USB port of the PC by checking if the background of the USB label is green in the USB Interface block:



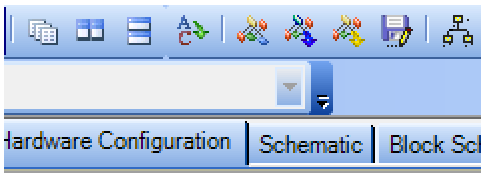
7. When SigmaStudio cannot detect the USBi connected to the PC USB port, the background of the USB label is red. This error can occur either when the USBi is not connected to the port or when the drivers are installed incorrectly.
8. To connect the USB Interface block to the IC 1 block, the ADAU1452, click and drag a line representing a wire between the blue pin of the USB Interface block and the green pin of the IC1 block. This connection allows the USBi to communicate with the ADAU1452. The corresponding drop-down field of the USB Interface block automatically fills with the default mode and channel for the connected IC1 block. With the ADAU1452, the default communications mode is SPI, the default slave select line is 1, and the default address is 0.



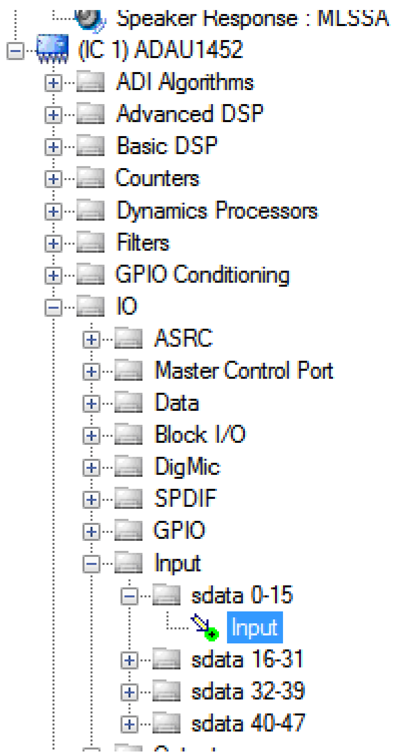
CREATING A BASIC SIGNAL FLOW

To create a signal processing flow, take the following steps:

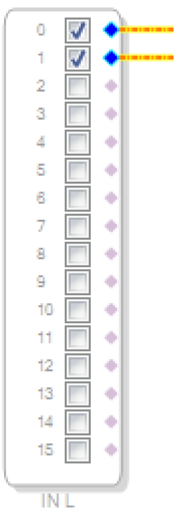
1. Click the Schematic tab near the top of the window.



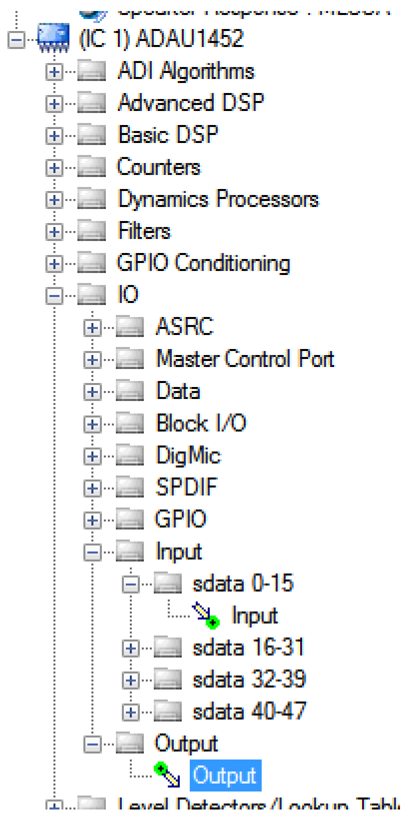
2. To add the appropriate elements to the project space, click and drag the elements from the Tree ToolBox to the empty white project space on the right of the window. The Tree ToolBox contains all the algorithms that can run in SigmaDSP.
3. Add an Input1 block by clicking the Input component from the (IC 1) ADAU1452 > IO > Input > sdata 0-15 folder (see following Figure),



and then dragging it into the empty project space. The Input1 block, which represents an input channel, then appears as shown in the Figure. By default, Channel 0 and Channel 1 are selected and this configuration matches the analog audio source hardware connections. Therefore, no modifications are needed.



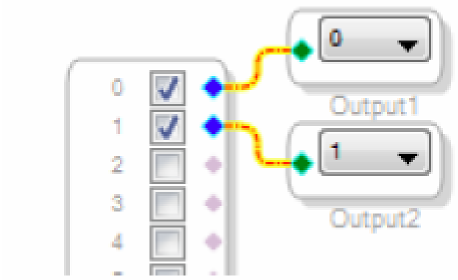
4. Add two output blocks, Output1 and Output2, by clicking the Output component from the (IC 1) ADAU1452 > IO > Output folder (see Figure), and then dragging it into the project space. Ensure that these blocks, which represent output channels, are assigned to Channel 0 and Channel 1.



5. Repeat Step 4 to add another output (see Figure).

6. To connect each Input1 channel to the corresponding output block, click and drag a line representing a wire between the blue pin of the Input1 block and the green pin of the corresponding output block (see Figure). Input1 Channel 0 connects to Output1 Channel 0 and Input1 Channel 1

connects to Output2 Channel 1.

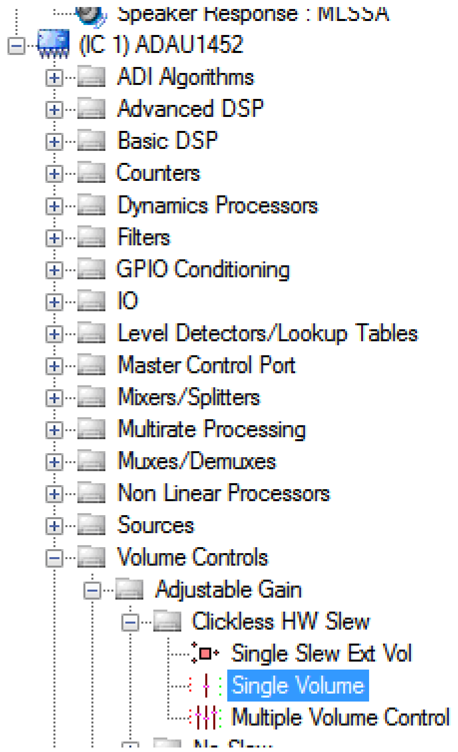


The default register settings in SigmaStudio are configured to match the AC-DAP board hardware, including the signal routing between the ADAU1452 and the AD1938 codec. When these steps are complete, the basic signal flow is complete and the stereo analog input source passes directly through the SigmaDSP and connects to the stereo analog output.

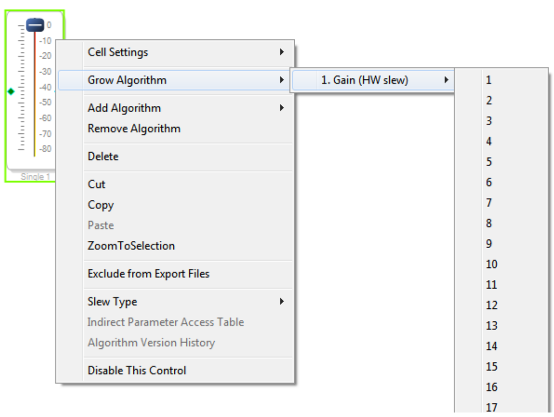
ADD VOLUME CONTROL

To add volume control functionality to the project, take the following steps:

1. Add a Single1 block (volume control) by clicking the Single Volume component from the Volume Controls > Adjustable Gain > Clickless HW Slew folder (see Figure), and then dragging it to the project space.

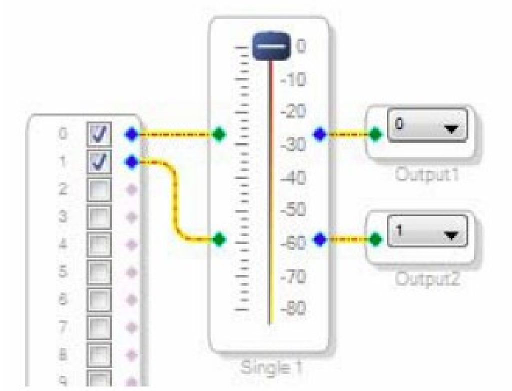


2. By default, the Single1 block has one input and one output, meaning it is a single channel. To add another channel, right-click the empty white space of the Single1 block and from the drop-down menu that appears, select the Grow Algorithm > 1. Gain (HW slew) > 1 option (see Figure).
3. To delete the existing yellow connection wires, the connections added in Step 6 of the Creating a Basic Signal Flow section, click the connection wires and then press the Delete key.



4. Connect the blocks as shown in the following Figure

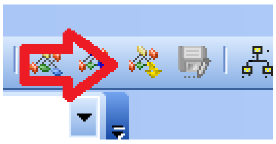
Digital Audio Processor for two-, three, or four-way Speaker Systems



After performing these steps, the schematic of the completed signal flow (see Figure) is ready to be compiled and downloaded to the board.

DOWNLOADING THE PROGRAM TO THE DSP

After the signal is completed, compile and download the DSP code to the DSP either by clicking the Link/Compile/Download button in the main toolbar of SigmaStudio (see Figure) or by pressing the F7 key.



After the code is downloaded to the DSP, the following events occur almost simultaneously:

- If the compiler finishes compiling the project, the compiled data downloads from SigmaStudio via the USBi to the ADAU1452, and the SigmaDSP starts operating.
- The status bar in the lower right corner of the SigmaStudio window turns from blue to green and the text changes from Design Mode to Active: Downloaded. Until this point, SigmaStudio is in design mode, as noted by the blue status bar and the Design Mode text in the lower right corner of the SigmaStudio window. Downloaded Text and Green Status Bar
- The signal flow runs on the AC-DAP board and the audio passes from the analog input to the analog output. To change the volume in real time, click and drag the volume control slider in the Single1 block in the Schematic tab.

ADDING S/PDIF INPUT TO THE PROJECT

The AC-DAP board has one optical S/PDIF interface. This interface is an optical input that converts the optical signal to an electrical signal that is sent to the ADAU1452 S/PDIF receiver (the SPDIFIN pin).

The connector is located on the top side of the PCB as connector LW301.

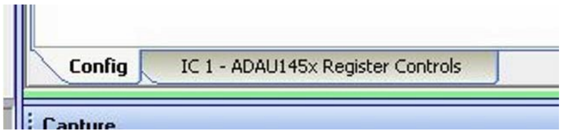
The S/PDIF receiver accepts signals with sample rates between 18 kHz and 192 kHz. Because the incoming signal is asynchronous to the system sample rate, use an ASRC to convert the sample rate of the incoming signal. Optionally, configure the SigmaDSP core to start processing audio samples based on the sample rate of the incoming S/PDIF receiver signal, negating the need for an ASRC. However, users are strongly recommended to use an ASRC for performance and reliability reasons.

The S/PDIF receiver carries two channels of uncompressed audio.

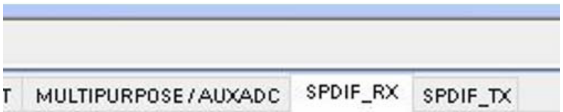
To add a stereo S/PDIF input to the project in SigmaStudio, take the following steps:

1. Connect an S/PDIF source to the AC-DAP board by connecting a standard TOSLINK® optical cable to the S/PDIF receiver connector LW301.
2. Configure the S/PDIF input and output by modifying the ADAU1452 registers with the following steps in order:

3. Click the Hardware Configuration tab, then click the IC 1 – ADAU145x Register Controls tab at the bottom of the window (see Figure).



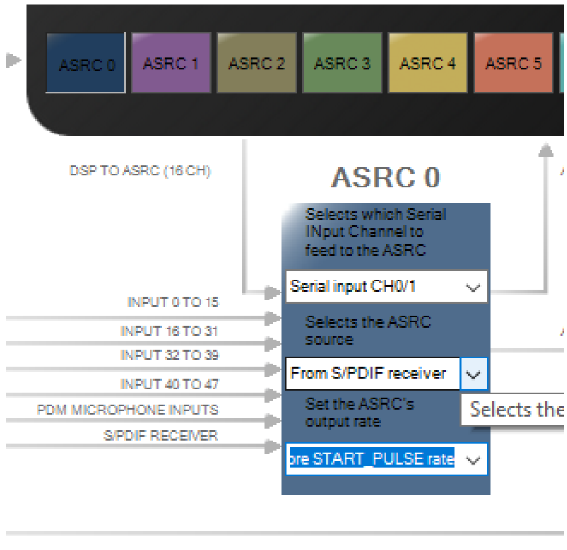
4. Next, click the SPDIF_RX tab (see Figure).



5. Enable the SPDIF_RESTART register by clicking the Do not automatically restart the audio on relock button (see Figure). After clicking this button, the text displayed on the button changes to restarts the audio automatically on relock and the color changes from red to green (see Figure).

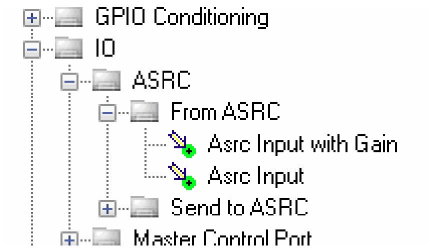
6. Click the ROUTING_MATRIX tab in the register controls tab shown in Figure to allow users to configure the routing matrix (see Figure below).

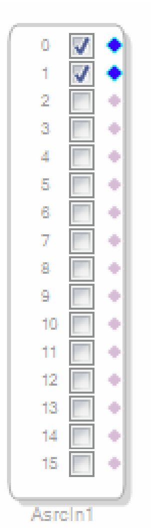
7. To configure the S/PDIF receiver signal routing, click the first asynchronous sample rate converter (ASRC) button, ASRC 0 (see Figure), in the ROUTING_MATRIX tab to open the window in the Figure. Configure the ASRC 0 routing matrix registers using the drop-down menus until the menus match the Figure. This configuration routes the S/PDIF receiver signal through an ASRC before the signal is accessed in the DSP core. It is necessary to route the signal through the ASRC because the clock recovered from the S/PDIF source is not synchronous to the ADAU1452.



8. Close the SPDIFTX INPUT dialog box.

9. Add an ASRC block (the S/PDIF input) to the project by clicking the ASRC Input component from the IO > ASRC > From ASRC folder (see Figure), and dragging it to the project space where it appears as shown in Figure.





Because the left and right signals of the S/PDIF receiver pass through the ASRC0, the input to the DSP program is the Asrcin1 block in SigmaStudio. The naming convention for the input and output blocks is such that all blocks in SigmaStudio are named from the perspective of the DSP core. Therefore, the Asrcin1 block in SigmaStudio represents the input to the DSP from the ASRC outputs.

The inputs to the ASRCs themselves are defined in the register window (see Figure). By default, Channel 0 and Channel 1 are active when the corresponding checkboxes are selected. Because the ASRC 0 outputs correspond to Channel 0 and Channel 1, use the default configuration shown in the Figure.

ASRC 0	Channel 0 and Channel 1

MULTIPURPOSE (MPx) PINS

The MPx pins on the ADAU1452 can use GPIOs when configured to do so by using the ADAU1452 control registers. Of the 14 MPx pins, three are connected to switches (SW301) that pull the signals low or tie them high, two are connected to high impedance inputs to LED drivers, and two GPIO pins are available for communication with the ATmega. Additional GPIOs may be used to detect the MUTE- and STBY-Status. The remaining pins are used for other functionalities and cannot be used as multipurpose pins.

A table in the data sheet section describes the MPx pins available for use as GPIOs, along with the corresponding functionalities on the board.

AUXADCx PINS

The ADAU1452 AUXADC is a 10-bit, successive approximation register (SAR) multiplexed across six input channels. These channels are used for analog control signals to the DSP. Channel AUXADC0, AUXADC1 and AUXADC2 are connected to Linear Potentiometer R311, R344 and R325.

COMMUNICATIONS HEADER

The communications header (SC301) is a 10-pin header designed to work with the EVAL-ADUSB2EBZ (USBi). The SPI signals are wired from the communications header to the corresponding SPI slave port pins on the ADAU1452. The I2C pins are not used in this design. A reset line is also included, which allows the user to reset the devices on the EVAL-ADAU1452REVBZ board via a command in SigmaStudio, search the EngineerZone™ forums or contact SigmaDSP@analog.com for more information. When the EVAL-ADUSB2EBZ board is connected to the AC-DAP board and PC, powered, and recognized by the PC, LED301 illuminates. SigmaStudio also controls the 5 V output to LED301.

SELF BOOT

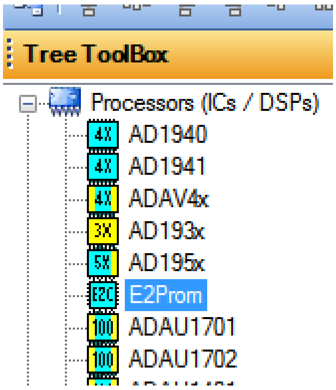
A 1 Mb, 20 MHz, SPI, serial EEPROM memory is included on the AC-DAP board. The ADAU1452 can boot and execute a program without help from an external microcontroller. This self-boot feature allows any project developed within SigmaStudio to execute on a rising edge of the RESET pin signal or when the ADAU1452 powers up.

Switch 1 of the SW301 DIP switch sets the state of the ADAU1452 SELFBOOT pin. Setting

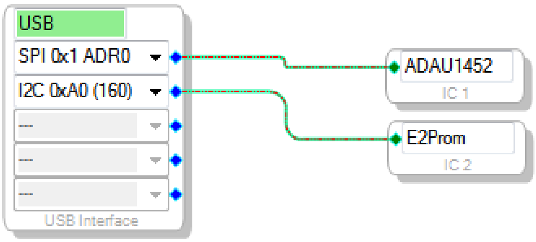
Section 1 of the SW301 switch to **ON** disables the self-boot feature.

To use the self-boot functionality, take the following steps:

1. Add an IC2 (E2Prom) block to the project space of the Hardware Configuration tab from the Tree ToolBox. From the Processors (ICs / DSPs) folder, click E2Prom (see Figure) and drag it to the project space.

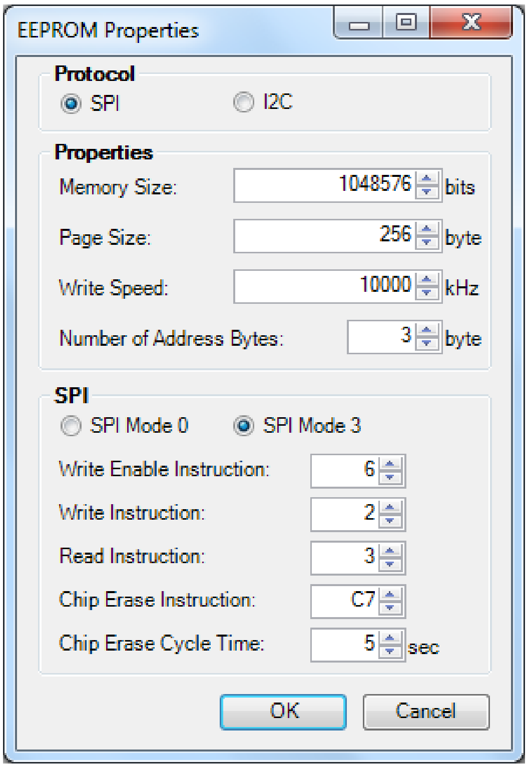


2. Connect the green input pin of the IC2 block to one of the available blue output pins of the USB Interface block in SigmaStudio.
3. Set the communication mode to SPI 0x1 ADR0 by clicking the text fields of the USB Interface block (see Figure). There is no physical connection between the USBi connector (a connector to the evaluation board) and the evaluation EEPROM on the AC-DAP board. Therefore, SigmaStudio cannot directly communicate with the EEPROM. To circumvent this lack of communication, SigmaStudio writes a program to the ADAU1452, which then uses the master SPI port to write the self-boot data to the EEPROM. Adding the IC2 block allows users to configure the EEPROM and informs SigmaStudio that a hex file must be produced.
4. The pull-down text field in the USB Interface block sets the type of communication the EEPROM uses. I^2C is supported communication type and can differ from the communication standard used by ADAU1452 the slave port. As shown in the Figure, users are recommended to connect the IC2 block to an unused pin on the USB Interface block.



5. Before downloading the self-boot data to the EEPROM, click the Link/Compile/Download button or press the F7 key to compile the SigmaStudio project file.
6. When writing to the EEPROM, set the self-boot switch (Section 1 of Switch SW301) to the disabled position (**ON**) [17]).
7. To write to the EEPROM through the DSP master SPI port, right-click the white space in the IC1 block in the Hardware Configuration tab. From the pull-down menu that appears, select the Self-boot Memory >Write Latest Compilation through DSP option.
8. An EEPROM Properties dialog box appears. Enter the values shown in following Figure to the text fields and then click OK.

[17]) Not necessary for boards with version 2.0 and higher



9. A warning dialog window appears to remind the user that executing an external memory write erases and overwrites any data currently stored on the EEPROM. Click OK to proceed.
10. SigmaStudio begins the EEPROM write operation. This operation can take several minutes to complete. Users can view the progress of the write operation with the status window shown. When the window disappears, the operation is complete. [18])

To execute a self-boot operation, take the following steps:

1. Set the SELFBOOT switch (Section 1 of SW301) to the enabled position (**OFF**) on AC-DAP board.
2. Turn the system off and on again to initialize a self-boot operation and the DSP runs the user created program.

For PCB version 2 and higher:

1. Turn the system off
2. Disconnect the USBi from the AC-DAP
3. Turn the system on again to initialize a self-boot operation and the DSP runs the user created program.

The Content of this chapter is based on the Analog Devices Document "EVAL-ADAU1452REVBZ User Guide" UG-1662".

[18]) Check parameter settings after first programming cycle and repeat the procedure in case parameters are changed