

Audio Codec Adapter Setup for Cinterion[®] Modules

User Guide

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0 Document History

Preceding document: "Audio Codec Adapter Setup for Cinterion® Modules" v03

New document: "Audio Codec Adapter Setup for Cinterion® Modules" v04

Chapter	What is new
Throughout document	Added NAU8822 audio codec adapter.
1.2	Added ELS61x, ELS81x, and ALS5 as supported products.

Preceding document: "Audio Codec Adapter Setup for Cinterion® Modules" v02

New document: "Audio Codec Adapter Setup for Cinterion® Modules" v03

Chapter	What is new
Throughout document	Removed ELS51 and ELS31-USA products.
3.1.1.4	Added settings for AT^SAIC command for ELS31 in Table 4 .

Preceding document: "Audio Codec Adapter Setup for Cinterion® Modules" v01

New document: "Audio Codec Adapter Setup for Cinterion® Modules" v02

Chapter	What is new
3.1.1.1	Added note that ELS31/ELS51 modules cannot be configured as masters.
4	New Appendix A - DAI Signal Timings at Oscilloscope .

New document: "Audio Codec Adapter Setup for Cinterion® Modules" v01

Chapter	What is new
---	Initial document setup.

1 Introduction

The Audio Codec Adapter is a simple and easy-to-use universal adapter board designed to quickly evaluate the digital audio functionality of the supported Gemalto M2M (evaluation) products. Using this reference adapter, it is possible to develop and test external audio applications for the supported Cinterion® modules (see [Section 1.2](#)).

This document¹ guides you through the process of connecting the Audio Codec Adapter correctly and getting started with the evaluation of the module's audio functionality.

The Audio Codec Adapter is intended for use in test environments only. All persons handling the Audio Codec Adapter must be properly trained in electronics and observe good engineering practice standards.

The Audio Codec Adapter is a test/development board, and has not been designed to be embedded into other products.

1.1 Related Documents

- [1] AT Command Set for your product
- [2] Hardware Interface Description for your product
- [3] DSB75 Hardware Description, v14
- [4] Getting Started with your product

1.2 Supported Products

This User Guide applies to the following products:

- AGS2-E, AGS2-W
- BGS2-E, BGS2-W
- BGS5, BGS8
- EHS5-E, EHS5-US, EHS6, EHS8
- ELS31-VA
- ELS61-E, ELS61-E2, ELS61-ER2, ELS61-US, ELS61-AUS, ELS81-E, ELS81-US
- PDS5, PDS6, PDS8
- AHS3-E, AHS3-W
- PHS8-P, PHS8-E, PHS8-US, PHS8-USA
- ALS3-E, ALS3-US, ALS3-C
- PLS8-E, PLS8-US, PLS8-V, PLS8-X, PLS8-J
- ALAS5

¹ The document is effective only if listed in the appropriate Release Notes as part of the technical documentation delivered with your Gemalto M2M module.

2 Audio Codec Adapter Overview

This chapter gives an overview of the available Audio Codec Adapters, i.e., MAX9860 in [Section 2.1](#) and NAU8822 in [Section 2.2](#) employed by Gemalto M2M to evaluate the digital audio interface functionality of the supported modules.

2.1 MAX9860 Audio Codec Adapter

The MAX9860 Audio Codec Adapter is based on the MAX9860 16-bit mono audio voice codec from Maxim Integrated. The adapter board can be plugged to the DSB75 (or DSBmini) in order to be used with a variety of Cinterion® modules (see [Section 1.2](#)), and supports the following features:

The Audio Codec Adapter supports the following features:

- Directly connects to DSB75 (or Mini-DSB)
- Most important configuration settings are accessible via DIP switch
- Almost all configuration settings are accessible via RS232 serial interface
- PCM/I2S support
- BCLK = 256(only in slave mode), 512, 1024, 2048kHz
- Master/Slave mode
- 16kHz WB/8kHz NB mode
- Handset/Headphone support
- Short/long FSYNC²
- Sending DTMF sound sequences on analogue or on digital (long FSYNC Intel mode)
- Recognizing beep sounds by means of FFT on analogue or on digital (long FSYNC Intel mode)

2.1.1 Interfaces

[Figure 1](#) and [Figure 2](#) show various interfaces on the top and bottom sides of the Audio Codec Adapter.



Figure 1: Audio Codec Adapter top view

² **Note:** The short frame master mode is not usable because the MAX9860 voice codec generates wrong AD converter output at bit 14.

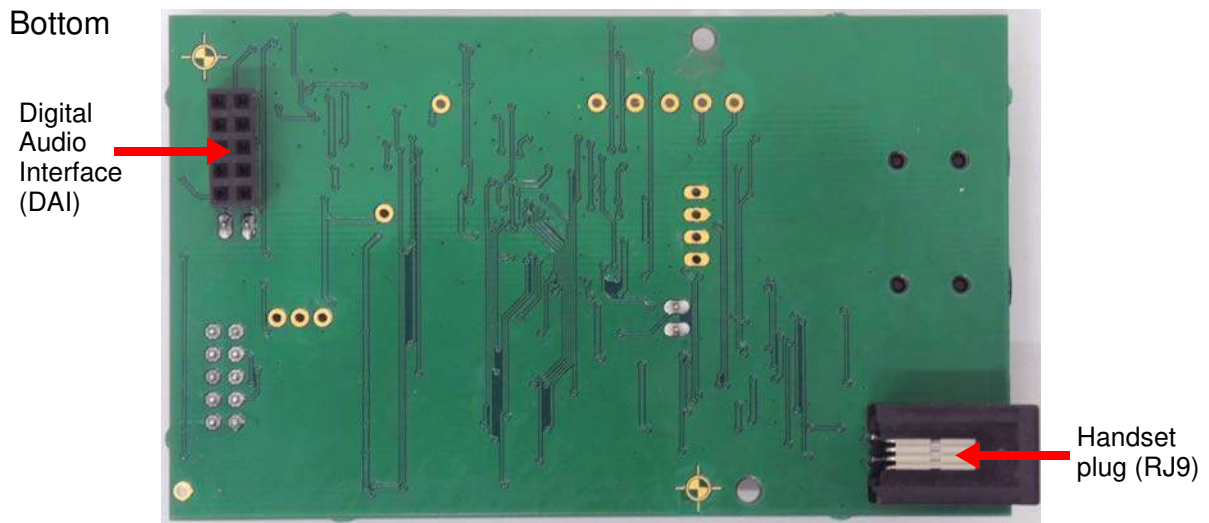


Figure 2: Audio Codec Adapter bottom view

Note: Only one out of Handset and Headset interface can be connected and operated at a time.

2.1.1.1 DIP Switch Default Settings

The following figure shows the default settings for the DIP switch. These settings configure a basic slave mode that should straightforwardly work with most Cinterion® modules. For more information on how to set the DIP switch see [Section 3.1.1.4](#).

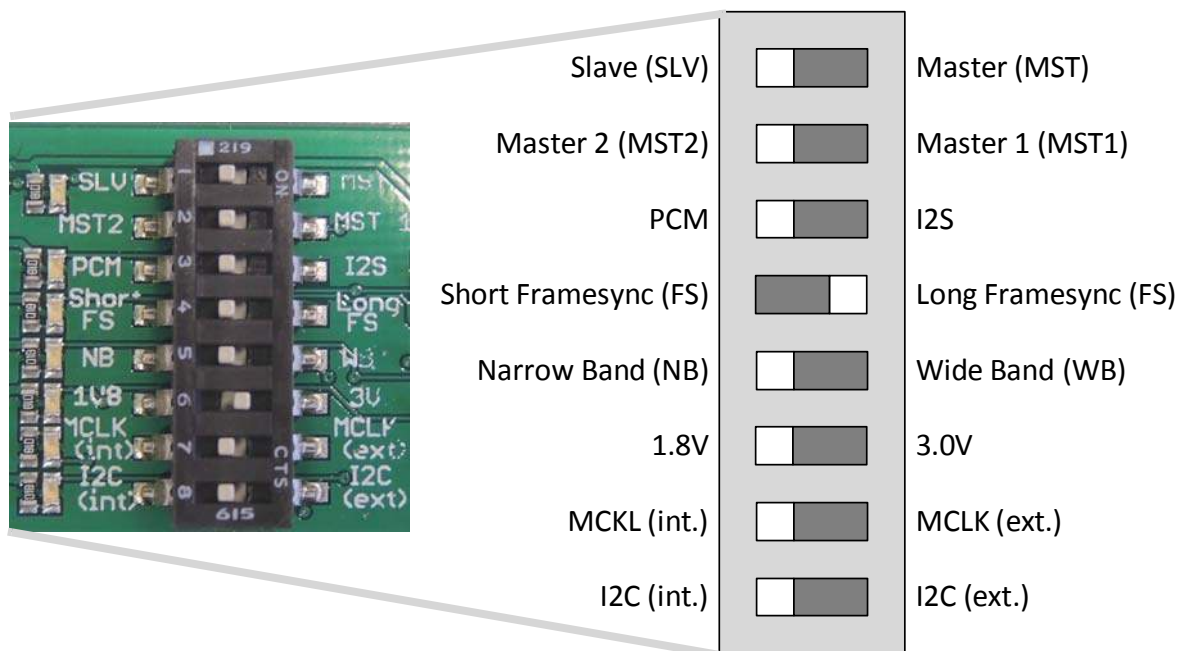


Figure 3: DIP switch defaults (e.g., ELS61-ER2)

2.1.2 Access

For update, configuration and programming purposes the Audio Codec Adapter can be accessed via the RS232 serial interface on its top side.

The RS232 interface on the Audio Codec Adapter can be connected to a PC's serial port by using a crimped cable that links the 10-pin connector at the Audio Codec Adapter to a 9-pin connector at the PC. Such a crimped cable is shown in [Figure 4](#):

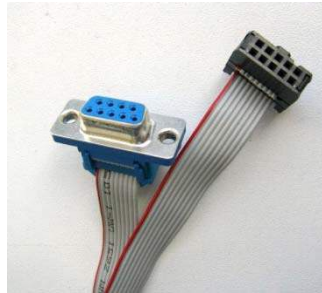


Figure 4: Crimped serial cable

Once connected to the PC, the Audio Codec Adapter should be available at a defined COM port for update, configuration, and programming as described in the below sections.

2.1.2.1 Update

Please complete the following steps to update the Audio Codec Adapter:

- Unzip the archive with the latest software for the Audio Codec Adapter to a directory on the PC. The following files should be available: CodecAdapterMAX.bin, Sloadhost.exe, and firmware_update_com_x.bat
- Call the batch file firmware_update_com_x.bat.
- Enter the COM port number the Audio Codec Adapter is connected at.
- Wait for a few seconds until the update is complete, and the adapter was reset, i.e., until the LEDs were restarted.

Having updated the Audio Codec Adapter's software, the adapter is ready to be employed for configuration and programming.

Note: Coming from the factory, the above described update procedure is also required to initialize the Audio Codec Adapter before it can be used for the first time, i.e., before any of the LEDs on the Audio Codec Adapter's top side will light to indicate the adapter's current status.

2.1.2.2 Configuration

There are two ways to configure the Audio Codec Adapter:

- Firstly, the hardware DIP switch can be used to configure the most important configuration settings. See [Section 2.1.1.1](#) and [Section 3.1.1](#) for default and other settings. This way to configure the adapter should normally be sufficient.
- Secondly, the RS232 interface can be used via its COM port to configure almost all configuration settings. This type of configuration can be only done while in the so-called Terminal mode.

To reach the Terminal mode, the following steps should be completed:

- Below settings must be specified for a terminal program:
 - Baudrate: 115.2 kbits/s
 - Format: 8N1
 - Flow Control: Hardware flow control must be disabled. This means DTR must be set inactive (off or low level), and RTS must be set inactive (off or low level).
- Now, the RTS line must be reset to enable the Terminal mode, i.e. the RTS line has to be activated and deactivated again.



Having enabled the Terminal mode the following commands are available to configure the Audio Codec Adapter.

Table 1: Terminal mode commands

Command	Description/Remark
? or Help	Displays short help
Status or Mode	Displays the current status, and measures BCLK and FSC frequencies
Master or Slave	Selects master or slave mode from the adapter's point of view. In slave mode FSC is set automatically in case 8KHz or 16KHz were measured.
Interface=1.8V/3.0V	Sets interface voltage level. 1.8V required for DSBmini (with Starter Kit B60).
MCLK=int/ext/bclk	Selects MCLK (mainly for ext. MCLK mode; effective in master mode)
BCLK=256/512/1024/2048	Sets the BCLK to the selected value (kHz; effective in master mode)
FSC=short/long/l2S FSC=8kHz/16kHz	1. Sets the FSC to the selected value: Short, long, or l2S 2. Sets the FSC to the selected value: 8kHz=NB or 16kHz=WB
Vol=-3dB	Speaker, headphone attenuation in +-1dB steps
Micvol=28dB	Microphone volume in 1dB steps
Boost=20dB	Displays and sets microphone preamplifier in dB (20dB or 30dB)
Reg	Lists all audio codec registers
Reg 0x06	Lists audio codec register 6, <8 bits>
Reg7=0x55	Sets audio codec register to 0x55
DTMF=seq<n> local/netw	Sends a DTMF sequence to the headset (<local>) or module (<netw>). <n> can be 1, 2, 3 or 4, each with a predefined DTMF sequence variant. See also Section 3.1.2.3 .

Table 1: Terminal mode commands

Command	Description/Remark
SINUS=<Hz>, <dur> local/netw	Generates a sine wave to the headset (<local>) or module (<netw>). See also Section 3.1.2.3 .
DSP=detect	Detects audio frequencies in local input signals. See also Section 3.1.2.4 .
Reset	Resets audio codec adapter

Commands are case insensitive, blanks are ignored.

Commands without "=" are typically status requests (i.e., READ commands), and commands with "=" are assignments or settings (i.e., WRITE commands).

Commands must be terminated with at least "\r" = 0x0D (under Windows). End of line is entered as "\r\n" = 0x0D, 0X0A (under Windows).

2.1.3 Block Diagram

Figure 5 shows a block diagram of the Audio Codec Adapter. For further details please refer to the schematics given in [chapter 5](#).

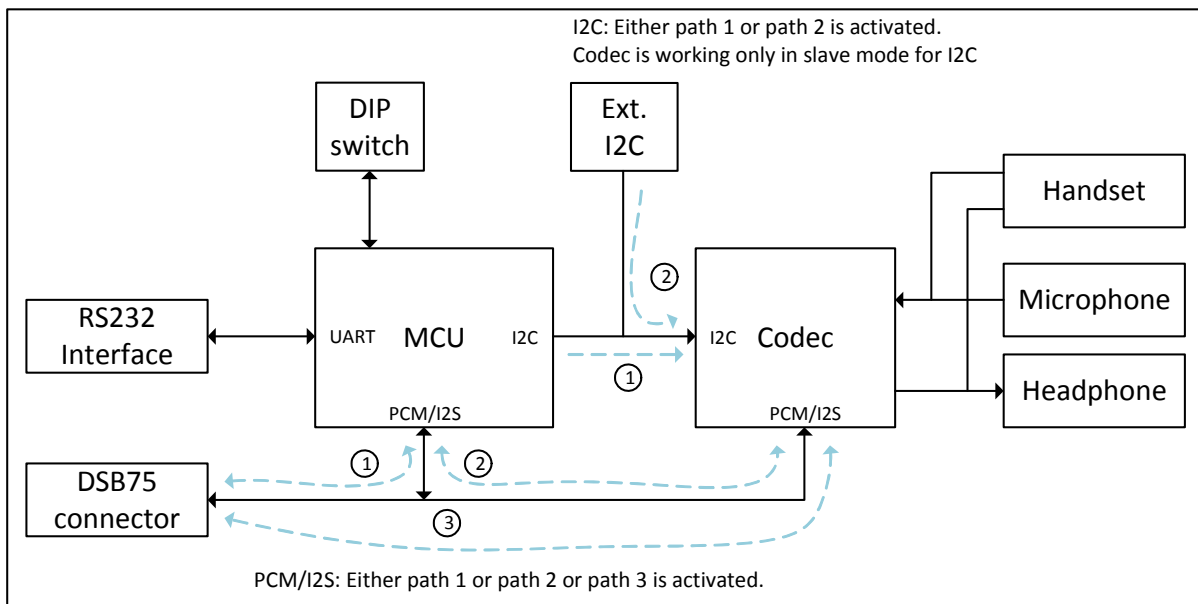


Figure 5: Block diagram

2.2 NAU8822 Audio Codec Adapter

The NAU8822 Audio Codec Adapter is based on the NAU8822 stereo audio voice codec from Nuvoton. The adapter board can be plugged to the DSB75 (or DSBmini) in order to be used with a variety of Cinterion® modules (see [Section 1.2](#)), and supports the following features:

The Audio Codec Adapter supports the following features:

- Directly connects to DSB75 (or Mini-DSB)
- Most important configuration settings are accessible via DIP switch
- Almost all configuration settings are accessible via RS232 serial interface
- PCM/I2S support
- BCLK = 128, 256, 512, 2048kHz
- Master/Slave mode
- 16kHz WB/8kHz NB mode
- Handset/Headphone support
- Short/long FSYNC
- Sending DTMF sound sequences on analogue or on digital (long FSYNC Intel mode)
- Recognizing beep sounds by means of FFT on analogue or on digital (long FSYNC Intel mode)

2.2.1 Interfaces

Figure 6 and Figure 7 show various interfaces on the top and bottom sides of the Audio Codec Adapter.

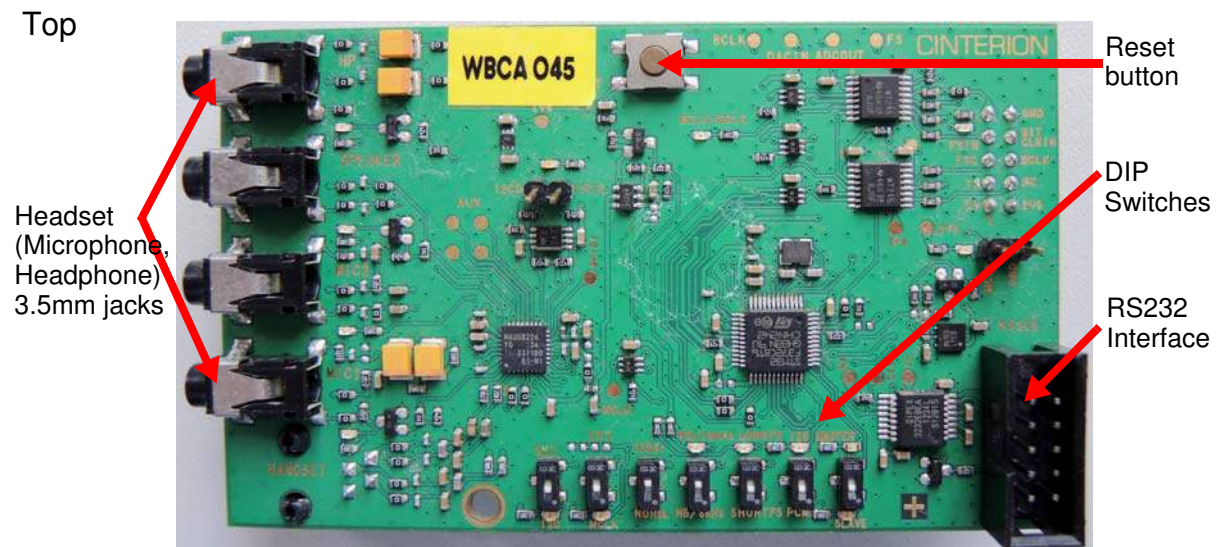


Figure 6: Audio Codec Adapter top view

Bottom

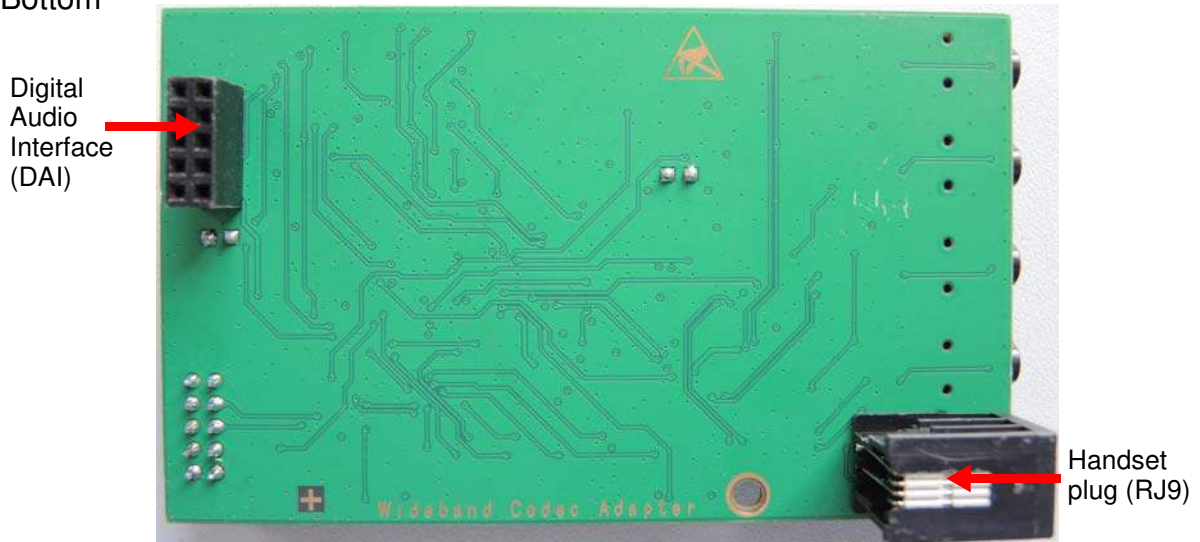


Figure 7: Audio Codec Adapter bottom view

Note: Only one out of Handset and Headset interface can be connected and operated at a time.

2.2.1.1 DIP Switch Default Settings

The following figure shows the default settings for the DIP switch. These settings configure a basic slave mode that should straightforwardly work with most Cinterion® modules. For more information on how to set the DIP switch see [Section 3.1.1.4](#).

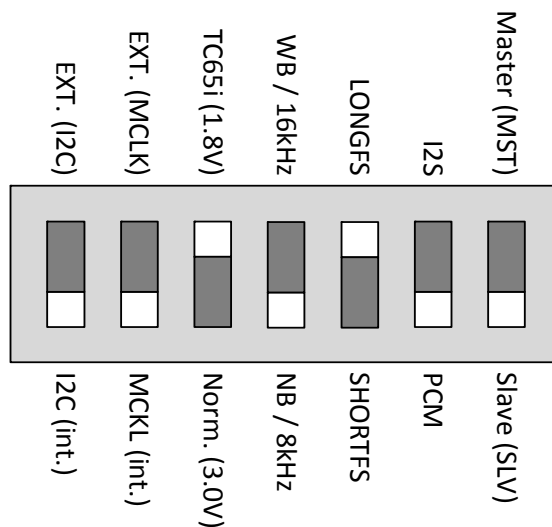


Figure 8: DIP switch defaults (e.g., ELS61-ER2)

2.2.2 Access

For update, configuration and programming purposes the Audio Codec Adapter can be accessed via the RS232 serial interface on its top side.

The RS232 interface on the Audio Codec Adapter can be connected to a PC's serial port by using a crimped cable that links the 10-pin connector at the Audio Codec Adapter to a 9-pin connector at the PC. Such a crimped cable is shown in [Figure 4](#):

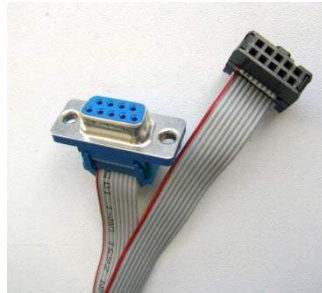


Figure 9: Crimped serial cable

Once connected to the PC, the Audio Codec Adapter should be available at a defined COM port for update, configuration, and programming as described in the below sections.

2.2.2.1 Update

Please complete the following steps to update the Audio Codec Adapter:

- Unzip the archive with the latest software for the Audio Codec Adapter to a directory on the PC. The following files should be available: CodecAdapterMAX.bin, Sloadhost.exe, and firmware_update_com_x.bat
- Call the batch file firmware_update_com_x.bat.
- Enter the COM port number the Audio Codec Adapter is connected at.
- Wait for a few seconds until the update is complete, and the adapter was reset, i.e., until the LEDs were restarted.

Having updated the Audio Codec Adapter's software, the adapter is ready to be employed for configuration and programming.

Note: Coming from the factory, the above described update procedure is also required to initialize the Audio Codec Adapter before it can be used for the first time, i.e., before any of the LEDs on the Audio Codec Adapter's top side will light to indicate the adapter's current status.

2.2.2.2 Configuration

There are two ways to configure the Audio Codec Adapter:

- Firstly, the hardware DIP switch can be used to configure the most important configuration settings. See [Section 2.1.1.1](#) and [Section 3.1.1](#) for default and other settings. This way to configure the adapter should normally be sufficient.
- Secondly, the RS232 interface can be used via its COM port to configure almost all configuration settings. This type of configuration can be only done while in the so-called Terminal mode.

To reach the Terminal mode, the following steps should be completed:

- Below settings must be specified for a terminal program:
 - Baudrate: 115.2 kbits/s
 - Format: 8N1
 - Flow Control: Hardware flow control must be disabled. This means DTR must be set inactive (off or low level), and RTS must be set inactive (off or low level).
- Now, the RTS line must be reset to enable the Terminal mode, i.e. the RTS line has to be activated and deactivated again.



Having enabled the Terminal mode the following commands are available to configure the Audio Codec Adapter.

Table 2: Terminal mode commands

Command	Description/Remark
? or Help	Displays short help
Status or Mode	Displays the current status, and measures BCLK and FSC frequencies
Master or Slave	Selects master or slave mode from the adapter's point of view. In slave mode FSC is set automatically in case 8KHz or 16KHz were measured.
Interface=1.8V/3.0V	Sets interface voltage level. 1.8V required for DSBmini (with Starter Kit B60).
MCLK=int/ext/bclk	Selects MCLK (mainly for ext. MCLK mode; effective in master mode)
BCLK=128/256/512/2048	Sets the BCLK to the selected value (kHz; effective in master mode)
FSC=short/long/l2S FSC=8kHz/16kHz	1. Sets the FSC to the selected value: Short, long, or l2S 2. Sets the FSC to the selected value: 8kHz=NB or 16kHz=WB
Vol=-3dB	Speaker, headphone attenuation in +-1dB steps
Micvol=28dB	Microphone volume in 1dB steps
Boost=20dB	Displays and sets microphone preamplifier in dB (20dB or 30dB)
Reg	Lists all audio codec registers
Reg 0x06	Lists audio codec register 6, <8 bits>
Reg7=0x55	Sets audio codec register to 0x55
DTMF=seq<n> local/netw	Sends a DTMF sequence to the headset (<local>) or module (<netw>). <n> can be 1, 2, 3 or 4, each with a predefined DTMF sequence variant. See also Section 3.1.2.3 .

Table 2: Terminal mode commands

Command	Description/Remark
SINUS=<Hz>, <dur> local/netw	Generates a sine wave to the headset (<local>) or module (<netw>). See also Section 3.1.2.3 .
DSP=detect	Detects audio frequencies in local input signals. See also Section 3.1.2.4 .
Reset	Resets audio codec adapter

Commands are case insensitive, blanks are ignored.

Commands without "=" are typically status requests (i.e., READ commands), and commands with "=" are assignments or settings (i.e., WRITE commands).

Commands must be terminated with at least "\r" = 0x0D (under Windows). End of line is entered as "\r\n" = 0x0D, 0X0A (under Windows).

2.2.3 Block Diagram

Figure 5 shows a block diagram of the Audio Codec Adapter. For further details please refer to the schematics given in [chapter 5](#).

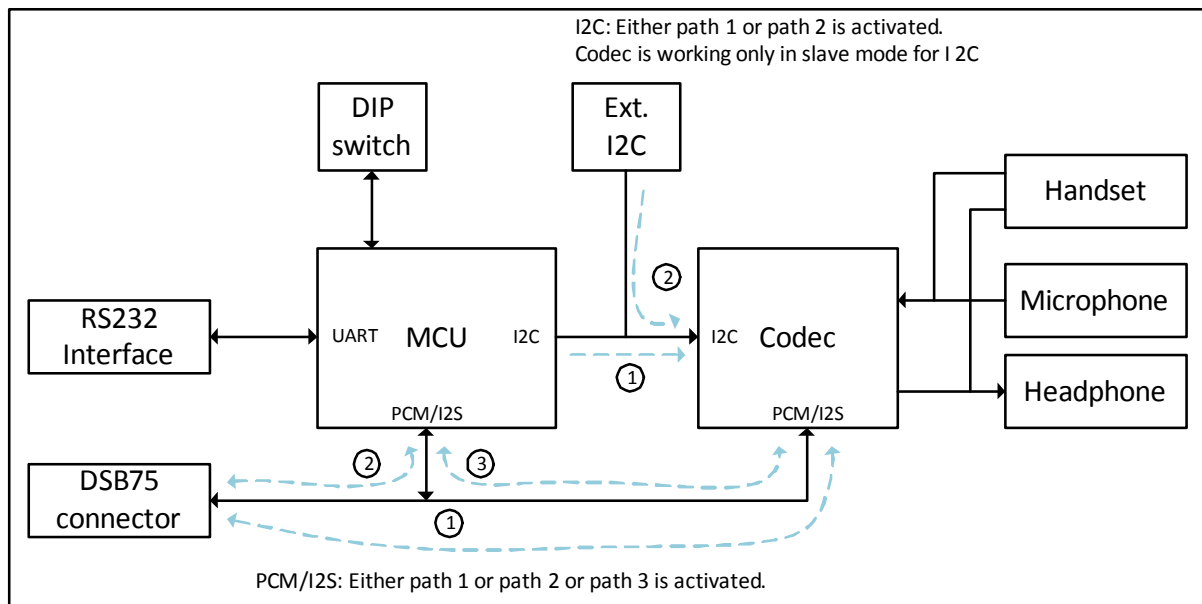


Figure 10: Block diagram

3 Audio Codec Adapter Setup

3.1 Assembly with DSB75 and AH6-DSB75 Adapter

The Audio Codec Adapter setup with DSB75 and AH6-DSB75 Adapter is based on the reference environment for development, test and approval purposes that is described in the Startup User Guide of a specific module. For detailed information on the startup environment please refer to [4]. The Audio Codec Adapter setup comprises the following hardware components: DSB75 Development Support Board, AH6-DSB75 Adapter, Evaluation module as well as Audio Codec Adapter. The below Figure 11 shows how these hardware components are assembled - with all jumper settings and switches in their default positions.

Crimped cable to access
Audio Codec Adapter
via RS232 interface

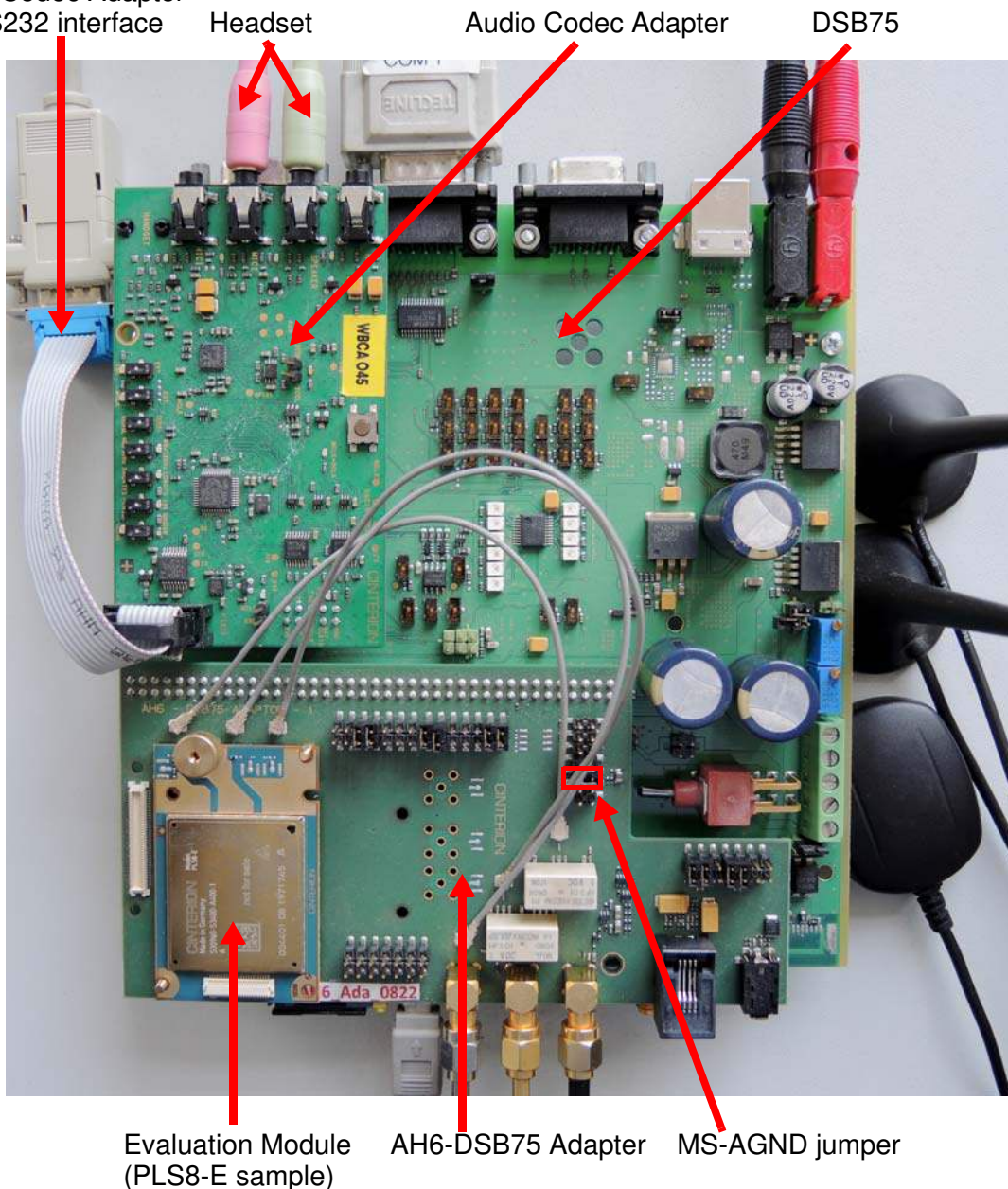


Figure 11: Audio codec adapter setup

3.1.1 Configuration

Depending on the module and the required audio scenario, the Audio Codec Adapter may have to be configured in different ways - and with different settings. As described in [Section 2.1.2.2](#), the Audio Codec Adapter can be configured in two ways - by means of a DIP switch or via RS232 COM port and a special Terminal mode. Both possibilities are referred to in the below sections that describe the possible configuration settings for various modules, including DIP switch settings and settings for the module via AT command.

3.1.1.1 Master/Slave

Master and slave configuration settings are seen from the point of view of the Audio Codec Adapter. For example, if a module is configured as master (see [\[1\]](#): AT^SAIC), the Audio Codec Adapter must be configured as slave. In order to avoid outputs driving outputs, this setting is important. **Note** that ELS31 modules cannot be configured as masters. As slaves they are therefore always the receiver of the master clock.

In master mode the bit clock (BCLK; see [Section 3.1.1.2](#)) and sampling rate (Framesync = FSC; see [Section 3.1.1.3](#)) are driven by the Audio Codec Adapter. The appropriate level converters are set to output. Also note that with MAX9860, BCLK and FSC can be set separately as Master or Slave, while NAU8822 can only be switched to either Master or Slave.

3.1.1.2 Bit Clock

In slave mode the bit clock is derived from the PCM/DAI interface lines. There is no need to adjust anything at the Audio Codec Adapter.

In master mode however, the bit clock is derived directly from the master clock (MCLK) provided by the Audio Codec Adapter's MCU.

While in master mode, the following command can be specified in order to configure the bit clock:

```
BCLK=128 (NAU8822 only) / 256 / 512 / 1024 (MAX9860 only) / 2048 (Default  
BCLK rating is 2048MHz)
```

See [Section 2.1.2.2](#) for details on how to access the Audio Codec Adapter via Terminal mode for configuration purposes.

3.1.1.3 Framesync

The sampling rate (=Framesync or word select) must be selected for master and slave modes. The settings can be either 8KHz (NB) or 16KHz (WB) frequency, and either long or short PCM frame or I2S framesync. The default FSC setting is 8KHz (NB=Narrow Band).

In SLAVE mode the FSC is set automatically once after a reset or by querying the FSC setting using the command FSC. In this case the FSC frequencies are measured at 8KHz and set to this value.

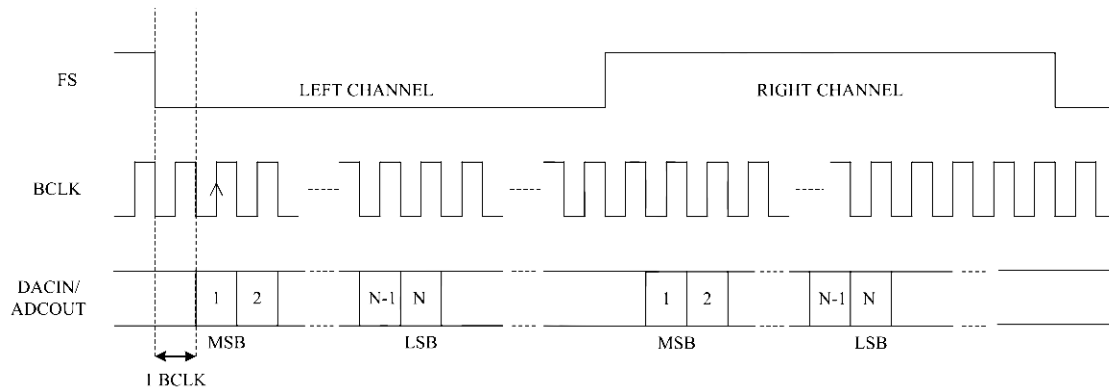
See [Section 2.1.2.2](#) for details on how to access the Audio Codec Adapter via Terminal mode for configuration purposes.

3.1 Assembly with DSB75 and AH6-DSB75 Adapter

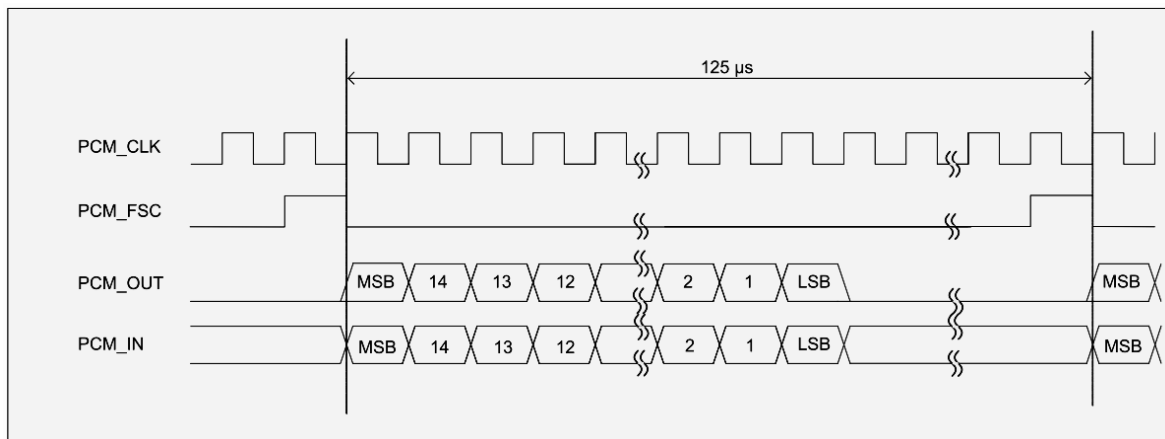
Please note that all PCM data and frame synchronization signals are written to the PCM bus on the rising clock edge and read on the falling edge, whereas all I2S data and frame synchronization signals are written to the I2S bus on the falling clock edge and read on the rising edge.

The following figures show the various timings for the Framesync modes. For a further measured sample please also refer to [Appendix A - DAI Signal Timings at Oscilloscope](#).

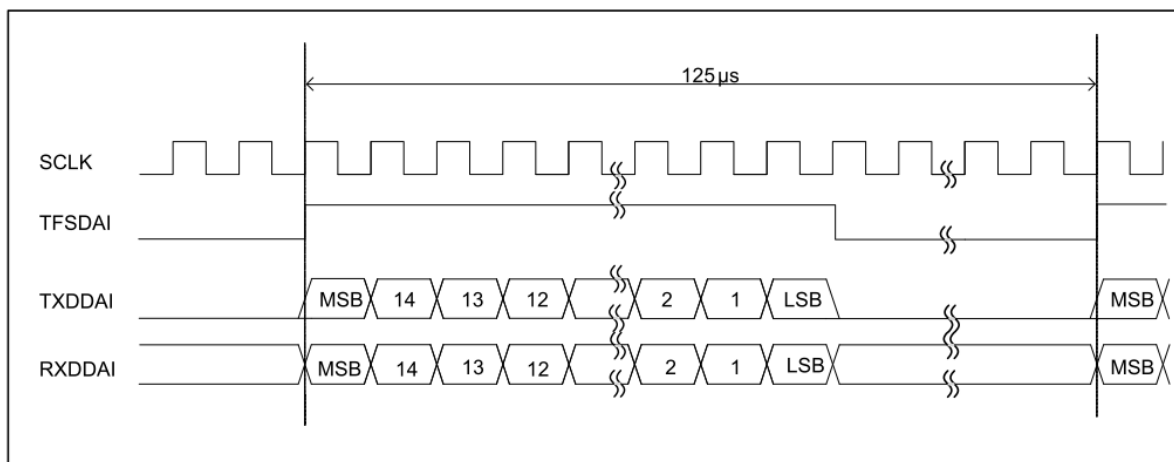
I2S mode³



PCM short FSYNC mode (NB)



PCM long FSYNC mode (NB)



³ Most modules use I2S left channel only. PHS8 can use the right channel for microphone array functionality.

3.1.1.4 DIP Switch and AT Command Settings

MAX9860

Table 4 list the default DIP switch and AT^SAIC command settings required to setup the MAX9860 for a specific (sample) module.

Table 3: MAX9860 DIP switch and AT command settings for selected products

Product	AT^SAIC setting	Audio Codec Adapter DIP switch settings								Comment
		Slave/Master	MST2/MST1	PCM/I2S	Short/Long	NB/WB	1.8V/3.0V	Int./Ext. MCLK	Int./Ext. I2C	
EHS5/6	-	0 ●	X	0 ●	1 ●	0 ●	1 ●	0 ●	0 ●	
BGS2/AGS2	SAIC=1,1	0 ●	X	0 ●	1 ●	0 ●	1 ●	0 ●	0 ●	
	SAIC=3,1	0 ●	X	1 ●	0 ●	0 ●	1 ●	0 ●	0 ●	
PHS8 R3	SAIC=1,1,1,2,0,0,1	0 ●	X	0 ●	0 ●	0 ●	1 ●	0 ●	0 ●	
	SAIC=1,1,1,1,0,0,1	0 ●	X	0 ●	0 ●	0 ●	1 ●	0 ●	0 ●	
	SAIC=1,1,1,2,1,0,1	1 ●	1 ●	0 ●	0 ●	0 ●	1 ●	0 ●	0 ●	Check master setting on the AH6-DSB75 Adapter. ¹
PLS8 R3	SAIC=1,1,1,3,0,0,1,0	0 ●	X	0 ●	0 ●	0 ●	1 ●	0 ●	0 ●	
	SAIC=1,1,1,2,1,0,1,1	1 ●	1 ●	0 ●	0 ●	1 ●	1 ●	0 ●	0 ●	Check master setting on the AH6-DSB75 Adapter. ¹
	SAIC=3,1,1,1,0,0,1,0	0 ●	X	1 ●	X	0 ●	1 ●	0 ●	0 ●	
	SAIC=3,1,1,1,0,0,1,1	0 ●	X	1 ●	X	1 ●	1 ●	0 ●	0 ●	
PLS8 R2	SAIC=3,1,1,0,0,0,1,0	0 ●	X	1 ●	X	0	1 ●	0 ●	0 ●	
ALAS3	SAIC=1,1,1,0,0,0,1,0	0 ●	X	0 ●	0 ●	0 ●	0 ●	0 ●	0 ●	DSB75 adapter for ALAS3 requires 1.8V levels. ²
ELS31	SAIC=1,1,1,4,1,1,1,1	1 ●	1 ●	0 ●	1 ●	1 ●	1 ●	0 ●	0 ●	Check master setting on the AH6-DSB75 Adapter. ¹

¹ The module (configuration) is a slave for clock as well as frame synchronization signals generated on the Audio Codec Adapter. **Note:** In this case the MS – AGND jumper on AH6-DSB75 Adapter (X305) must be removed for correct signal directions through the AH6-DSB75 Adapter’s voltage converters. The MS - AGND jumper is show in Figure 11.

² **Note:** With ALAS3 evaluation modules the ALAS6-DSB75 Adapter is required instead of the AH6-DSB75 Adapter. As this adapter supports 1.8V voltage levels, the corresponding level switch must be set to “1V8”.

Legend:

0: Switch set to the left position; 1: Switch set to right position; X: Switch setting irrelevant

"LEDs": The colors ●, ●, and ● indicate the color of the LEDs that light up at the Audio Codec Adapter.

NAU8822

Table 4 list the default DIP switch and AT^SAIC command settings required to setup the NAU8822 for a specific (sample) module.

Table 4: NAU8822 DIP switch and AT command settings for selected products

Product	AT^SAIC setting	Audio Codec Adapter DIP switch settings							Comment
		Master/ Slave	I2S/ PCM	Long/ Short	WB/ NB	3.0V/ 1.8V	Ext./Int. MCLK	Ext./Int. I2C	
EHS5/6	-	0 ●	0 ●	1 ●	0 ●	0 ●	0 ●	0 ●	
BGS2	SAIC=1,1	0 ●	0 ●	1 ●	0 ●	0 ●	0 ●	0 ●	
AGS2	SAIC=3,1	0 ●	1 ●	0 ●	0 ●	0 ●	0 ●	0 ●	
ELS61	SAIC=1,"", "", "", "", 0,0,1,0,0	0 ●	0 ●	1 ●	0 ●	0 ●	0 ●	0 ●	
ELS31	SAIC=1,1,1,2,1,1,1,0	1 ●	0 ●	1 ●	0 ●	0 ●	0 ●	0 ●	Check master setting on the AH6-DSB75 Adapter. ¹
PHS8 R3	SAIC=1,1,1,1,1,1,0,0	0 ●	0 ●	0 ●	0 ●	0 ●	0 ●	0 ●	
	SAIC=1,1,1,1,1,1,1,0	1 ●	0 ●	0 ●	0 ●	0 ●	0 ●	0 ●	Check master setting on the AH6-DSB75 Adapter. ¹
	SAIC=1,1,1,1,3,1,0	1 ●	0 ●	0 ●	0 ●	0 ●	0 ●	0 ●	Check master setting on the AH6-DSB75 Adapter. ¹
PLS8 R2	SAIC=1,1,1,3,0,0,1,1	0 ●	0 ●	0 ●	1 ●	0 ●	0 ●	0 ●	
	SAIC=3,1,1,0,0,0,1,0	0 ●	1 ●	0 ●	0 ●	0 ●	0 ●	0 ●	
ALAS5	SAIC=3,1,1,0,0,0,1,0	0 ●	1 ●	0 ●	0 ●	1 ●	0 ●	0 ●	

¹ The module (configuration) is a slave for clock as well as frame synchronization signals generated on the Audio Codec Adapter. **Note:** In this case the MS – AGND jumper on AH6-DSB75 Adapter (X305) must be removed for correct signal directions through the AH6-DSB75 Adapter’s voltage converters. The MS - AGND jumper is show in Figure 11.

Legend:

1: Switch set to up position

0: Switch set to the down position

"LEDs": The colors ●, ●, and ● indicate the color of the LEDs that light up at the Audio Codec Adapter.

3.1.2 Operation

Having completed the Audio Codec Adapter assembly and configuration described in the previous sections, the Audio Codec Adapter can be employed. This section describes the various possible ways to operate and test the Audio Codec Adapter.

3.1.2.1 Voice Calls

The Audio Codec Adapter processes outgoing and incoming voice calls. Before setting up a voice call, please ensure that the following prerequisites are completed: SIM pin is entered, module has the appropriate functionality level, module is attached to network, and the module is registered in IMS (for VoLTE).

Now, the following AT commands can be entered to process voice calls:

Outgoing calls can be started by:

ATD Mobile originated call to specified number

Incoming calls can be connected :

ATA Connect to incoming call

Call status can be verified by

AT+CLCC List of current calls

Calls can be terminated by the following AT command for the module:

AT+CHUP Hang up call

Please refer to your product specific AT Command Set ([1]) for details on the AT commands to complete the prerequisites and to process a voice call. Not all commands may be available for all modules.

3.1.2.2 Module Generated Tones

Some AT commands - not all of them necessarily available for every module - can be used to generate tones that can be processed by the Audio Codec Adapter:

AT^SNFG Generate Tone

AT^SRTC Ring Tone Configuration

AT+VTS DTMF and Tone Generation

Please refer to your product specific AT Command Set ([1]) for details on these AT commands to generate tones. Not all commands may be available for all modules.

3.1.2.3 Audio Codec Adapter Generated Tones

The Audio Codec Adapter itself can also be used to generate DTMF tones or Sine waves. If employing this feature there are the following restrictions:

- The Audio Codec Adapter can either generate or recognize tones (see [Section 3.1.2.4](#)), but not both at the same time.
- Only a few audio settings and BCLK/FSC frequency combinations are possible:
 - BCLK=256kHz
 - FSC=8kHz
 - Long/short frame sync
 - From/to local audio interface, from/to module's DAI interface
 - Module master and slave are supported

DTMF tones can be generated by the Audio Codec Adapter as follows:

```
DTMF=seq2 local    Play sequence #2; output on headphone interface
DTMF=seq3 netw    Play sequence #3; output to DAI interface
DTMF=off          Stop playing DTMF tones
```

The following DTMF sequences are defined and can be used with the "DTMF" command:

Seq1 (quick mode)	Tones: "1 2 3 4 5 6 7 8 9 0 * # A B C D", level: -6dBfs (full scale) Tone duration: 41msec, silence: 31msec No interrupts, no volume twist
Seq2 (twist)	Tones: "1 2 3 4 5 6 7 8 9 0 * # A B C D", level: -6dBfs (full scale) Tone duration: 60msec, silence: 50msec No interrupts, volume twist: 8dB
Seq3 (gaps)	Tones: "1 2 3 4 5 6 7 8 9 0 * # A B C D", level: -6dBfs (full scale) Tone duration: 41msec, silence: 31msec Interrupts: 10msec, no volume twist
Seq4 (low volume)	Tones: "1 2 3 4 5 6 7 8 9 0 * # A B C D", level: -46dBfs (full scale) Tone duration: 60msec, silence: 50msec No interrupts, volume twist: 8dB
Seq	This parameter can be used to configure DTMF sequence characteristics: DTMF=seq, dur:70, break:40, local (with dur: Duration 1...124msec

As no active call is needed for the "local" DTMF mode, this local feature can be used to generate analogue DTMF tones that can directly feed the headset's microphone.

Sine waves can be generated by the Audio Codec Adapter as follows:

```
SINUS = 750Hz, 124ms local    Play tone; output on headphone jack
SINUS = 800Hz, 1000ms netw   Play tone; output to DAI interface
SINUS=off                   Stop playing Sine wave
```

If the duration is greater than 124msec the frequency has a deviation of +/- 4Hz.

3.1.2.4 Tone Recognition

The Audio Codec Adapter itself is also able to recognize tones from the adapter's local headset interface (i.e., the microphone) even without an active call. This can be done by calling the following command in Terminal mode:

```
DSP=detect          Detect tones on local codec interface without call
> f=1000Hz, -38dB, 119msec detected
> f= 720Hz, -34dB, 119msec detected
DSP=off            Stop tone recognition
```

As no active call is required for this "local" mode, the feature can be used to detect tones that can directly come from the headset's microphone.

The tone recognition is limited to 500msec tone duration. Longer tones are recognized as 2 or more tones.

Note: The Audio Codec Adapter can either generate or recognize tones, but not both at the same time.

4 Appendix A - DAI Signal Timings at Oscilloscope

DAI signal timings as shown in [Section 3.1.1.3](#) can be visualized using an oscilloscope. The oscilloscope’s probe heads should be attached to the appropriate signal pads at the Audio Codec Adapter’s top side as shown below in [Figure 12](#) (Din corresponds to a module’s PCM_IN or RXDDAI signal, Dout to PCM_OUT or TXDDAI, FSC to PCM_FSC or TFSDAI, and BCLK to PCM_CLK or SCLK).

i

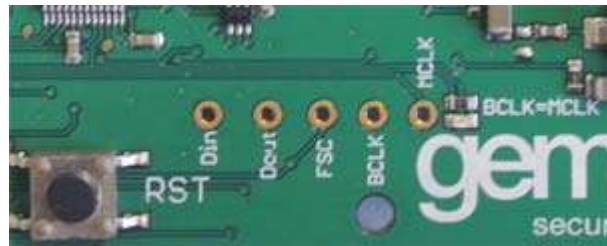


Figure 12: DAI signal lines at (MAX9860) Audio Codec Adapter’s top side

The following figures show sample measurement results for an ELS31 module’s DAI interface lines. [Figure 13](#) gives a short frame detail, while [Figure 14](#) shows a complete frame. Compare these to the figures given in [Section 3.1.1.3](#).

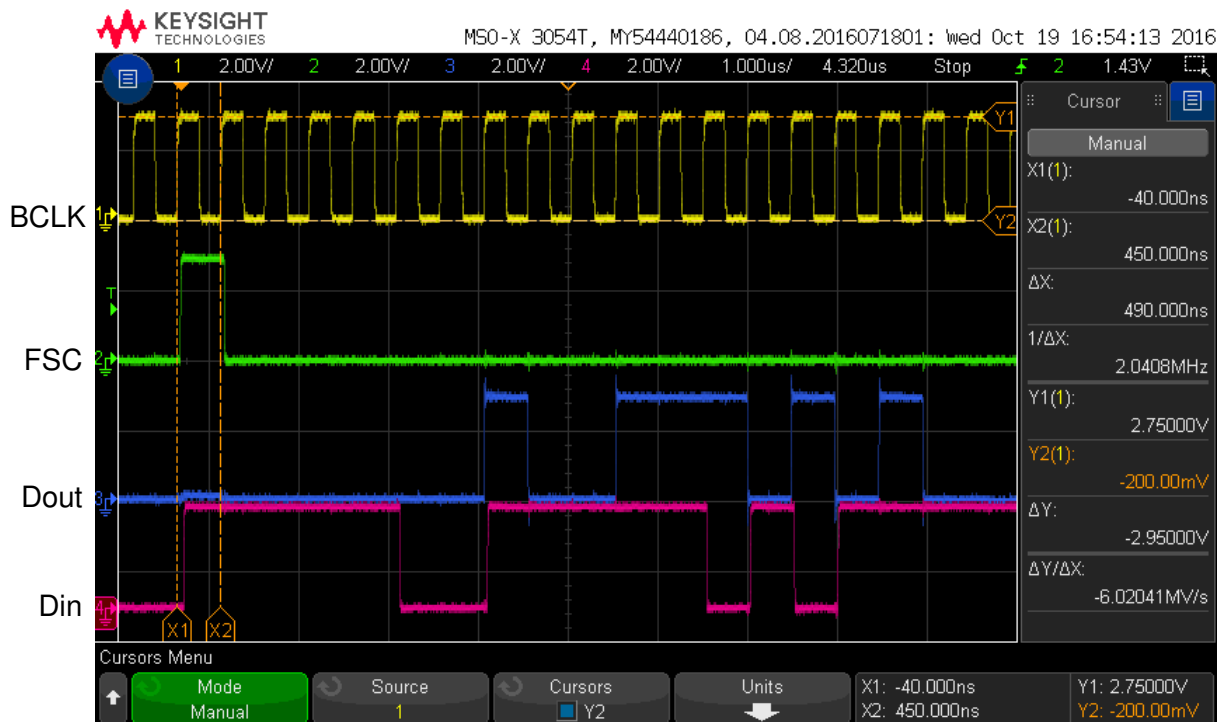


Figure 13: Timing sample (details) for DAI signal lines

4 Appendix A - DAI Signal Timings at Oscilloscope

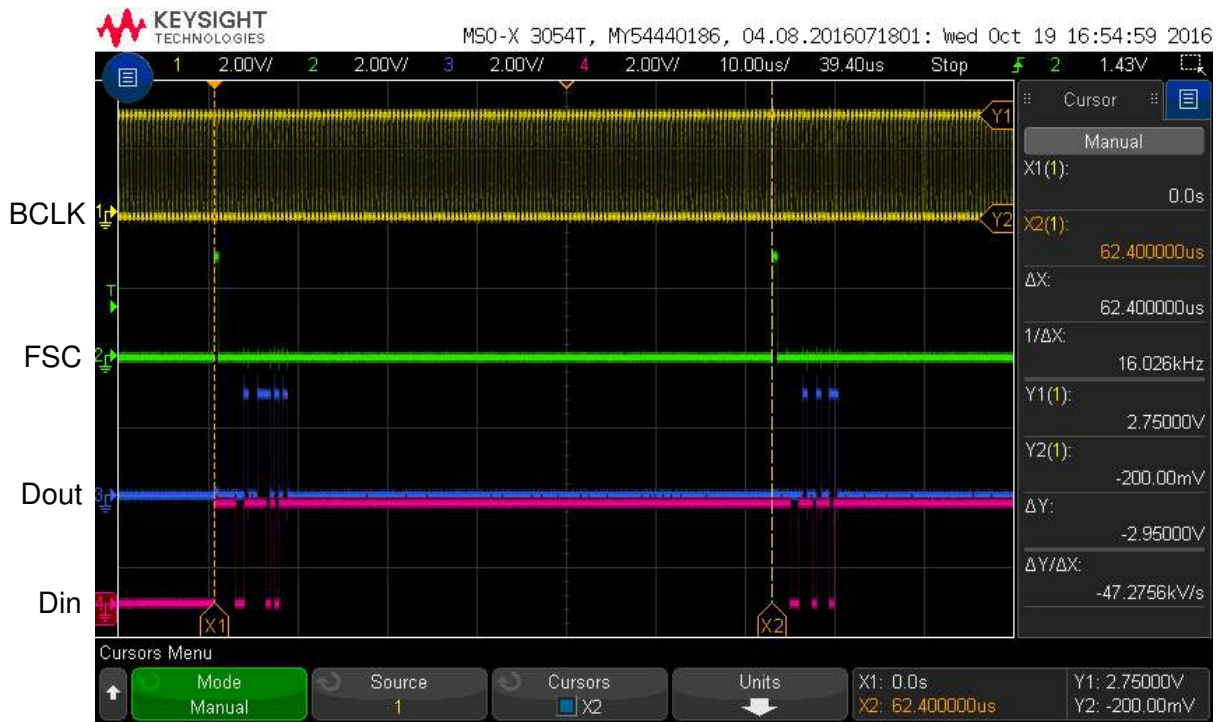


Figure 14: Timing sample (short frame - 62,5μs; wide band) for DAI signal lines

5 Appendix B - MAX9860 Codec Adapter Schematics

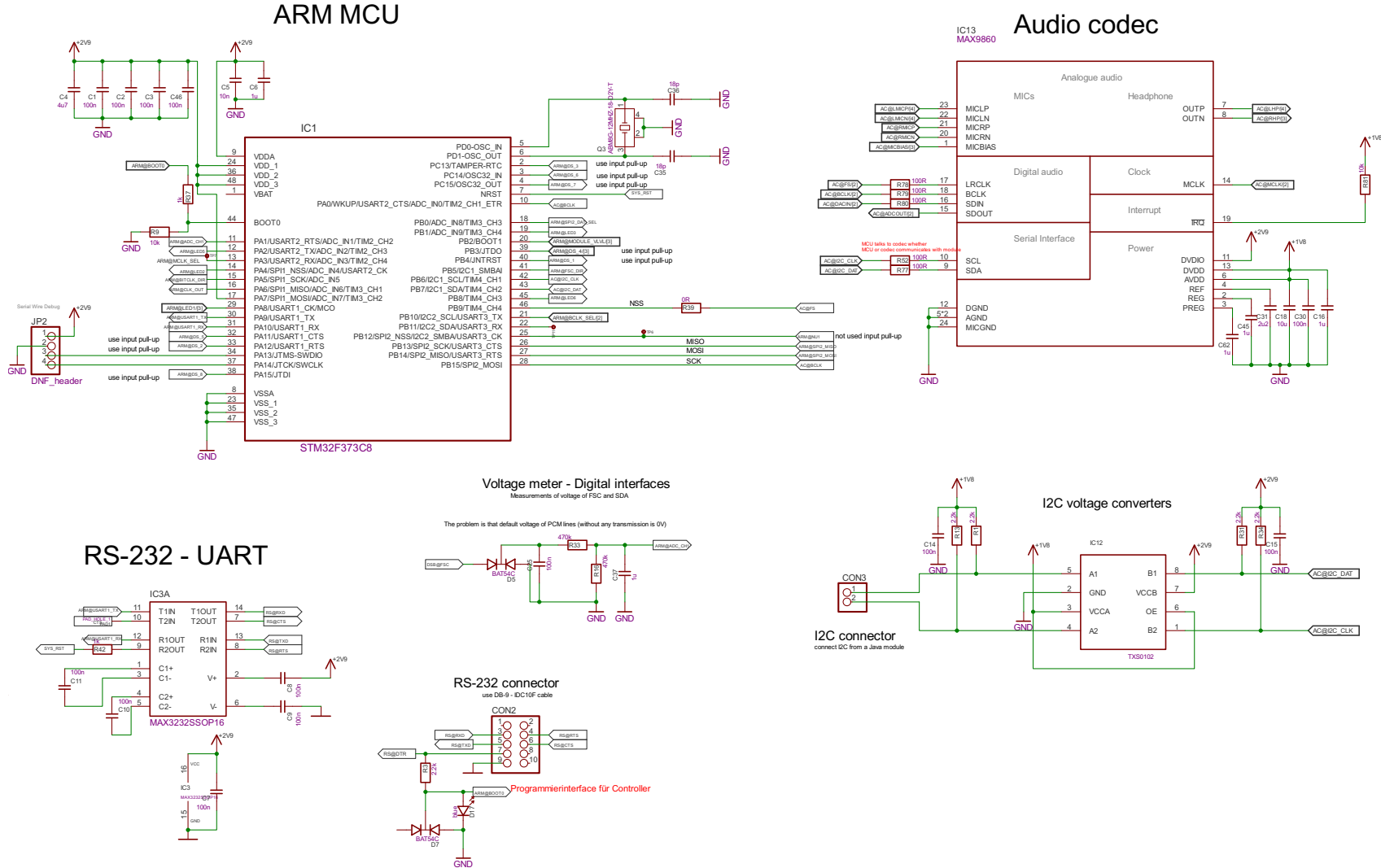


Figure 15: MAX9860 Codec adapter board schematics (sheet 1)

DIGITAL AUDIO PATH

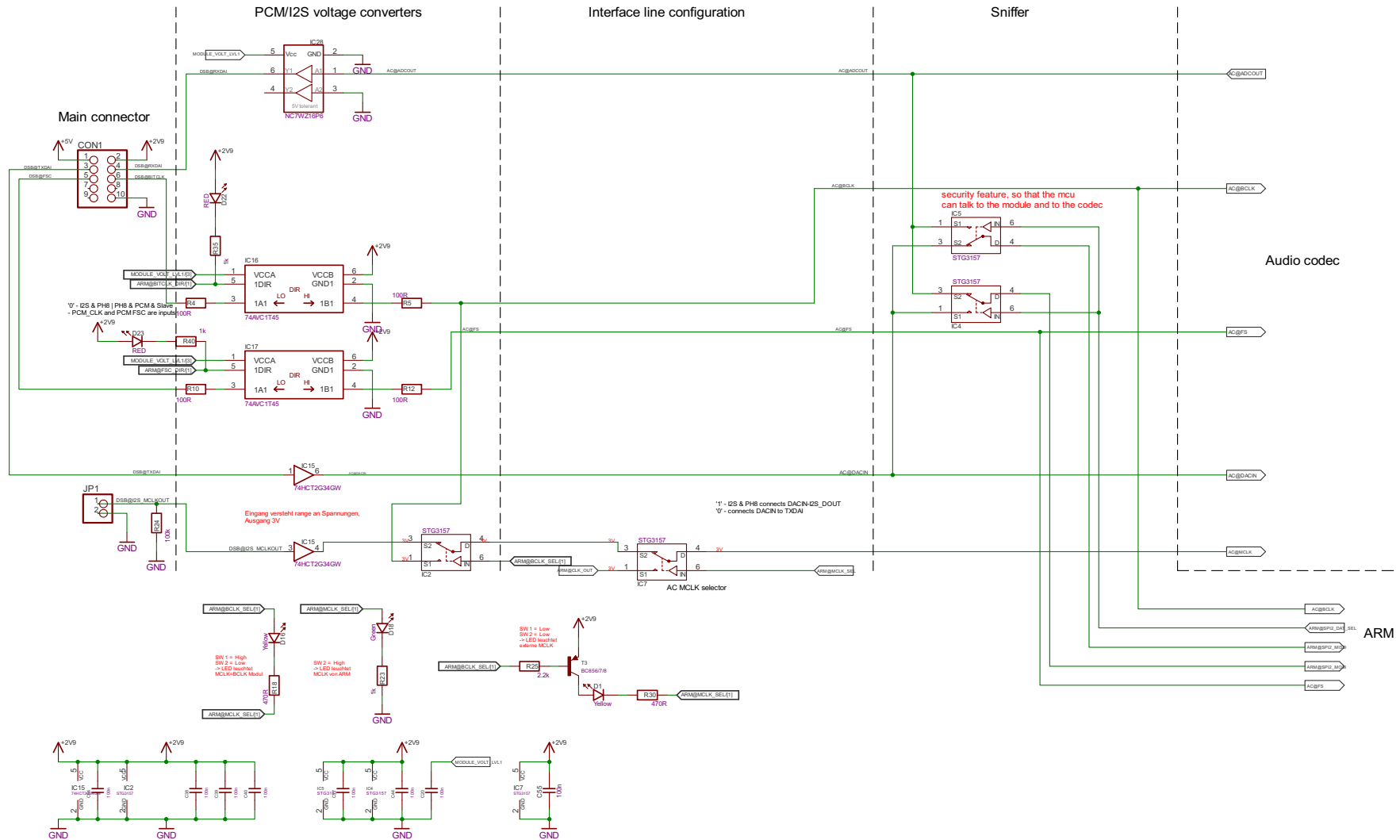


Figure 16: MAX9860 codec adapter board schematics (sheet 2)

5 Appendix B - MAX9860 Codec Adapter Schematics

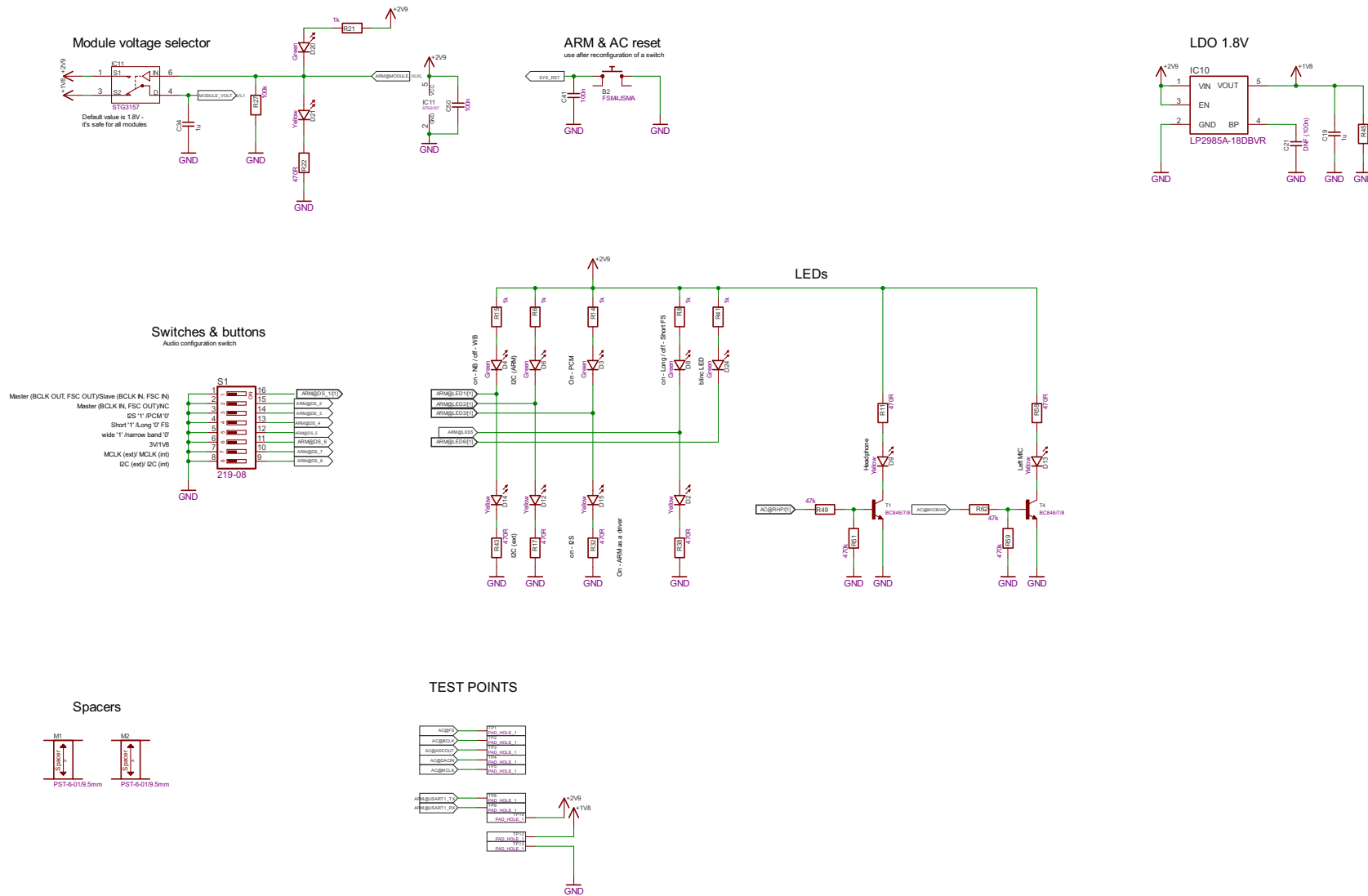


Figure 17: MAX9860 codec adapter board schematics (sheet 3)

Analog Audio

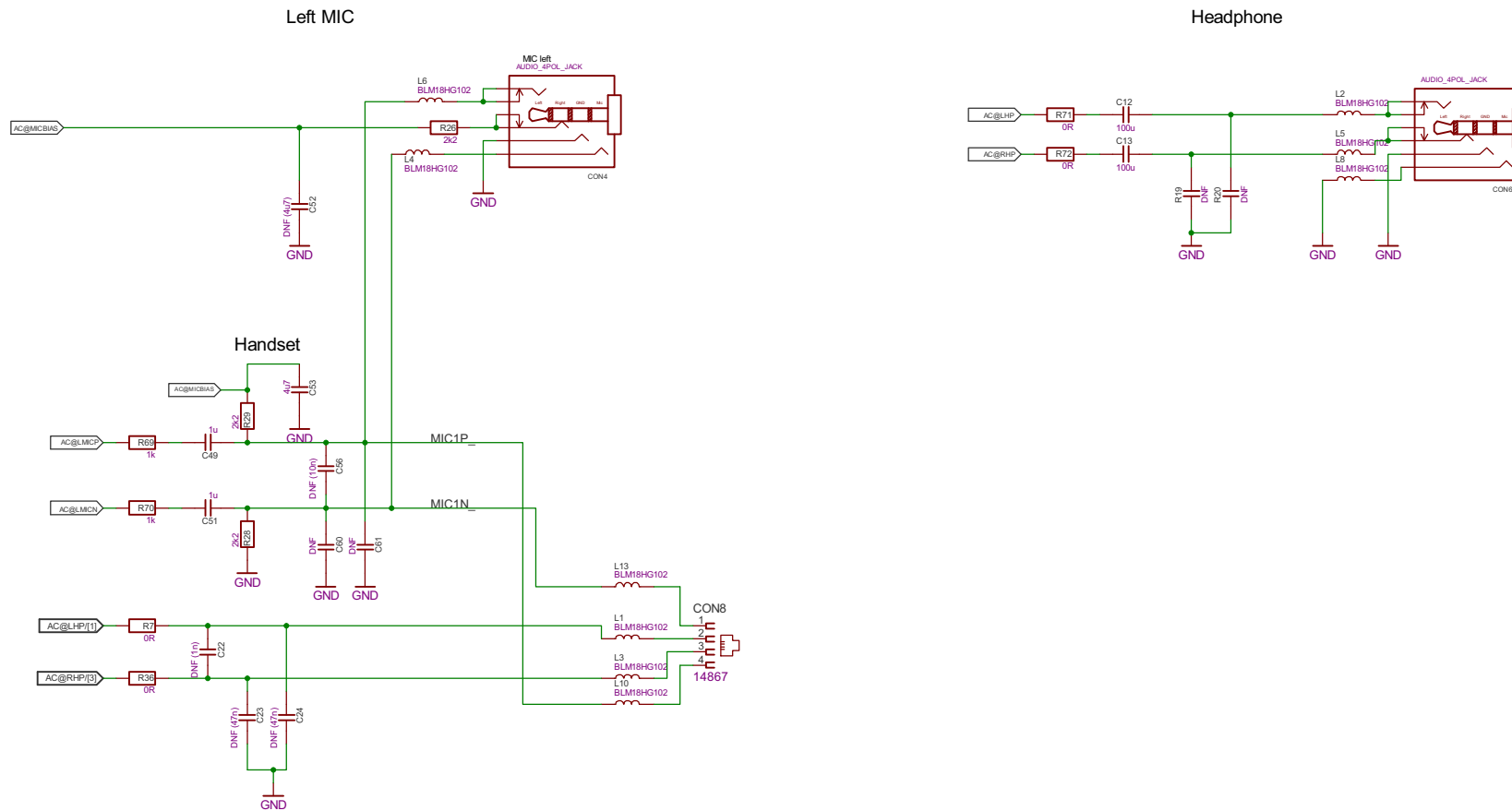


Figure 18: MAX9860 codec adapter board schematics (sheet 4)

6 Appendix C - NAU8822 Codec Adapter Schematics

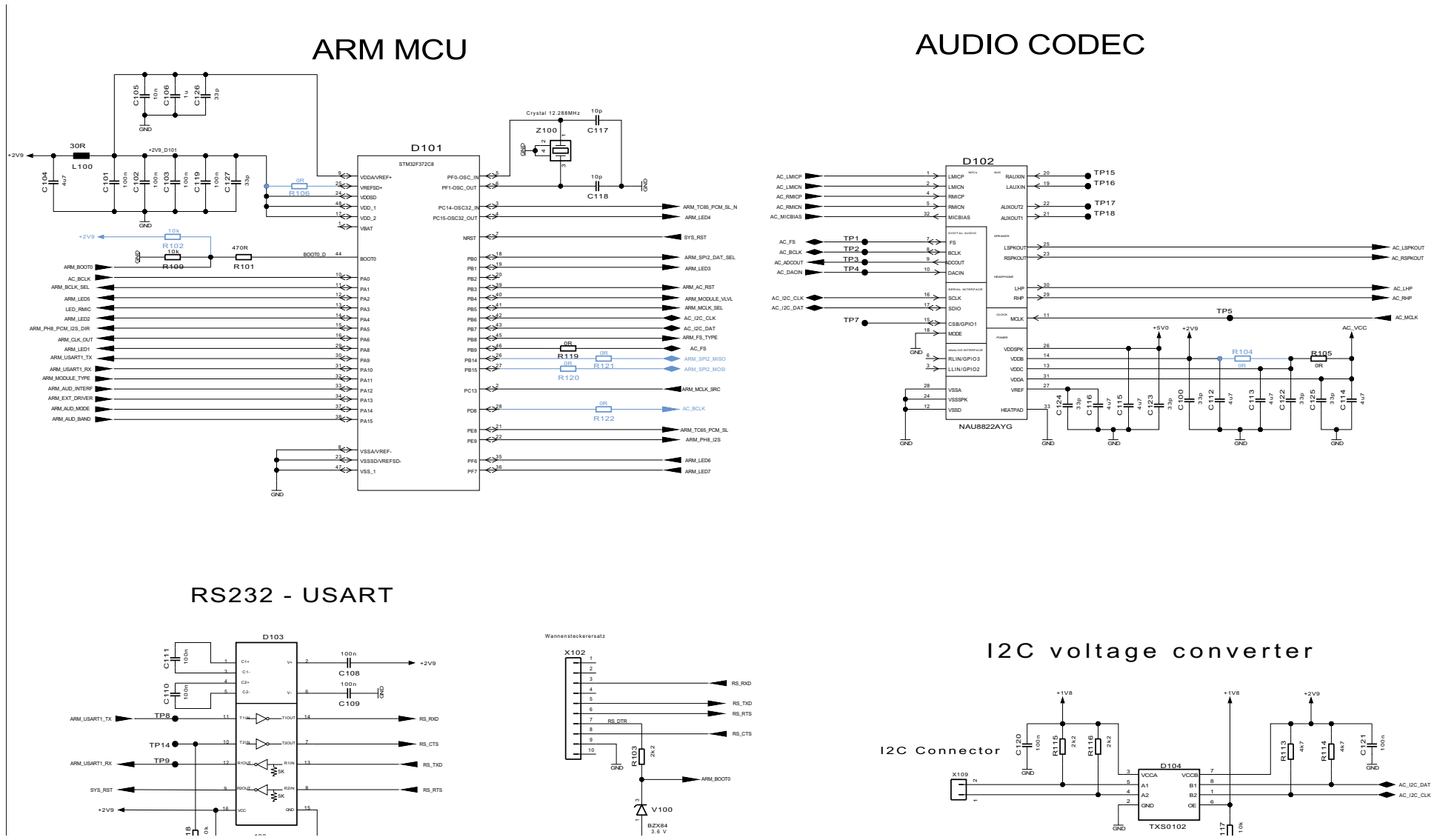


Figure 19: NAU8822 Codec adapter board schematics (sheet 1)

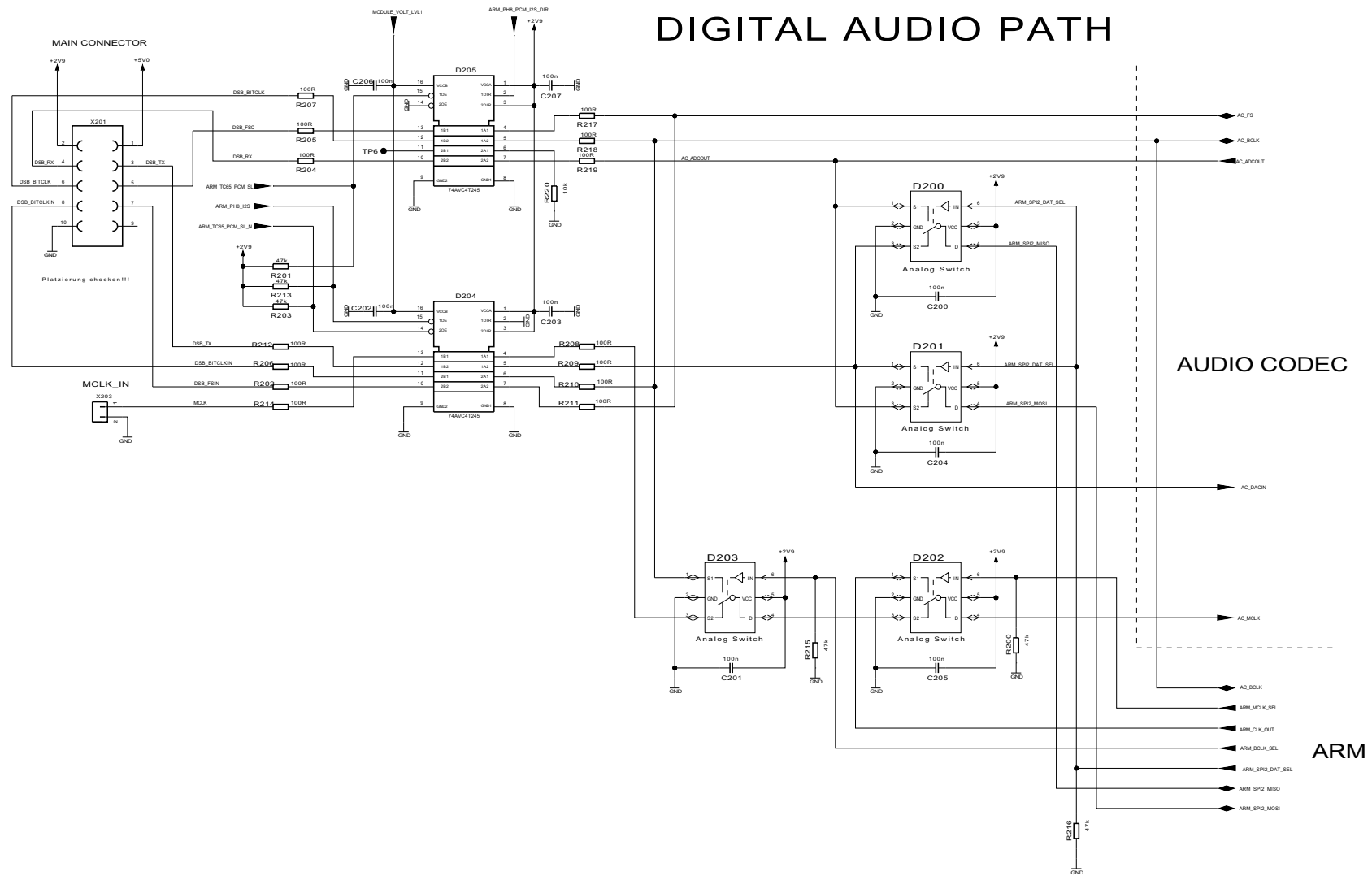


Figure 20: NAU8822 codec adapter board schematics (sheet 2)

6 Appendix C - NAU8822 Codec Adapter Schematics

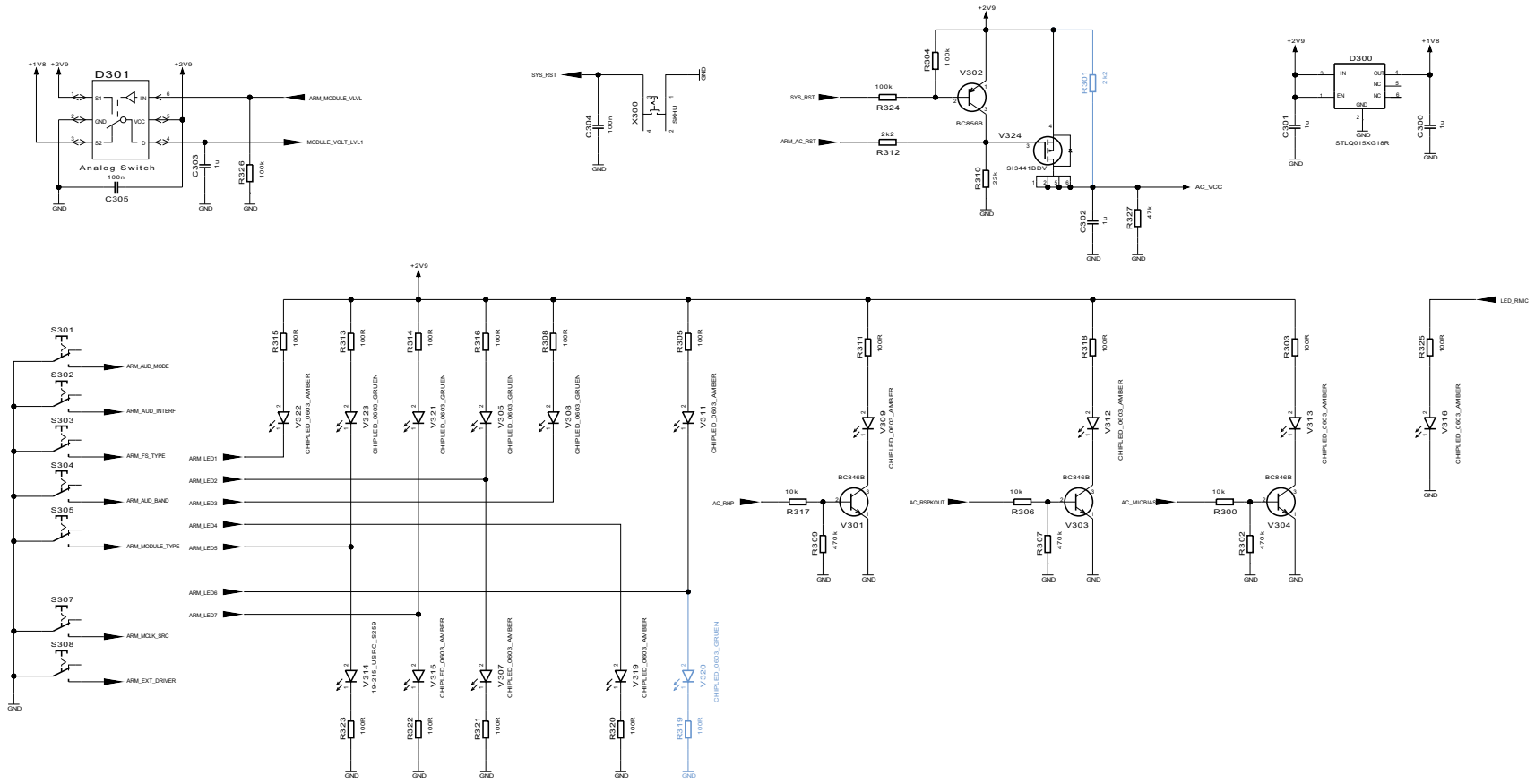


Figure 21: NAU8822 codec adapter board schematics (sheet 3)

ANALOG AUDIO

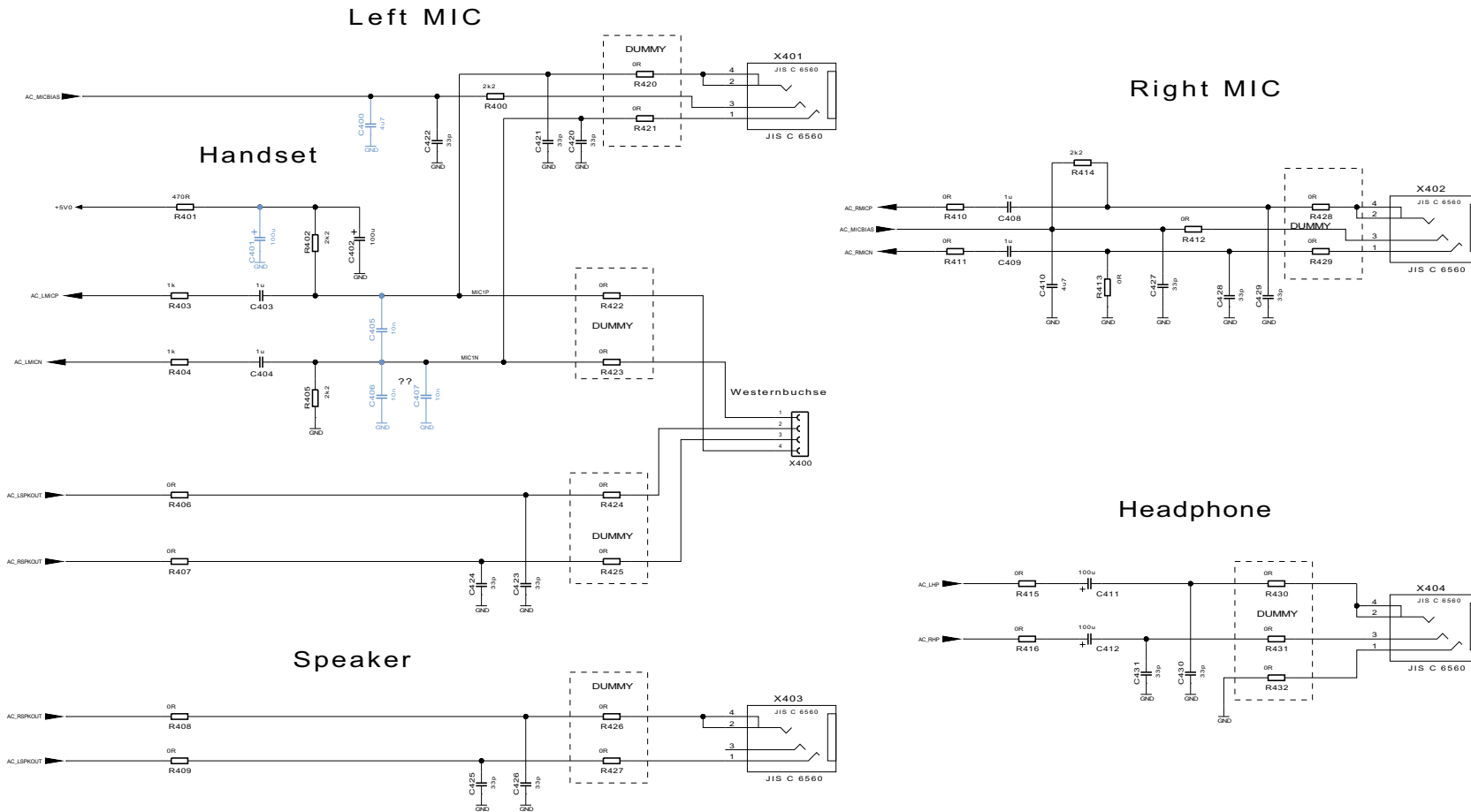


Figure 22: NAU8822 codec adapter board schematics (sheet 4)

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