POLYTECHNIC OF TURIN

The Department of Electronics and Telecommunications

(DET)

Master of Science Thesis in Electronic Engineering

(Embedded Systems)

Utilizing the A2B Protocol for Bi-Directional Audio Streaming in Automotive Sound Systems

Supervisor

Prof. Mihai Teodor LAZARESCU

Candidate

Payamreza POURREZA

October 2024

Abstract

In the modern automotive industry, where mass production and technological advancement drive fierce competition, manufacturers strive to differentiate their vehicles by enhancing the in-cabin experience. Among various improvements, the demand for superior in-car audio quality is particularly significant, leading to an increased number of audio components integrated within vehicle interiors. However, this growth introduces a series of challenges, including heightened wiring complexity, escalated production costs, added vehicle weight, and synchronization difficulties between audio components. These obstacles necessitate innovative solutions for efficient audio system integration that also uphold high performance and quality standards.

To address these challenges, the Automotive Audio Bus (A2B) protocol, developed by Analog Devices Inc., offers a robust solution for streamlined audio distribution in automotive environments. A2B enables both audio and control data transmission through a single unshielded twisted pair (UTP) cable, establishing a daisy-chain network linking a master device with multiple slave devices. In contrast to traditional audio distribution methods, which often suffer from excessive wiring, synchronization issues, latency, and fidelity loss, the A2B protocol significantly simplifies system architecture, reduces weight, and lowers costs. This thesis presents a system utilizing the A2B protocol to enable bi-directional, multi-stream audio distribution within a car audio system, facilitating the simultaneous transmission of multiple audio streams from master to slave devices and vice versa, thereby increasing system flexibility.

The hardware implementation of this research employs Analog Devices components, specifically two evaluation boards (EVAL-AD2428WD1BZ), chosen for their versatile communication interfaces, high configurability, and real-time processing capabilities. One board is designated as the master, while the other functions as a slave within the A2B network. Additionally, a USB interface board (EVAL-ADUSB2EBZ) is utilized to streamline A2B network configuration and data transfer between the master device and a PC, enabling easy control through USB input and I2C/SPI output. SigmaStudio software is used to design the system schematic and configure master and slave device registers, allowing for comprehensive and efficient system setup. The proposed system's functionality is validated by transmitting various audio streams between the master and slave devices, with captured outputs analyzed at each end to ensure accuracy and fidelity.

To validate the practicality of the proposed A2B audio system, a real-world test was conducted in a Tesla Model 3 to assess bidirectional audio streaming capabilities. The Tesla Model 3 was chosen due to its modern electrical architecture, which aligns well with the integration of advanced audio protocols. The A2B system was integrated into the vehicle, establishing a 3-node A2B network consisting of one master and two slave devices. During testing, multiple audio streams were transmitted simultaneously between the master and slave devices, covering both music playback and voice signals to emulate real in-car conditions. The results demonstrated successful low-latency, high-fidelity audio streaming in both directions, validating the A2B protocol's potential to enhance in-cabin audio experiences while addressing key challenges like wiring reduction and system scalability. This implementation underscores the feasibility of A2B as a scalable solution for future automotive audio architectures, particularly in electric vehicles where weight and power efficiency are critical.

Future developments will focus on expanding the system to support a larger number of slave devices, incorporating additional audio components to enable an even richer and more immersive audio environment. Advanced audio processing modules available in SigmaStudio will also be explored to enhance audio quality, enabling dynamic customization for optimal in-cabin sound. Additionally, this system could serve as a foundation for Autobus applications, which traditionally require extensive wiring to accommodate numerous audio sources and sinks. By leveraging the A2B protocol, future implementations could dramatically reduce wiring complexity while supporting high-performance, multi-stream audio, making it a versatile and scalable solution for automotive audio networks.

Keywords: Automotive Audio Bus (A2B), Sony/Philips Digital Interface Format (SPDIF), Inter-Integrated Circuit (I2C), Inter-Integrated Circuit Sound (I2S), Serial Peripheral Interface (SPI), Time-Division Multiplexing (TDM), Phase-Locked Loop (PLL), Pulse Density Modulation (PDM), Pulse Code Modulation (PCM), Analog-to-Digital Converter (ADC), Digital-to-Analog Converter (DAC), SigmaDSP Development Software by Analog Devices (SigmaStudio), Analog Devices Inc. (ADI), Unshielded Twisted Pair cable (UTP cable).

Acknowledgements

I wish to extend my heartfelt gratitude to my supervisor, *Professor Mihai Teodor Lazarescu*, whose invaluable guidance and mentorship have shaped this work from its inception to its completion. His expertise, encouragement, and constructive feedback have been instrumental at every stage of my thesis, for which I am sincerely grateful.

This thesis is the culmination of months of rigorous research, productive collaborations, and invaluable support from numerous individuals. I would like to express my sincere thanks to **2Electron**, and particularly to *Francesco Spagnolo*, the CEO of the company, for providing essential materials, insights, and resources that made this project possible.

I am deeply grateful to my parents for their constant love, encouragement, and support throughout my life, as well as to my wife, whose patience and unwavering support have been my cornerstone. Their belief in me has been a source of strength, motivating me to pursue my dreams and achieve my goals.

To my friends, whose encouragement and understanding have been unwavering, I owe a great debt of gratitude. Your support has been a source of strength and motivation throughout my journey. Thank you for helping make this experience both rewarding and memorable.

Turin, November 10th, 2024

Payamreza Pourreza

Table of Contents

List of Tables

List of Figures

Acronyms

A2B: Automotive Audio Bus

ADC: Analog-to-Digital Converter

ADI: Analog Devices Inc.

ASRC: Asynchronous Sample Rate Converter

AVB: Audio Video Bridging

BCLK: Bit Clock

DAC: Digital-to-Analog Converter

DSP: Digital Signal Processor

GPIO: General Purpose Input/Output

GUI: Graphical User Interface

I2C: Inter-Integrated Circuit

I2S: Inter-Integrated Sound

LRCLK: Left/Right Clock

MISO: Master In Slave Out

MOSI: Master Out Slave In

MOST: Media Oriented Systems Transport

PDM: Pulse Density Modulation

PCM: Pulse Code Modulation

PLL: Phase-Locked Loop

SCL: Serial Clock Line

SDA: Serial Data Line

SPI: Serial Peripheral Interface

SPDIF: Sony/Philips Digital Interface Format

SCF: Synch Control Frame

SRF: Synch Response Frame

SD: Serial Data

SCK: Serial Clock

CS: Chip Select

SCL: Serial Clock Line

SDA: Serial Data Line

SDI: Serial Data Input

SDO: Serial Data Output

SNR: Signal to Noise Ratio

SPL: Sound Pressure Level

TDM: Time-Division Multiplexing

UTP: Unshielded Twisted Pair cable

WS: Word Select

Chapter 1

Introduction

The evolution of car audio systems has paralleled the overall advancements in automotive technology, with an increasing emphasis on delivering high-quality audio experiences for vehicle occupants. Early car audio networks were relatively simple, relying on basic analog systems where audio signals were transmitted over individual copper wires to each speaker. This approach required extensive wiring, as each speaker needed a direct connection from the audio source, resulting in a bulky and complex wiring harness. The audio signal itself, being analog, was more susceptible to noise and interference, which could degrade the sound quality, especially in electrically noisy environments such as automobiles. Furthermore, adjusting audio settings for individual speakers was challenging because the control had to be implemented at the source or within the audio amplifier, leading to limited flexibility in managing sound distribution across the vehicle.

With the rise of digital technology, automotive audio systems began to incorporate digital audio transmission methods. Digital audio networks, such as MOST (Media Oriented Systems Transport) and Ethernet AVB (Audio Video Bridging), emerged as solutions for distributing high-quality audio within vehicles. MOST networks, for instance, used optical fiber or electrical cables to transmit digital audio and control signals [1]. This reduced noise issues associated with analog systems and allowed for multi-channel audio streaming. However, MOST networks were complex and costly to implement, requiring specific transceivers and connectors, and the optical fiber infrastructure increased the system's weight and installation difficulty [1]. Moreover, while the digital format provided better sound quality, it required significant bandwidth, especially for high-resolution audio streaming, and the system design demanded careful consideration of data synchronization to prevent latency issues.

Another digital approach involved using Ethernet AVB, which supported real-time audio and video data transmission over standard Ethernet cables. While this technology offered high bandwidth and supported a larger number of audio channels than MOST, it had its own limitations [2]. Ethernet AVB networks required specialized switches and software to manage data traffic and ensure synchronization [2]. These requirements increased system complexity and cost. Additionally, although the Ethernet cables reduced the total amount of wiring, they still contributed to the vehicle's weight, and careful planning was needed to optimize the routing of cables throughout the car [3]. In both MOST and Ethernet AVB networks, achieving low latency and precise synchronization between different audio components remained challenging, especially in complex audio setups with multiple speakers and microphones.

In contrast to these traditional audio distribution methods, the A2B (Automotive Audio Bus) protocol offers several advantages that address the aforementioned limitations. Unlike earlier systems, A2B allows audio and control data to be transmitted over a single unshielded twisted pair (UTP) cable in a daisy-chain configuration, significantly reducing the amount of wiring required, Figure 1.1. This streamlined approach minimizes weight and installation complexity while providing reliable digital audio transmission with low latency. Additionally, A2B's ability to support bi-directional audio streaming and multiple slave devices on the same network enables flexible audio system designs and facilitates better synchronization of audio signals across the vehicle. By addressing the challenges associated with previous car audio solutions, the A2B protocol presents a more efficient, cost-effective, and scalable approach for modern automotive audio systems [4].

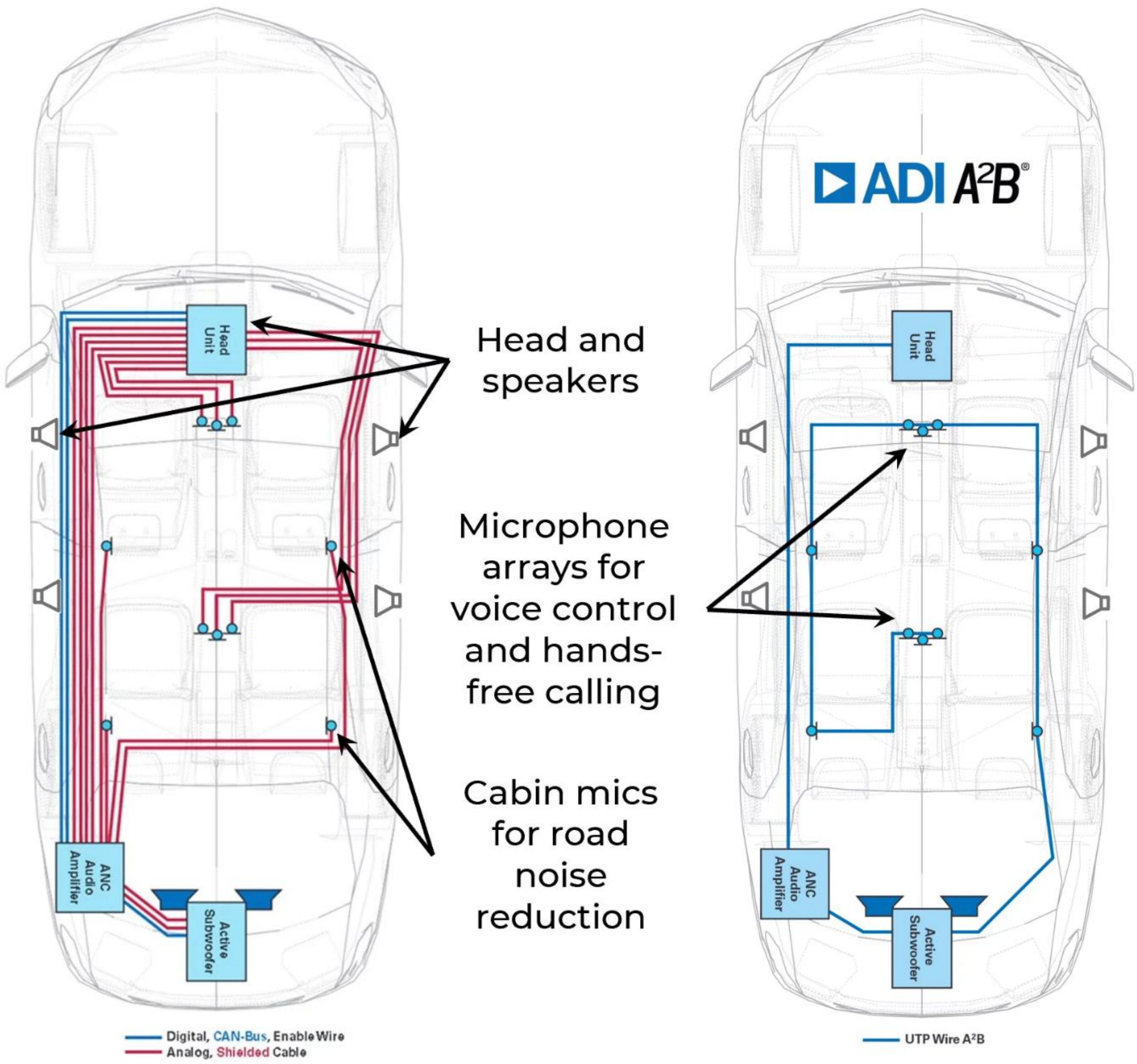

Figure 1.1: Traditional Audio System Wiring vs. Simplified A2B Daisy-Chain Configuration

Chapter 2

Literature Review

Various protocols and systems have been developed to address the specific challenges faced by automotive audio systems, including complexity in wiring, latency issues, and synchronization challenges. Among these, A2B (Automotive Audio Bus), MOST (Media Oriented Systems Transport), and AVB (Audio Video Bridging) have emerged as prominent technologies. Understanding the current landscape of automotive audio systems is key to identifying the uniqueness and innovation of the A2B-based solution presented in this thesis.

2.1 Automotive Audio Systems and Protocols Overview

A2B, developed by Analog Devices, aims to simplify wiring and improve audio synchronization through a single unshielded twisted pair (UTP) cable capable of transmitting both data and power in a daisy-chain configuration. In contrast, MOST and AVB focus on high bandwidth for advanced multimedia capabilities, with differing approaches to cost, complexity, and performance.

MOST has been extensively used for automotive infotainment systems, primarily due to its ability to handle multimedia streaming through either optical fibers or electrical cables. This approach results in high-quality audio and video transmission, but it comes with increased costs and complexity. AVB, meanwhile, provides a flexible solution for audio and video transmission over Ethernet, offering higher bandwidth at the cost of added system complexity due to the need for specialized switches and synchronization protocols. Both MOST and AVB are effective but expensive in terms of implementation and require significant bandwidth and power, thus complicating vehicle design.

In comparison, the A2B protocol simplifies the system by reducing the number of physical connections, which not only saves weight but also reduces cost and implementation complexity. This makes A2B particularly advantageous for applications where audio quality, latency, and efficient system integration are critical factors.

2.2 Quality, Latency, and Synchronization in Automotive Audio Systems

The performance of automotive audio systems is often judged based on three main parameters: quality, latency, and synchronization. Various studies have focused on measuring these aspects across different audio transmission protocols to establish industry benchmarks and identify areas needing improvement.

Latency is a key issue in automotive systems, especially when audio must be synchronized with visual cues or interactions within infotainment systems. Traditional systems, such as MOST, tend to suffer from higher latency due to the inherent design complexity of optical and electrical signaling. AVB, while capable of maintaining low latency, demands careful management of data streams and relies on AVB-capable switches to ensure timely data delivery. The A2B protocol, in contrast, offers deterministic latency with a fixed sample rate period, making it particularly attractive for systems where timing is crucial, such as voice-based applications or interactive infotainment.

Regarding audio quality, both MOST and AVB have demonstrated high-fidelity capabilities, suitable for high-resolution audio and complex entertainment environments. However, they require substantial bandwidth to support such capabilities. The A2B protocol provides comparable audio quality but does so in a more resource-efficient manner by leveraging lower data rates, which is particularly beneficial in scenarios where weight and power consumption are considerations, such as electric or compact vehicles.

Synchronization is another critical aspect in automotive audio systems. MOST and AVB networks require complex synchronization mechanisms to align data across multiple nodes, which can be challenging and costly. In comparison, A2B maintains synchronization using an embedded clock that helps in reducing synchronization discrepancies across nodes, facilitating seamless multi-point audio distribution without requiring additional hardware.

2.3 Comparative Analysis of A2B and Other Protocols

The following section offers a detailed comparison between A2B and other major automotive audio protocols, specifically MOST and AVB, highlighting the relative advantages and limitations of each approach.

A2B vs. MOST

MOST has been a preferred choice for multimedia applications due to its ability to provide high-quality streaming through dedicated optical or copper channels. It supports a large number of audio and video streams simultaneously, which makes it ideal for infotainment systems in high-end vehicles. However, its implementation is costly, with complex networking requirements and specialized components, making it impractical for cost-sensitive applications.

In contrast, A2B provides a more streamlined solution for audio distribution, primarily focusing on reducing wiring complexity and overall system cost. The daisy-chain architecture of A2B is simple, allowing for easier installation and maintenance. While MOST offers superior multimedia support, A2B excels in applications requiring efficient audio transmission with minimal infrastructure, particularly for audio-only systems or vehicles with constrained space and weight budgets.

A2B vs. AVB

AVB provides a highly scalable solution for transmitting high-bandwidth audio and video data using standard Ethernet infrastructure. Its ability to handle large numbers of audio channels makes it a suitable choice for vehicles that demand complex infotainment systems. However, its reliance on AVB-specific network switches and rigorous clock synchronization methods can introduce significant complexity.

A2B, on the other hand, targets use cases that require cost efficiency, low power consumption, and simpler integration. A2B's deterministic latency and straightforward clock synchronization make it advantageous for transmitting audio data in real-time applications, such as active noise control or voice assistance systems. While AVB may have advantages in high-bandwidth environments, A2B provides a more efficient and specialized solution for typical automotive audio needs, particularly where resources are limited.

2.4 Identified Gaps and Open Challenges

From the broader context of the literature, it is evident that several challenges remain unaddressed in the realm of automotive audio systems. Despite the improvements offered by A2B, MOST, and Ethernet AVB, gaps remain in areas such as power efficiency, cost of implementation, and the trade-off between scalability and complexity. Specifically, MOST remains limited by its cost and inflexibility, while Ethernet AVB, though highly scalable, is complex and not well suited for environments with stringent resource limitations.

The A2B protocol presents an appealing compromise between simplicity, efficiency, and performance; however, there is still scope for further development in areas such as increasing the number of slave nodes supported and optimizing audio quality under varying vehicle conditions. One critical gap is the need for enhanced support for multi-channel audio that integrates seamlessly with in-cabin systems while maintaining low power usage and minimal latency.

This thesis aims to address some of these challenges by demonstrating the application of the A2B protocol in a bi-directional, multi-stream automotive audio system. By optimizing the implementation, improving the system's scalability, and enhancing audio quality, this work seeks to bridge the identified gaps and push forward the capabilities of automotive audio technologies.

Chapter 3

Methodology

In this section, the methodology for implementing a bi-directional audio streaming system using the A2B (Automotive Audio Bus) protocol is presented. This methodology encompasses systematic problem-solving approaches, rigorous engineering practices, and a structured process to achieve a reproducible and reliable design.

3.1 Proposed Design

The proposed design addresses the main challenges in in-vehicle audio systems by using the A2B protocol to achieve low-latency, synchronized, and efficient bi-directional audio streaming. Traditional audio setups often face issues like excessive wiring, added weight, and signal degradation over long distances, while the A2B protocol enables streamlined data transfer through a single unshielded twisted pair (UTP) cable, forming a master-slave network.

In this configuration, the A2B master device is considered to be located near the car's main computer and is responsible for receiving digital audio signals via an SPDIF connection from the car's main computer, which serves as the primary audio source for the vehicle's speakers. The master device then transmits this digital audio to the slave device for further distribution.

On the slave side, which is considered to be positioned near the car amplifier, the downstream audio received from the master is outputted to the amplifier via either an analog output (e.g., through a 3.5mm jack) or a digital SPDIF output, enabling flexibility in connecting to different audio systems within the vehicle. In addition, the slave device transmits audio captured from microphones back to the master device, facilitating applications such as hands-free calling, voice command processing, and other in-car audio interactions, as shown in Figure 3.1. This design choice ensures that the system maintains high-quality, synchronized audio transmission and compatibility with both analog and digital audio sources across the vehicle.

3.2 Objective and Scope

The main objective of this thesis is to explore the design and implementation of an A2B (Automotive Audio Bus) network configuration tailored for bi-directional audio streaming within a car audio system. This includes integrating A2B evaluation board interfaces into the vehicle's audio system, enabling both analog and digital audio streams at the input and output levels. A central goal is to ensure seamless, synchronized bi-directional audio streaming between the Master and Slave devices. Additionally, this work aims to assess the system's feasibility by evaluating its performance in real-world automotive conditions, with extensive testing conducted to verify that audio streaming operates in real-time and maintains precise synchronization across the network.

The scope of this thesis encompasses implementing the A2B system through structured steps, resulting in a functional audio network prototype. Limitations and assumptions include focusing solely on in-vehicle audio systems rather than broader vehicle network integrations and assuming the availability of A2B-compatible components for testing. The study will also limit environmental testing to controlled lab conditions, with real-world testing confined to evaluating standard vehicle noise and interference levels.

3.3 Hardware Requirements

The hardware components are essential to the successful implementation of the A2B-based audio streaming system. Key hardware components selected for this system include two A2B evaluation boards (EVAL-AD2428WD1BZ) configured as the master and slave, and an additional evaluation board (EVAL-ADUSB2EBZ) for interfacing with a PC to configure the A2B network. This hardware setup was chosen for its compatibility with the A2B protocol, ease of configuration, and real-time processing capabilities.

- **Master and Slave Devices:** The EVAL-AD2428WD1BZ evaluation boards serve as the primary devices in the A2B network, with one board configured as the master and the other as the slave. These boards support high-quality audio transmission over the A2B protocol and provide versatile input/output options for peripheral devices such as microphones, speakers, and amplifiers. Their configurability and low-latency communication capabilities are integral to achieving synchronized audio streaming.
- **PC Interface for Configuration:** The EVAL-ADUSB2EBZ evaluation board enables a USB interface to facilitate A2B network configuration through a PC. It supports both I2C and SPI output, essential for programming the master device. The connection between the PC and the A2B master is crucial for deploying the configuration and tuning parameters designed in SigmaStudio, allowing for real-time adjustments during testing.
- **Peripheral Connections:** Essential audio peripherals, including speakers and a microphone, are connected to the master and slave devices to test the functionality of the bi-directional audio streaming system. These connections simulate a real-world automotive audio environment, ensuring that the system meets performance standards for sound clarity and latency.

Each hardware component was rigorously tested to confirm compatibility, functionality, and resilience under simulated automotive conditions. Additionally, the hardware configuration considered factors such as power requirements, signal attenuation, and potential interference to ensure that the audio quality and synchronization were not compromised.

3.4 Software Requirements

The software setup and configuration are crucial to achieving a functional and efficient A2B network. The primary tool for software implementation in this system is **SigmaStudio**, developed by Analog Devices, which is integral for designing the system's schematic, configuring device registers, and conducting real-time system tuning and testing. Using SigmaStudio, a complete schematic of the A2B network is created, capturing all necessary components, connections, and parameters to achieve a reliable and synchronized audio system. The software enables fine-tuned control over audio data handling, latency management, and synchronization settings by configuring the registers on the master and slave devices.

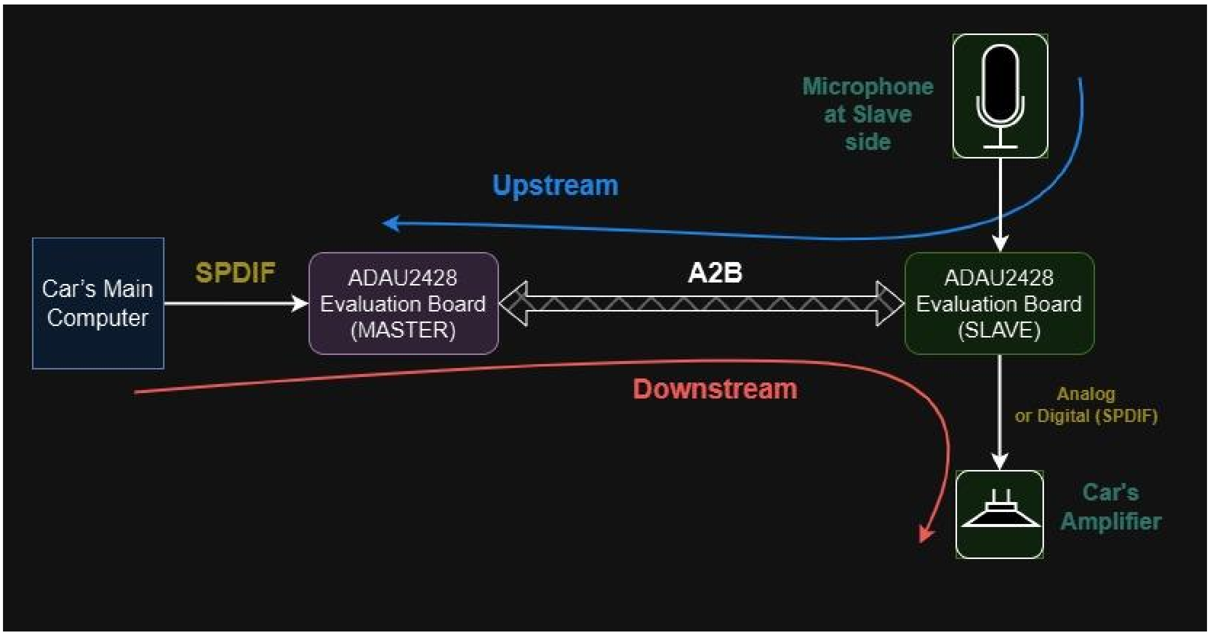

Figure 3.1: A2B Network Schematic, consist of a Master and a Slave, showing the Downstream and Upstream

Chapter 4

Background

4.1 Automative Audio Bus (A2B)

A2B, developed by **Analog Devices**, is a digital audio bus designed to meet modern automotive audio system requirements. It enables the transmission of multi-channel audio, control data, and up to 2.7 W of power over a single unshielded twisted pair (UTP) cable. Configured in a daisy-chain topology, an A2B network consists of a single master node and up to 10 slave nodes [5].

With a maximum bandwidth of 50 Mbps, A2B can support up to 32 audio channels. Supported sample rates include 44.1 kHz and 48 kHz, with channel widths of 12, 16, 24, and 32 bits, allowing for flexible audio configuration. The network can extend up to 40 meters in total length, with a maximum distance of 15 meters between consecutive nodes. Access to the A2B network is managed by A2B transceivers, ensuring reliable data and power transmission across all nodes [6].

4.1.1 A2B Network

An A2B network configuration, shown in Figure 4.1, includes a master and multiple slave nodes, interconnected in a daisy-chain topology. The master node, represented by a red box, consists of an **Audio Host**, a **Target Processor**, and an A2B transceiver. In some setups, the audio host and target processor are combined into a single device. The audio host is responsible for managing the audio data flow and interfaces with the A2B transceiver using Inter-Integrated Circuit Sound (I2S) or Time Division Multiplexing (TDM) formats. The target processor, typically a microcontroller, is required for network configuration at system startup. Once configured, the A2B transceiver manages data communication across the bus, handling the transfer of audio data and Inter-Integrated Circuit (I2C) commands between the audio host, target processor, and the bus [7].

Slave nodes, shown as blue boxes in Figure 4.1, are equipped with A2B transceivers that receive data from the preceding node and transmit it to the next node. These nodes can contain additional devices based on application requirements. Locally, audio data is transferred using I2S/TDM formats, common for inter-IC digital audio communication, or PDM format, which is often used by MEMS microphones. In addition to audio channels, I2C commands and General-Purpose Input/Output (GPIO) signals can also be transmitted across nodes. This feature is especially beneficial when devices connected to slave nodes need configuration at power-on, typically handled by a microcontroller within the slave node. Alternatively, the master node’s host processor can configure all slave node devices, streamlining setup and operation across the network [7].

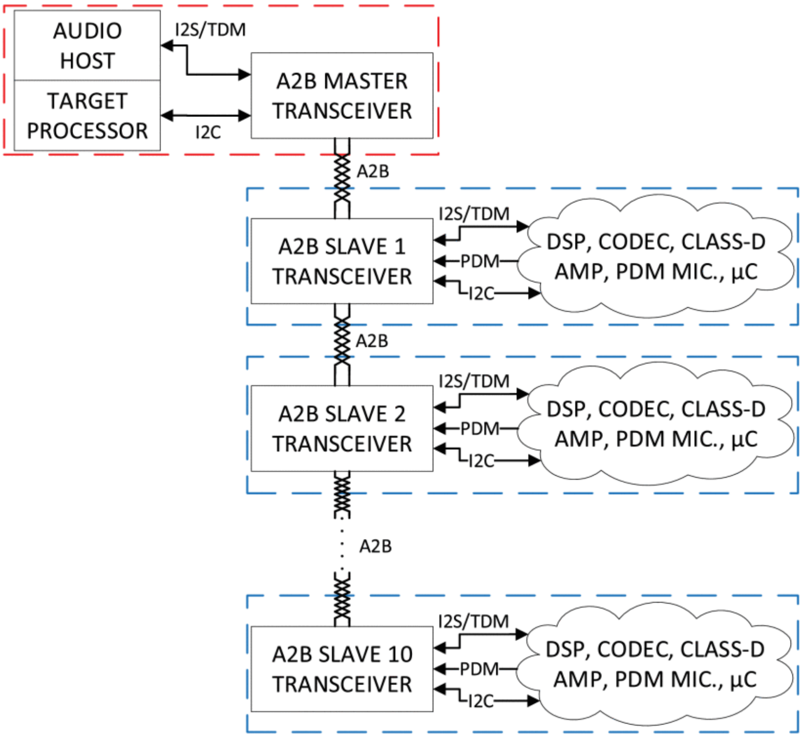

Figure 4.1: Architecture of an A2B network, composed by a daisy-chain of a master node and up to ten slave nodes, includes the Audio Host and Target Processor

4.1.2 A2B Configuration Process

The A2B configuration process, managed by the target processor, consists of two main phases: **Discovery** and **Initialization**. During the discovery phase, the target processor scans the bus for connected nodes and initializes each one. Key transceiver parameters are configured in this phase, including I/O port settings, selection of I2S/TDM or PDM formats, and assignment of audio channel slots. Beyond configuring the A2B slave transceivers, the target processor can also initialize additional peripherals on the slave nodes as needed [8].

Once the discovery and initialization are complete, the audio data flow begins, and the target processor's role becomes minimal, limited to optional diagnostics if required [7]. In typical applications for which A2B is designed, the network configuration is static; once installed, the system's node count, order, channel allocation, and routing are fixed. This predictable setup allows the target processor to predefine the network characteristics and perform configuration automatically at power-on.

During normal operation, the audio host provides the A2B master transceiver with a frame synchronization clock, set to the audio sampling frequency, ensuring all nodes on the network remain synchronized. Communication over the A2B bus occurs in two phases: **Downstream** and **Upstream**. The downstream phase transmits data from the master to the last slave node, while the upstream phase flows from the last slave node back to the master. These two phases together complete within a single sample period, resulting in a fixed latency of two sample times (less than 50µs at a 48 kHz sampling rate), regardless of the node's position within the network.

4.1.3 A2B Superframe

Data transmission over the A2B bus is facilitated by a data packet called a superframe, illustrated in Figure 4.2. Each superframe consists of both a header and a payload for each of the two communication phases—downstream and upstream. In these phases, the headers are designated as the Synch Control Frame (SCF) for downstream and the Synch Response Frame (SRF) for upstream. Each header includes a preamble that supports I2C commands and enables Phase-Locked Loop (PLL) synchronization for the nodes [7]. Following each header, the payload carries one audio sample for each active channel. When a superframe reaches an A2B transceiver, it can either consume specific audio slots or add data intended for downstream nodes or insert data for upstream nodes.

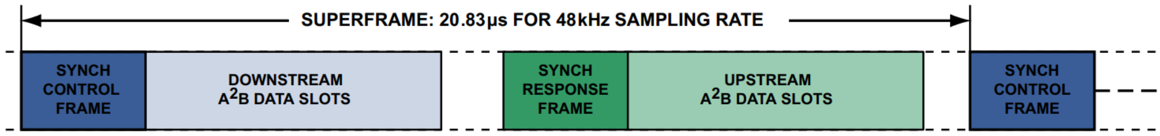

Figure 4.2: Structure of the A2B superframe, consist of the Synch Control Frame (SCF) for downstream and the Synch Response Frame (SRF) for upstream, with sampling rate of 48KHz

It is important to note that all nodes are sampled in sync within the same A2B superframe. For this synchronization, data received through the I2S/TDM port of an A2B transceiver are forwarded over the A2B bus in the following superframe, while data from the A2B bus are directed to the I2S/TDM port in the next superframe. This results in a latency of two superframes for data exchanged between any two nodes, as shown in Figure 4.3. Synchronization across all nodes achieves precise phase alignment. The host generates the I2S/TDM signals, which are then sent to the master node, which transmits the superframes throughout the A2B network at the frame clock signal rate. As previously mentioned, the SCF includes a preamble used by the slaves' PLLs to recover the clock signal. Because the frame clock derived from superframe transmission provides the timing for all slave nodes, it is essential that the clock signal feeding the master node has minimal jitter [6].

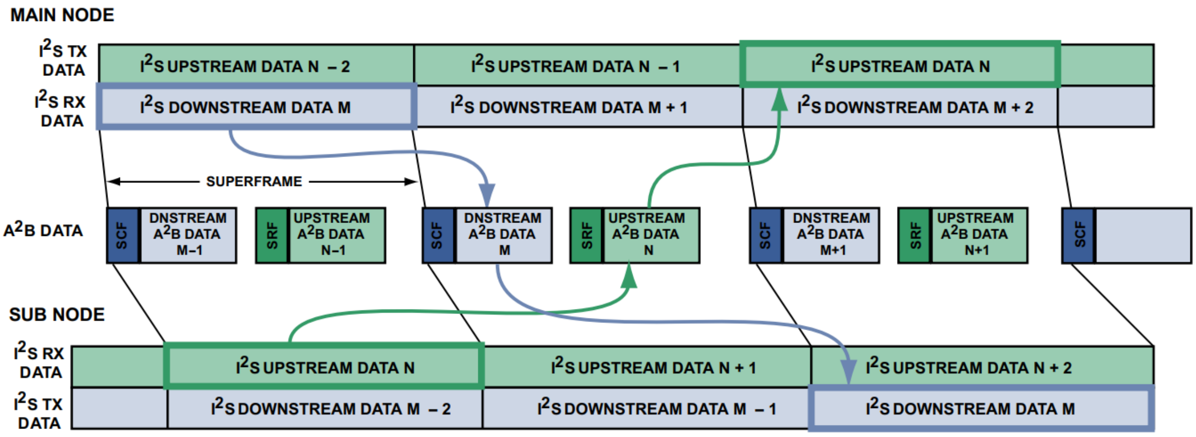

Figure 4.3: A2B Bus Synchronous Data Exchange, showing a latency of two superframes for data exchanged between any two nodes in the Network

4.2 Hardware Overview

Given the wide range of possible configurations for an A2B system, several evaluation platforms, all produced by ADI, are available and can be combined to verify system operation, as shown in Figure 4.4.

Figure 4.4: EVAL-AD2428 family members Evaluation Boards

The EVAL-AD2428WD1BZ stands out from other EVAL-AD2428 family members (such as WB1, WC1, and WG1) due to its enhanced flexibility and expanded GPIO options, making it better suited for complex automotive audio applications. It offers more configurable I/O pins, allowing for better integration with diverse peripherals and improved system customization. Additionally, the WD1BZ variant supports more advanced configurations and features, providing greater versatility in developing and testing A2B network setups, Table 4.1.

Table 4.1: Compare main Characteristics from EVAL-AD2428 family members

A2B Evaluation Board	Main/Subordinate	Power Supply	I2S/TDM	PDM Mics
EVAL-AD2428WD1BZ	Main/Subordinate	Local Power	Yes	3
EVAL-AD2428WB1BZ	Subordinate	BUS Power	Yes	2
EVAL-AD2428WC1BZ	Subordinate	BUS Power	No	4
EVAL-AD2428WG1BZ	Subordinate	Local Power	Yes	0

4.2.1 EVAL-AD2428WD1BZ Evaluation Board

The EVAL-AD2428WD1BZ evaluation board is a versatile tool designed for configuring and testing A2B networks. It can be programmed as either the main (master) or a subordinate (slave) node, making it adaptable for various network setups. The board supports a multichannel I2S/TDM interface for high-quality audio data transfer, allowing integration with multiple audio channels [8]. Key components include the **ADAU1452**, **ADAU1761**, and **AD2428** chips, which together enable efficient audio processing, conversion, and A2B communication, Figure 4.5.

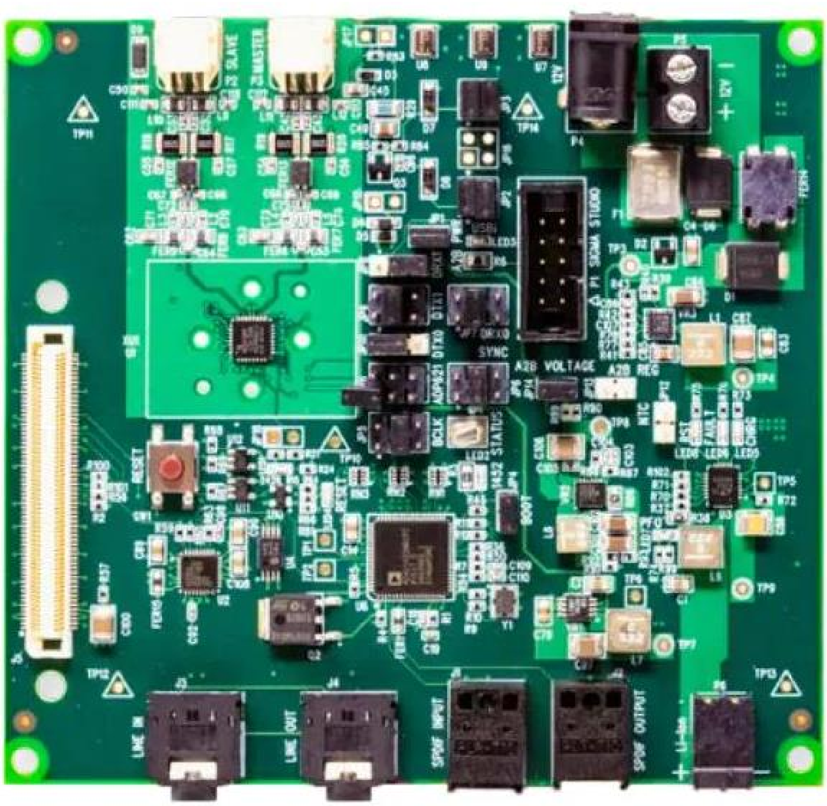

Figure 4.5: EVAL-AD2428WD1BZ evaluation board, version 1.1

Overview:

- I2C interface
- Three PDM microphone inputs
- Locally powered
- Configurable with SigmaStudio™ graphical software tool
- Unique ID register for each transceiver
- Support for crossover or straight-through cabling
- Reset, Status, and Interrupt LEDs
- Jumper Controls
  - Power, clock, and synchronization signal source
  - Data routing

4.2.1.1 EVAL-AD2428WD1BZ Jumpers

The EVAL-AD2428WD1BZ evaluation board features multiple configurable jumpers that are critical for defining its operating mode, power source, clock routing, and data signal pathways, Figure 4.6. The correct setup of these jumpers is necessary for achieving optimal functionality in various test and evaluation scenarios [9]. Below is a description of the jumpers available on the EVAL-AD2428WD1BZ evaluation board.

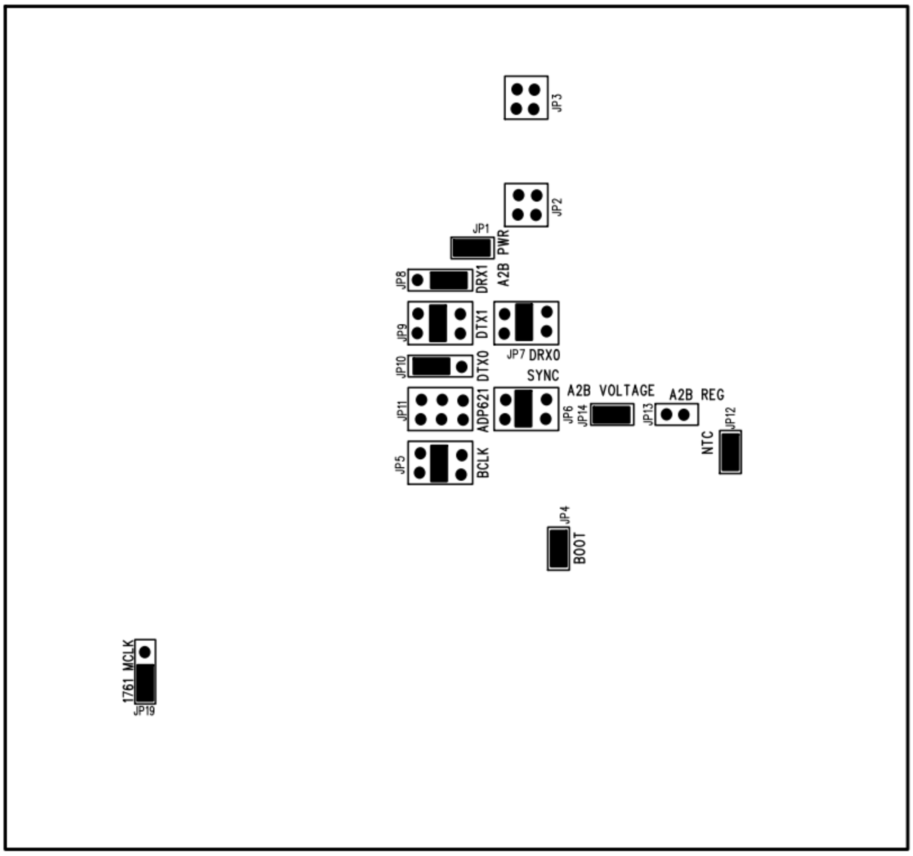

Figure 4.6: Default Jumpers positions and names of the EVAL-AD2428WD1BZ

- **JP1 (A2B Power):** This jumper selects the power source for the A2B transceiver.
    - Installed: The jumper allows the A2B transceiver to receive power from the A2B network, enabling either phantom power or hybrid power support.
    - Uninstalled: Disables phantom power; the board must be locally powered.
- **JP2 (Hybrid/Local Power):** JP2 is used to select between hybrid power mode and local power mode.
    - Installed: When installed, the jumper enables hybrid power mode. In this mode, power is delivered over the A2B network along with communication signals, which reduces the number of external cables.
    - Uninstalled: Local power mode is enabled, meaning the board requires a dedicated external power supply rather than utilizing power from the A2B bus.

- **JP3 (Hybrid Power Support):** This jumper is used to further configure the board's hybrid power capabilities.
    - Installed: Enables hybrid power support, allowing the board to participate in an A2B network where power is shared over the bus.
    - Uninstalled: Disables hybrid power support, requiring the nodes to be powered locally instead of being powered via the A2B bus.
- **JP4 (Boot Mode):** This jumper determines the boot configuration of the board.
    - Installed: Self-boot mode is disabled, meaning the board relies on external configuration to initialize.
    - Uninstalled: Enables self-boot mode, allowing the board to load the configuration directly from the onboard EEPROM without external control.
- **JP5 (BCLK Routing):** JP5 routes the Bit Clock (BCLK) signal to different components on the evaluation board.
    - Pins 3-4: Connects the BCLK signal to ADAU1452_BCLK_OUT0.
    - Pins 1-3: Routes BCLK to ADAU1761_BCLK.
    - Pins 3-5: Connects BCLK to ADAU1452_BCLK_IN0.
- **JP6 (SYNC Signal Routing):** This jumper is used for routing the SYNC (LRCLK) signal on the board.
    - Pins 3-4: Connects the SYNC signal to ADAU1452_LRCLK_OUT0.
    - Pins 1-3: Routes SYNC to ADAU1761_LRCLK.
    - Pins 3-5: Connects SYNC to ADAU1452_LRCLK_IN3.
- **JP7 (DRX0 Data Line):** This jumper selects the destination for the DRX0 data line.
    - Pins 3-4: Routes DRX0 to ADAU1452_SDATA_OUT0.
    - Pins 1-3: Connects DRX0 to ADAU1761_ADC.
    - Pins 3-5: Connects DRX0 to ADMP621_DATA0.
- **JP8 (DRX1 Data Line):** JP8 configures the routing of the DRX1 data line.
    - Pins 2-3: Routes DRX1 to ADAU1452_SDATA_OUT1.
    - Pins 1-2: Connects DRX1 to ADMP621_DATA1.
- **JP9 (DTX1 Data Line):** This jumper determines the routing for the DTX1 data line.
    - Pins 3-4: Connects DTX1 to ADAU1452_SDATA_IN1.
    - Pins 1-3: Routes DTX1 to ADAU1452_SDATA_OUT1.
    - Pins 3-5: Connects DTX1 to ADAU1761_ADC.
- **JP10 (DTX0 Data Line):** JP10 determines the connection for the DTX0 data line.
    - Pins 1-2: Connects DTX0 to ADAU1452_SDATA_IN0.
    - Pins 2-3: Routes DTX0 to ADAU1761_DAC.
- **JP13 (A2B Regulator Selection):** JP13 allows for the selection of the voltage regulator used by the A2B transceiver.
    - Installed: Enables the onboard regulator for stable voltage supply.
    - Uninstalled: Disables the onboard regulator, allowing external regulation.
- **JP14 (A2B Voltage Level Selection):** This jumper is used to select the operating voltage for the A2B transceiver.
    - Pins 1-2: Sets the operating voltage to 3.3V.
    - Pins 2-3: Sets the operating voltage to 1.8V.

4.2.1.2 ADAU1452 Digital Audio Processor

The ADAU1452 Digital Audio Processor is a high-performance audio DSP designed for advanced audio processing applications [10]. It offers a powerful processing core capable of handling complex algorithms, making it ideal for tasks such as filtering, equalization, and audio effects. The ADAU1452 is highly compatible with multichannel audio systems and integrates seamlessly into A2B networks. The functional block diagram is shown in Figure 4.7.

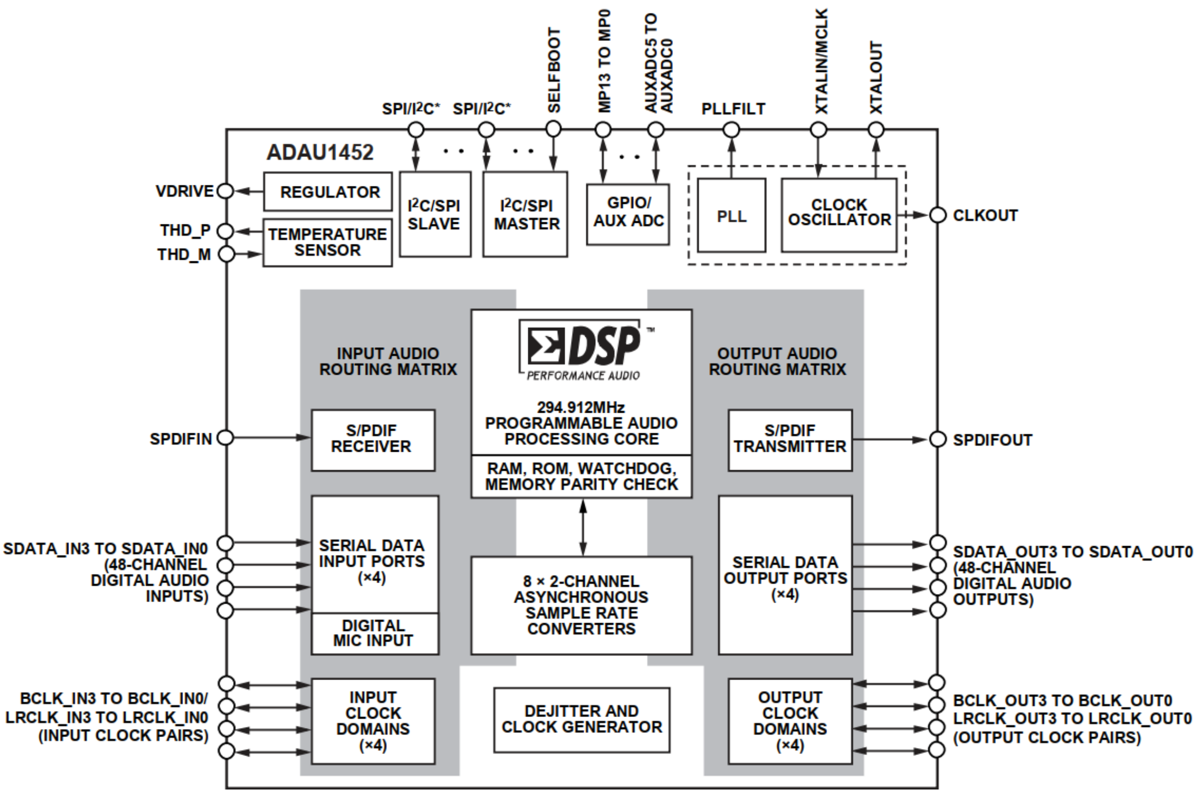

Figure 4.7: Functional Block Diagram of ADAU1452, showing the Serial Data Input/Output ports, Clocks, In/Out SPDIF and GPIO/AUX ADC port and the Temperature Sensor

Important Features:

- 32-bit DSP core
- Up to 294 MHz operating frequency
- Multichannel I2S/TDM audio interface
- Audio I/O and routing
    - 4 serial input ports, 4 serial output ports
    - 48-channel, 32-bit digital I/O up to a sample rate of 192 kHz
    - Flexible configuration for TDM, I2S, left and right justified formats, and PCM
- Up to 8 stereo ASRCs (Asynchronous sample Rate Converters)
- Integrated SigmaStudio software for audio algorithm design
- Low latency processing
- Extensive GPIO options for system integration

4.2.1.3 ADAU1761 Stereo Audio Codec Processor

The ADAU1761 Stereo Audio Codec Processor is a compact audio codec featured on the EVAL-AD2428WD1BZ Evaluation Board. It combines stereo ADCs and DACs with a capable audio processing engine, making it well-suited for real-time audio enhancement [11]. Optimized for low power consumption, it provides high-quality audio conversion and playback, integrating seamlessly within A2B network setups. The functional block diagram is shown in Figure 4.8.

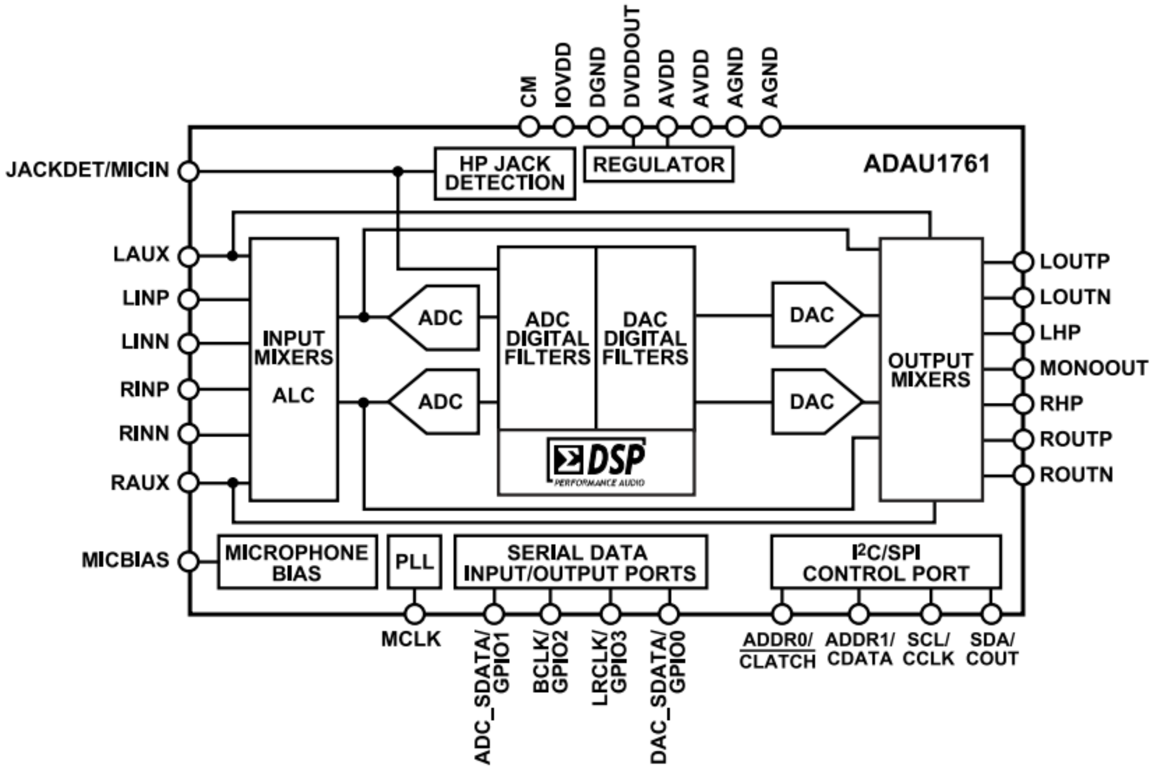

Figure 4.8: Functional Block Diagram of ADAU1761, showing the ADDR0 and ADDR1 for addressing, I2C/SPI Control Port, Serial Data Input/Output ports and the PLL

Important Features:

- 24-bit stereo audio ADC and DAC: >98 dB SNR
- Sampling rates from 8 kHz to 96 kHz
- 6 analog input pins, configurable for single-ended or differential inputs
- Stereo digital microphone input
- Analog outputs: 2 differential stereo, 2 single-ended stereo, 1 mono headphone output
- Phase-locked Loop (PLL) supporting input clocks from 8 MHz to 27 MHz
- I2C and serial peripheral interface (SPI) control interfaces
- Digital audio serial data I/O: stereo and time-division multiplexing (TDM) modes
- GPIO pins for digital controls and outputs
- Low power: 7 mW record, 7 mW playback, 48 kHz at 1.8 V

4.2.1.4 AD2428 A2B Main/Subordinate Processor

The AD2428 Automotive Audio Bus (A2B) Transceiver integrates a range of critical functional blocks that streamline audio processing and communication in automotive systems, as illustrated in Figure 4.9. At its core, it features a robust A2B controller that manages the low-latency transmission of audio and control data across multiple network nodes, ensuring synchronization and efficient routing. The device incorporates dual voltage regulators, VREG1 and VREG2, which generate stable supply voltages for its core logic and I/O circuits, reducing external component requirements and ensuring reliable operation under varying conditions. An integrated phase-locked loop (PLL) provides precise clock signals to support high-quality audio synchronization across multiple channels, while the audio interfaces, compatible with formats such as I²S and TDM, enable seamless integration with other digital audio peripherals. Additionally, the AD2428 includes configurable I²C and SPI interfaces that allow a host microcontroller to control device settings and monitor status. Its comprehensive diagnostic features ensure real-time monitoring of power, connectivity, and signal integrity, enhancing system reliability in challenging automotive environments. Together, these elements, as depicted in Figure 4.9, make the AD2428 a highly efficient and versatile solution for distributed automotive audio networks.

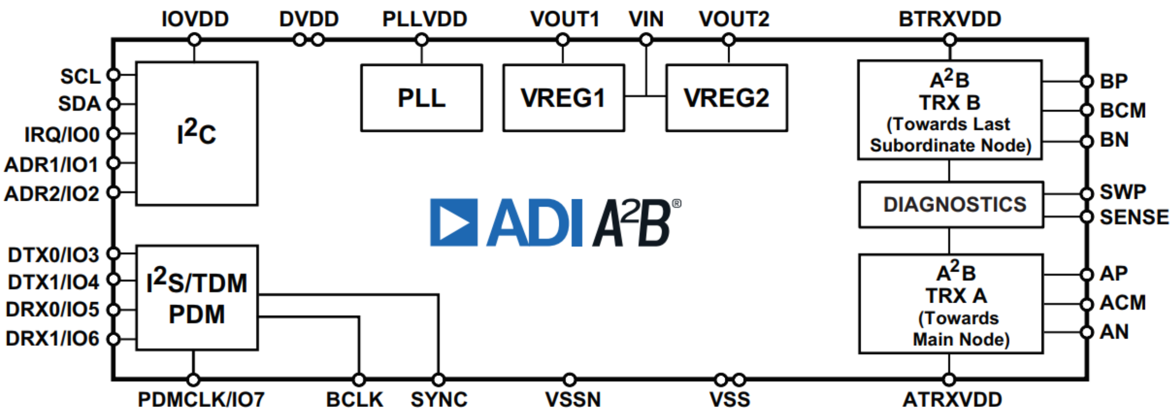

Figure 4.9: Functional Block Diagram of AD2428, showing the A2B TRX A/B, I2C unit and ports, I2S/TDM and PDM unit and ports and the PLL

Important Features:

- Supports single master and up to 10 slave nodes
- 50 Mbps bandwidth for multi-channel audio
- Daisy-chain configuration with up to 40 meters total cable length
- Integrated I2S/TDM and PDM audio interfaces
- Low latency, deterministic audio transport

4.2.2 EVAL-ADUSB2EBZ Interface Board

The EVAL-ADUSB2EBZ is a high-performance USB interface board designed to facilitate communication between a PC and A2B-enabled evaluation boards, supporting seamless integration and control of audio networks during testing and development, Figure 4.10.

Figure 4.10: EVAL-ADUSB2EBZ USB Interface Board, receiving Data on the USB and transfer them to A2B node through the I2C/SPI

This board provides an essential interface for configuring, monitoring, and managing A2B networks via a user-friendly software interface, enabling real-time adjustments and data analysis for optimal performance. Equipped with USB connectivity, the EVAL-ADUSB2EBZ enhances flexibility and ease of use for developers working on A2B audio applications. The block diagram is shown in Figure 4.11.

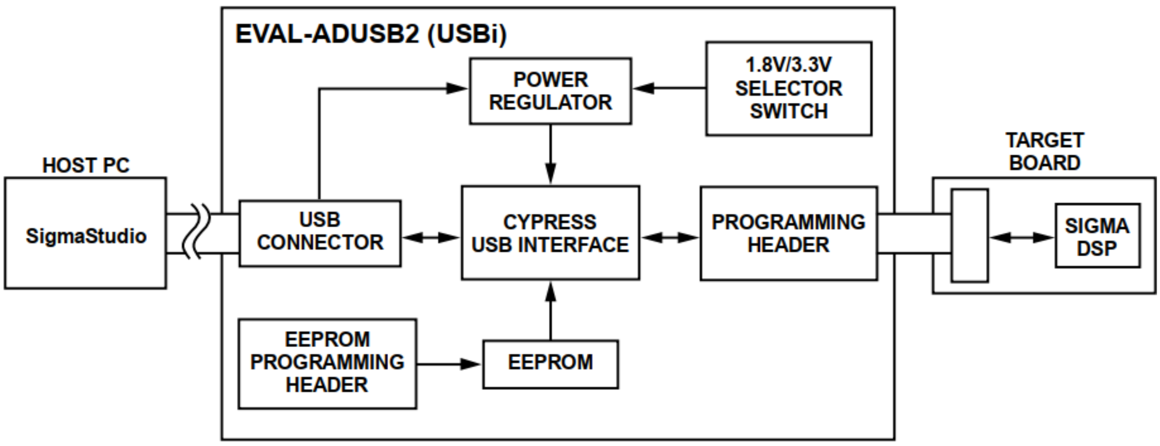

Figure 4.11: The functional block diagram of EVAL-ADUSB2EBZ USB Interface Board, showing the input and output and different functional units

4.3 Software Overview

SigmaStudio is a powerful graphical software tool developed by Analog Devices, specifically tailored for configuring and programming their SigmaDSP digital signal processors [12]. The software offers an intuitive drag-and-drop interface, enabling engineers to design and implement audio signal processing tasks with ease. From fundamental operations like equalization and filtering to advanced dynamics processing and custom audio algorithms, SigmaStudio provides a comprehensive library of audio processing blocks. It is particularly valuable for designing and managing A2B networks, where it facilitates the seamless integration of audio and control data over a single unshielded twisted-pair cable. By using SigmaStudio, engineers can visually assemble, configure, and validate their audio signal paths without requiring deep knowledge of programming languages, as shown in Figure 4.12.

In addition to its extensive audio signal processing capabilities, SigmaStudio serves as a versatile tool for real-time tuning and debugging [13]. The graphical interface not only allows for the easy prototyping of DSP designs but also enables on-the-fly adjustments and performance optimization. With its ability to interface directly with hardware, the tool significantly accelerates the development process by providing immediate feedback and enabling iterative testing. Furthermore, its compatibility with a wide range of Analog Devices hardware platforms ensures that it meets the diverse requirements of modern audio system designs, from automotive audio distribution to industrial signal processing applications. By streamlining complex signal processing tasks into an accessible and efficient workflow, SigmaStudio has become an indispensable resource for engineers seeking to innovate in the field of digital audio.

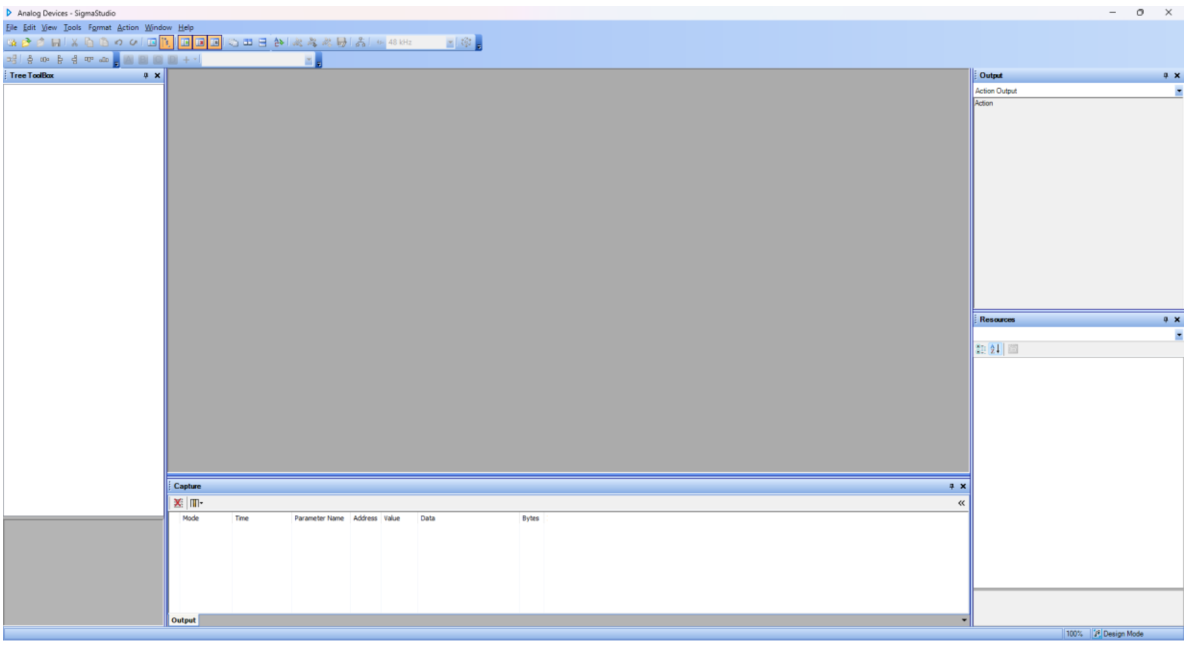

Figure 4.12: SigmaStudio Main Program Window

4.3.1 Hardware Configuration Tab

The Hardware Configuration Tab in SigmaStudio is where users establish the fundamental connections for their DSP and peripheral devices, defining the core hardware setup for the audio processing system. In this workspace, users can select one or multiple processors for their design and configure communication between SigmaStudio and the hardware by adding appropriate interface modules, Figure 4.13.

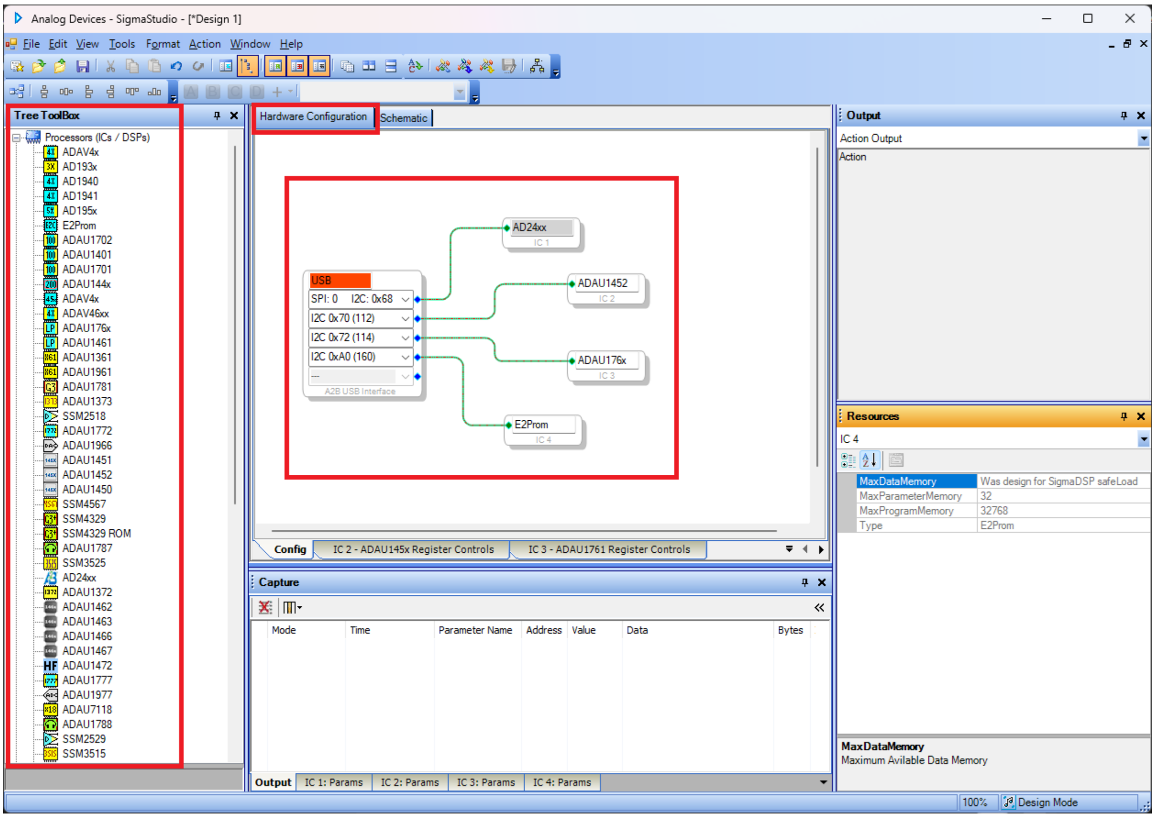

Figure 4.13: SigmaStudio Hardware Configuration Tab, where various modules and processors can be added by dragging them into the workspace

4.3.2 Schematic Tab

The Schematic Tab is the core design space in SigmaStudio, where users construct and visualize their DSP designs and manage the A2B network configuration. After setting up the hardware configuration, users are ready to build signal processing chains using this workspace. The Toolbox on the left provides access to a comprehensive set of libraries containing DSP building blocks, such as filters, equalizers, and mixers. These blocks can be easily dragged and dropped onto the schematic workspace on the right (Figure 4.14), allowing for a block-based programming approach that facilitates customization and real-time adjustments. The interconnections between these blocks can be defined by simply drawing lines, representing the flow of signals within the design. This visual representation ensures that the signal processing architecture remains clear and intuitive, even for complex configurations.

The Schematic Tab also includes features for debugging, such as monitoring signal levels and accessing real-time parameters, further enhancing the development experience. This workspace is pivotal for bridging conceptual designs with practical implementation, making it a cornerstone of SigmaStudio's functionality.

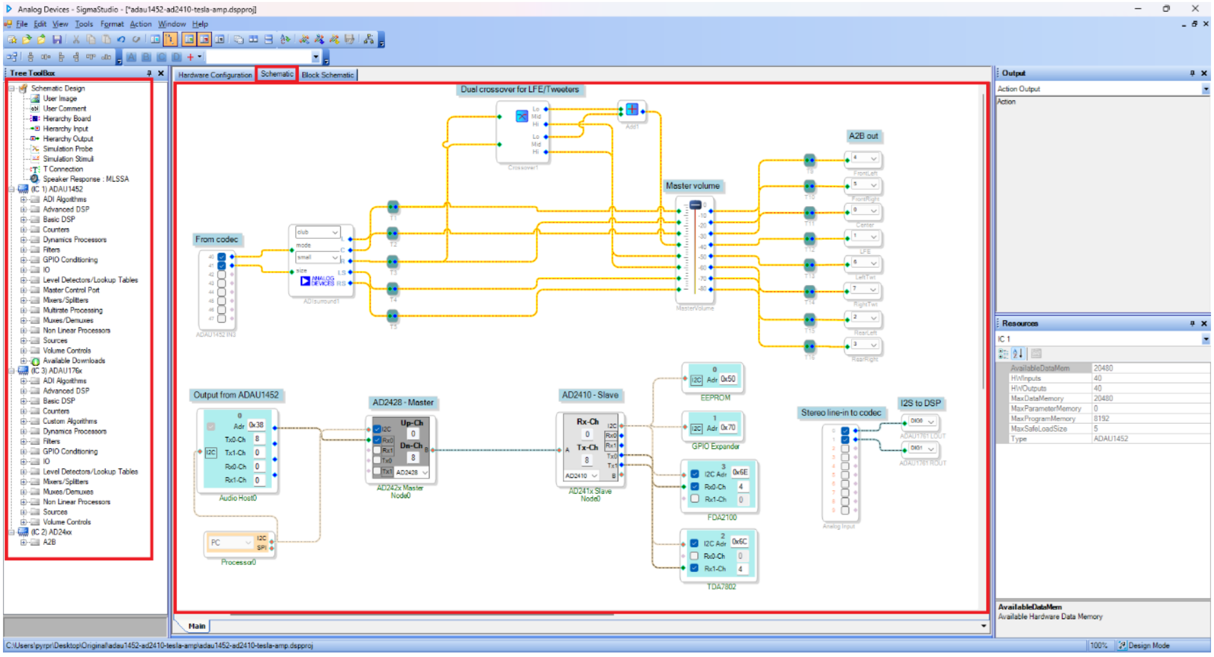

Figure 4.14: SigmaStudio Schematic Tab, where different modules, transceiver nodes, and input/output components can be added by dragging them into the workspace, with wiring easily established between them

4.3.3 Register Window Tab

The Register Window Tab is a crucial feature, providing detailed control over the internal registers of the DSP. This tab allows developers to fine-tune DSP behavior at a low level, which can be essential for debugging and optimizing performance, Figure 4.15.

Important Features:

- **Direct Register Access:** Offers a direct interface to view and edit the values in DSP registers, giving full control over the device's low-level operation.
- **Real-Time Monitoring:** Users can monitor register values in real-time, making it easier to identify and troubleshoot issues in the signal processing chain.
- **Debugging Support:** For issues not visible in the Schematic or Hardware Configuration tabs, the Register Window provides insights that help in isolating and resolving hardware or software configuration issues.
- **Register Read/Write Operations:** Supports read and write operations for individual registers.

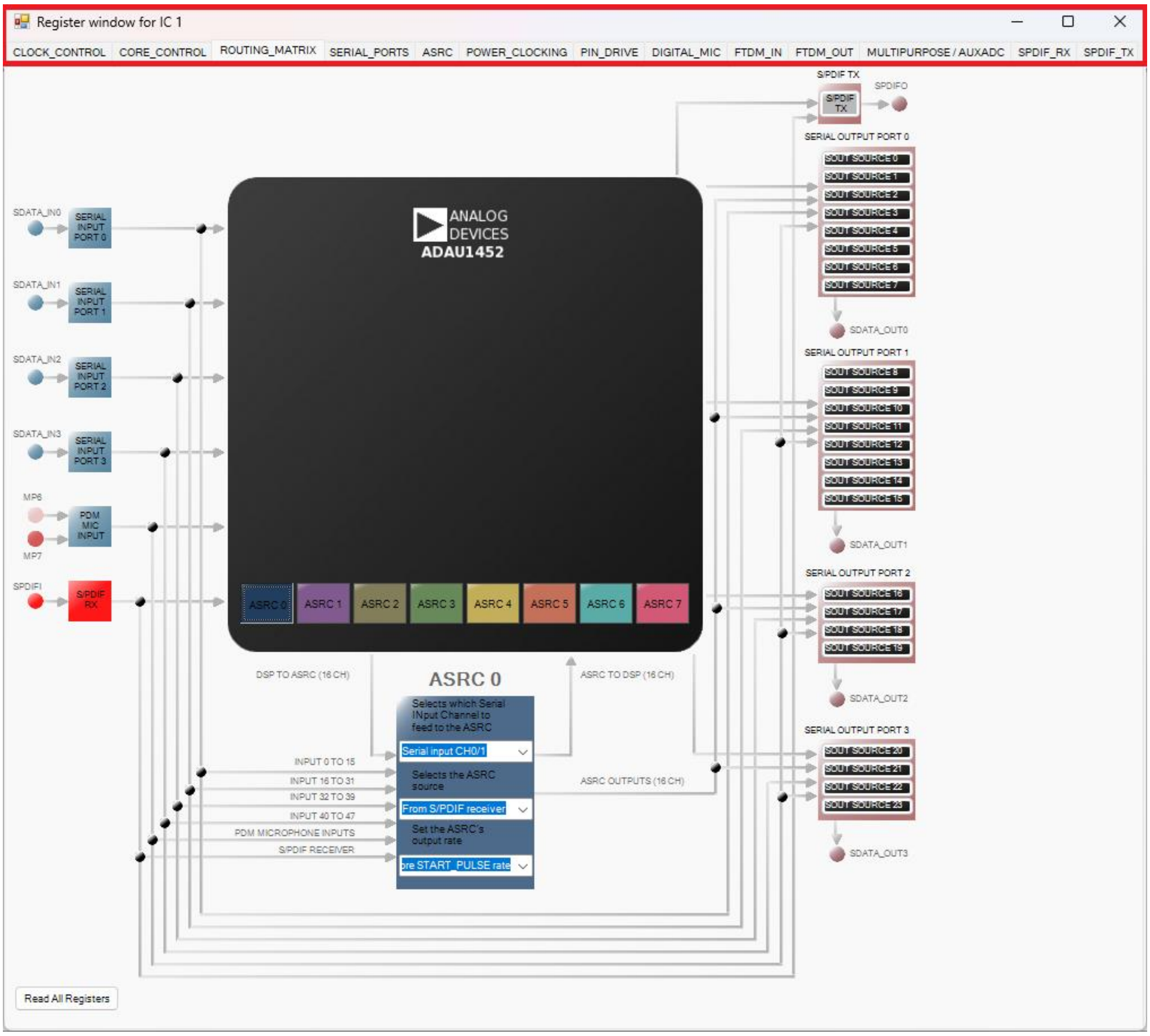

Figure 4.15: SigmaStudio Register Window Tab, which varies for each processor, allows configuration of all registers in a graphical mode

4.4 Digital Audio and Communication Protocols Overview

4.4.1 Inter-Integrated Circuit (I2C)

The Inter-Integrated Circuit (I2C) protocol is a widely used, two-wire, serial communication bus developed by Philips [14]. I2C is used to establish communication between a master device and multiple subordinate devices via two main signals, as shown in Figure 4.16: the serial data line (SDA) and the serial clock line (SCL).

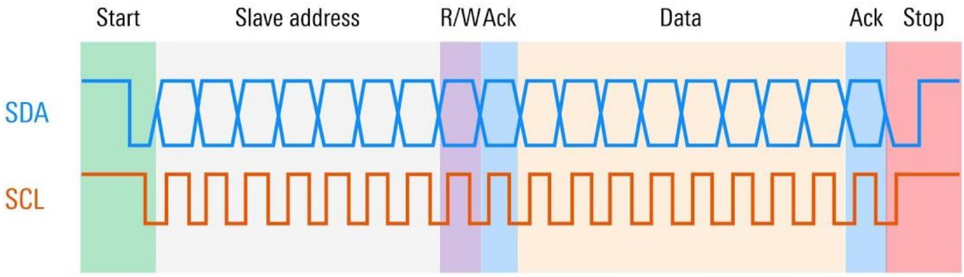

Figure 4.16: Overview of I2C frame, consist of the serial data line (SDA) and the serial clock line (SCL), showing the corresponding bits of the Start, Slave Address, Read/Write, Acknowledgement, Data, Acknowledgement and Stop

The master device generates the clock on the SCL line, synchronizing data transmission over SDA, allowing data to be transferred in both directions between master and subordinate devices. The protocol is flexible, supporting multiple subordinate devices on the same bus, each with a unique address for device identification, which simplifies the connection between low-speed peripherals in embedded systems. The SDA and SCL lines are typically pulled high using resistors, ensuring the bus remains idle until actively driven by a device.

The ADAU1452 and ADAU1761 both support a two-wire serial (I2C-compatible) bus for communication with a microprocessor, enabling them to interface with multiple peripherals. The maximum clock frequency for the I2C slave port is 400 kHz. The ADAU1452 and ADAU1761 each utilize two pins, serial data (SDA) and serial clock (SCL), to carry information to the system's I2C master controller. It is important to note that in I2C mode, both the ADAU1452 and ADAU1761 always function as subordinate devices, meaning they are incapable of initiating a data transfer on the I2C bus. Each subordinate device can be recognized by its unique address, and the specific address format can be found in Table 4.2.

Table 4.2: ADAU1452 and ADAU1761 I2C address

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	1	1	1	0	ADDR1	ADDR0	$R / \overline{W}$

The first seven bits of the I2C write operation contain the address. Bits [5:6] of the I2C address for the ADAU1452 and ADAU1761 are determined by the states of the ADDR1 and ADDR0 pins. The least significant bit (LSB) of the address, indicates whether the operation is a read or write.

4.4.2 Inter-Integrated Circuit Sound (I2S)

The Inter-IC Sound (I2S) protocol is a digital audio communication interface developed by Philips [15]. It is designed to enable the transfer of digital audio data between ICs, such as between audio codecs and processors. As shown in Figure 4.17, I2S consists of three main signals: Serial Clock (SCK), Word Select (WS), and Serial Data (SD). The SCK signal acts as the reference clock for synchronizing audio data transmission, while the WS signal, also known as the frame clock, determines whether the data being transmitted belongs to the left or right audio channel. The Serial Data (SD) line is used for carrying the actual audio data and can function either as Serial Data Input (SDI) or Serial Data Output (SDO) depending on the direction of transmission.

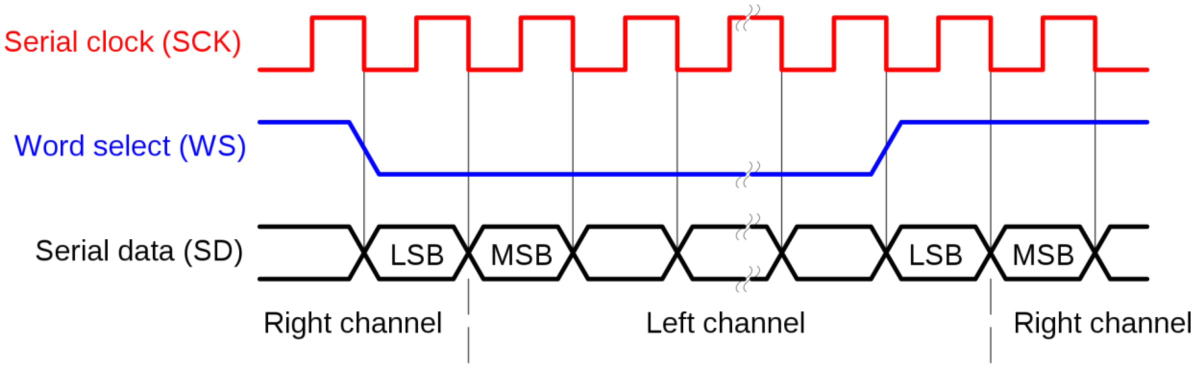

Figure 4.17: Overview of I2S frame, showing the Serial Clock (SCK), Word Select (WS), Serial Data (SD), the position of the MSB and LSB, and the Left/Right channel selection

According to the Philips I2S bus specification revised in June 1996, the WS signal is used to control the channel assignment: when WS is set to 0, it indicates that the left channel audio data is being transmitted, and when WS is set to 1, it indicates that the right channel audio data is being transmitted. Data is transmitted in binary two's complement format, ensuring precision in audio representation. Additionally, whenever the WS signal changes state, the transmitter sends the Most Significant Bit (MSB) of the audio data first, allowing for proper alignment of audio samples at the receiving end. This structure ensures that the left and right channel data are transmitted consistently and reliably, making I2S a preferred protocol for high-fidelity digital audio communication.

In an A2B (Automotive Audio Bus) network, I2S is used to transmit stereo audio (2-channel) between devices in an automotive audio environment. The A2B network leverages the simplicity and efficiency of I2S to enable the transmission of high-quality stereo audio signals from a source device, such as a head unit, to various audio endpoints like amplifiers or speakers. The A2B network architecture supports both downstream and upstream communication, allowing audio signals to flow from the head unit to peripheral devices and vice versa. This bi-directional capability is particularly useful for modern automotive systems where audio signals need to be both distributed to speakers and collected from microphones for features such as active noise cancellation or in-car communication.

4.4.3 Time-Division Multiplexing (TDM)

Time-Division Multiplexing (TDM) is a method of transmitting multiple data streams over a single communication channel by dividing the channel into discrete time slots. Each time slot is assigned to a different data stream, allowing for sequential but non-overlapping data transmission. This approach ensures efficient use of bandwidth by allowing multiple signals to share the same communication medium while preventing collisions between signals. In TDM, the data sources are sampled in a round-robin fashion, and each source is assigned a fixed time slot during which its data is transmitted. This multiplexing technique is widely used in telecommunications, networking, and digital audio systems to optimize the channel's utilization and manage the transmission of multiple data types concurrently [16].

In audio systems, TDM is particularly useful in scenarios where multiple audio channels need to be transmitted simultaneously over a single data line. Unlike other multiplexing techniques, TDM is especially effective for real-time applications, such as digital audio, where maintaining the quality of the transmitted signals is paramount. The synchronized nature of TDM ensures that each channel maintains its audio integrity and timing, making it an ideal solution for professional and consumer audio applications. TDM is also well-suited for applications that require low latency, high channel density, and reliable data integrity, particularly in automotive and industrial environments.

In an A2B (Automotive Audio Bus) network, TDM is used to facilitate the transmission of audio signals across multiple channels using a single communication pathway. A2B networks employ TDM to transport digital audio data between a master device, such as a head unit, and various slave devices, such as amplifiers, microphones, and speakers, connected in a daisy-chain configuration. This approach significantly reduces the wiring required in automotive audio systems, as a single twisted-pair cable is sufficient to transmit audio, control signals, and even power. By utilizing TDM, an A2B network can efficiently manage multiple audio channels, minimizing interference and ensuring reliable communication throughout the vehicle's audio system.

TDM audio transmission in an A2B network can operate in multiple configurations, typically supporting 2, 4, 8, or 16 channels. Each configuration corresponds to a specific number of audio signals being transmitted within a single TDM frame. For instance, in the 2-channel mode, the TDM frame is divided into two time slots, each allocated to one of the stereo channels (left and right). This mode is ideal for basic stereo audio transmission from the head unit to the front speakers. In the 4-channel mode, the TDM frame is extended to carry four separate audio signals, which could be used for additional rear speakers or for an enhanced audio experience involving more speakers positioned around the vehicle.

The 8-channel mode in TDM is often used in advanced automotive audio systems that require multi-zone audio, where different parts of the vehicle receive distinct audio streams. Similarly, the 16-channel mode is designed for high-end vehicles with sophisticated audio systems that integrate multiple microphones, speaker arrays, and subwoofers to create a surround sound environment.

4.4.4 Serial Peripheral Interface (SPI)

The Serial Peripheral Interface (SPI) is a synchronous serial communication protocol commonly used for short-distance data transmission between microcontrollers and peripheral devices. It was developed by Motorola [17] and remains popular in embedded systems for its simplicity and high-speed data transfer capabilities. SPI uses four main signals, as shown in Figure 4.18: the Serial Clock (SCLK), Master Output Slave Input (MOSI), Master Input Slave Output (MISO), and the Chip Select (CS). The SCLK signal provides the clock generated by the master device to synchronize data transmission. The MOSI line carries data from the master to the slave, while the MISO line carries data from the slave to the master. The CS line, also known as the slave-select line, is used by the master to activate the specific slave with which it wishes to communicate.

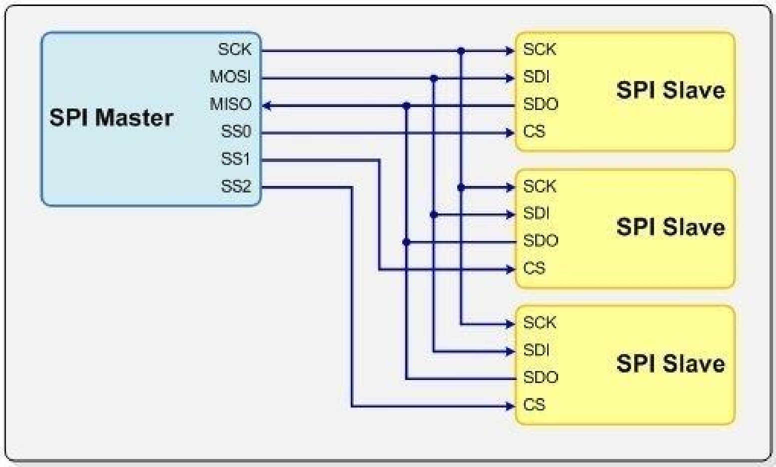

Figure 4.18: SPI bus signals with multiple Slaves, showing the Serial Clock (SCLK), Master Output Slave Input (MOSI), Master Input Slave Output (MISO), and the Chip Select (CS)

Unlike other serial communication protocols, SPI is well-known for its full-duplex communication, meaning that data can be transmitted and received simultaneously. The SPI interface is simple but highly effective for point-to-point communication or for interfacing with multiple slave devices, each uniquely controlled by a separate CS line. Data transmission in SPI is typically fast, and its straightforward, synchronous nature allows for efficient control and data exchanges between components in a variety of embedded systems.

In an A2B (Automotive Audio Bus) network, SPI is employed to facilitate communication between the master controller and slave devices, particularly in scenarios where control and configuration data need to be exchanged. SPI provides a robust and efficient way to set up peripheral devices within the A2B network, such as configuring amplifiers or other audio processing units, before transmitting audio data through the TDM or I2S protocols. In an A2B environment, the master node can use SPI to manage device settings and states, ensuring that all components are correctly synchronized and configured for optimal performance.

4.4.5 Pulse Density Modulation (PDM)

Pulse Density Modulation (PDM) is a type of digital signal representation that is often used in audio applications, particularly in digital microphones. PDM works by representing an analog audio signal as a series of rapid pulses, where the density of the pulses corresponds to the amplitude of the original analog waveform. Essentially, a higher density of pulses indicates a higher amplitude, and a lower density corresponds to a lower amplitude. PDM is simple to implement, which makes it highly suitable for low-cost and power-efficient audio solutions. PDM microphones, in particular, are popular in applications requiring small form factor and low power consumption, such as mobile devices, headphones, and hearing aids.

The AD2428WD1BZ is a transceiver that integrates PDM microphones, providing an efficient solution for automotive audio applications. By embedding PDM microphones on the AD2428WD1BZ, the transceiver can easily collect audio data directly from the microphones and transmit it through the A2B network. This integration not only reduces the complexity of the audio system but also helps to lower costs by minimizing the number of additional components needed. The AD2428WD1BZ with embedded PDM microphones is well-suited for use in automotive environments where robust and efficient audio capture is required, such as for voice recognition systems, in-cabin communication, and noise cancellation.

4.4.6 Pulse Code Modulation (PCM)

Pulse Code Modulation (PCM) is a method used to digitally represent analog signals, particularly for audio applications. PCM involves sampling the analog audio signal at regular intervals, quantizing the sample values to a finite set of levels, and then encoding these quantized values as binary data. The sampling process converts the continuous waveform into a series of discrete values, while the quantization assigns each value to a specific level, allowing the analog signal to be represented digitally. The encoded data is then transmitted or stored, and it can be decoded back into an approximation of the original analog signal. PCM is widely used for audio storage and transmission due to its simplicity and its ability to accurately capture the nuances of an analog audio signal.

In an A2B network, the PCM data is transported between a master device, such as a vehicle's head unit, and various slave nodes, including amplifiers and speaker systems. The ADAU1452 acts as a key component in processing this PCM audio, handling various audio processing tasks, such as equalization, filtering, or dynamic range control, before sending the processed data to the appropriate nodes. By using linear PCM, the ADAU1452 ensures that audio data within the A2B network maintains the integrity and quality required for automotive audio applications, allowing for a rich and immersive audio experience. This capability is particularly important in automotive environments where multiple audio sources and destinations coexist, requiring precise synchronization and robust communication across all components.

Chapter 5

Development of A2B Network

In this chapter, the hardware setup is outlined first, followed by the DSP schematic design in SigmaStudio, and the configuration of Master/Slave registers. Finally, the complete design is compiled and downloaded to the Master and Slave nodes via the USB interface connected to the Master.

5.1 Hardware Implementation

The hardware implementation of the A2B network is designed to facilitate bi-directional audio streaming within an automotive environment. In this setup, the system employs two A2B evaluation boards (EVAL-AD2428WD1BZ), one configured as the master and the other as a slave. These boards enable audio signal transmission through the A2B bus, leveraging the advantages of a daisy-chain topology that simplifies wiring and reduces overall system complexity.

In this thesis, the slave board is equipped with embedded Pulse Density Modulation (PDM) microphones. These PDM microphones are integrated directly onto the slave device, allowing for efficient audio capture with minimal external hardware. The PDM microphone modules are capable of providing high-quality audio input, and the system is designed to receive a 4-channel audio signal from these microphones. This configuration improves the system's ability to capture in-cabin audio, offering high fidelity and reduced latency. The integration of PDM microphones also helps to reduce wiring and enhances the overall audio system's performance by providing synchronized and high-quality audio input.

The A2B network in this implementation uses different types of ports, each designed for specific functions within the network. The A2B ports, referred to as 2-pin Duraclick ports, are used to connect the master and slave devices through an unshielded twisted pair (UTP) cable. This connection not only transmits audio and control data but also provides power to slave nodes, making it a versatile solution for distributed audio systems. The system also utilizes Sony/Philips Digital Interface (SPDIF) ports, which are configured for transmitting high-quality digital audio signals to and from the master node. The SPDIF ports provide a robust interface for ensuring high fidelity audio data exchange between the car's head unit and the A2B master.

A USB interface port, provided by the EVAL-ADUSB2EBZ evaluation board, is used to facilitate configuration and monitoring of the A2B network via a PC. This USB interface allows for easy control of the master device's settings and real-time monitoring during testing, making it a key component in system setup and evaluation. Additionally, the power supply for the A2B network is provided through a 12V Power Plug. In this thesis, a standard 12V Power Plug is used; however, a 12V Power Terminal could also be utilized to supply power directly to the evaluation boards if needed. This flexibility in power supply options makes the system adaptable for different automotive integration requirements.

The hardware configuration ensures that all components work seamlessly to achieve synchronized, bi-directional audio streaming. The master board transmits a 2-channel audio stream to the slave while simultaneously receiving a 4-channel audio stream from the PDM microphones on the slave. This bi-directional setup is crucial for applications such as in-car communication systems, active noise cancellation, and immersive in-cabin audio experiences. Proper configuration of ports, power supply, and audio channels was essential to ensure system stability and performance under different automotive conditions.

5.1.1 Hardware Setup and Audio Routing

As illustrated in Figure 5.1, the B port (A2B output to the Slave) is connected to the A port (A2B input from the Master), establishing the bi-directional A2B bus. A 2-channel audio stream is received by the Master's SPDIF-in port via the SPDIF (Sony/Philips Digital Interface). On the Slave side, this audio stream, generated by the vehicle's main computer on the Master, is sent through the SPDIF-out port to the vehicle amplifier. Additionally, instead of using an analog input from an external microphone, the slave board is equipped with embedded PDM microphones. These microphones are capable of providing a high-quality digital audio signal. Specifically, four channels of audio are captured through these PDM microphones and transmitted over the A2B bus to the Master, enhancing the overall audio capture capability and reducing latency and complexity compared to traditional analog inputs.

The PDM microphones integrated on the slave board are highly compact, offering robust in-cabin audio monitoring for automotive systems. PDM (Pulse Density Modulation) technology efficiently converts analog sound into digital data with minimal latency while maintaining high fidelity. This characteristic is particularly crucial for various automotive applications, including voice command systems, hands-free calling, and active noise cancellation, where accurate and timely audio capture is essential. The use of PDM microphones, rather than traditional analog microphones, enhances both the clarity and quality of captured audio, even in the challenging noise conditions typical of a car interior.

The PDM block on the A2B transceiver is responsible for converting PDM input streams into pulse code modulated (PCM) data, which can then be transmitted over the A2B bus or forwarded to the local node through the I2S/TDM port. This conversion process ensures compatibility with different nodes in the network while retaining high audio quality. Moreover, the PDM block is designed to support high dynamic range microphones, which boast a high signal-to-noise ratio (SNR) and extended maximum sound pressure level (SPL). This capability allows the system to handle a wide range of audio inputs, from soft speech to loud ambient sounds, while preserving fidelity. The interface also supports multiple frame rates, including 12 kHz, 24 kHz, and 48 kHz, providing flexibility for use in both main and subordinate transceivers.

Additionally, each PDM-enabled receive pin can handle up to two audio channels (stereo), which are synchronized with the clock signal. One audio channel is linked to the rising edge of the clock, while the other is associated with the falling edge. This dual-channel functionality optimizes audio data transmission and enables efficient use of resources, allowing the system to capture and process stereo audio without additional complexity. The A2B network relies on different types of ports to manage data and power transmission. The A2B ports, named 2-pin Duraclick ports, facilitate the connection between the master and slave devices through UTP (Unshielded Twisted Pair) cables. These Duraclick ports are specifically designed to handle both data and power transmission, simplifying the system architecture by reducing the need for multiple cables. The SPDIF ports on the master and slave boards handle the digital audio input and output, ensuring that the audio signal maintains high fidelity during transmission between the vehicle's main computer and the A2B nodes.

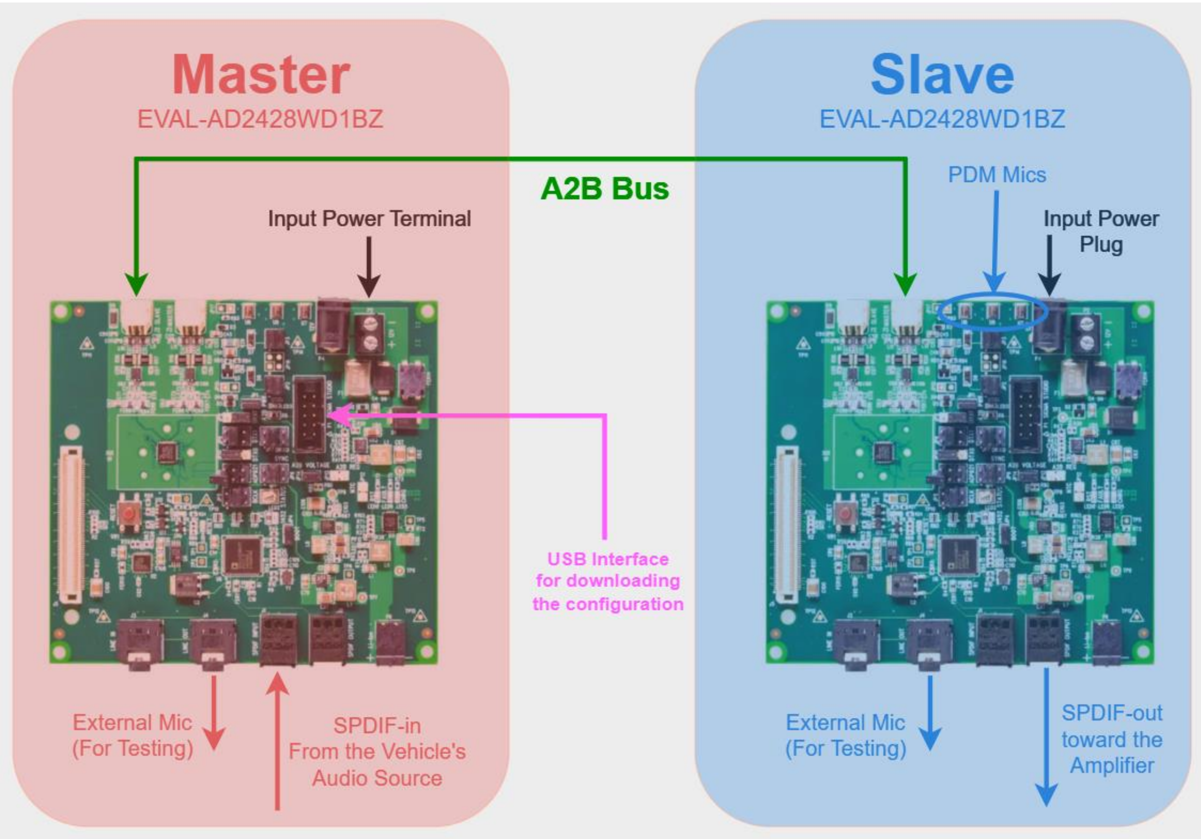

Figure 5.1: A2B Hardware Implementation, showing the routing between a Master and one Slave, the input power port/terminal and the audio input/output ports and the USB interface

Additionally, a USB interface is provided via the EVAL-ADUSB2EBZ evaluation board, which connects the A2B network to a PC. This USB interface is used for configuring the network, monitoring audio performance, and making necessary adjustments in real-time. The power for the A2B network is supplied via a 12V Power Plug, but there is also an option to use a 12V Power Terminal if a more permanent installation is required.

5.2 Schematic Design

5.2.1 Schematic of USB Interface Towards the Master

As previously mentioned, the USB interface evaluation board is connected to the Master board, providing direct access to various processors and the Master's EEPROM via I2C and SPI protocols. To enable this setup, the necessary blocks are added to the Hardware Configuration Tab, and the appropriate wiring connections are established. According to the datasheet, the correct device address for each processor and the EEPROM is selected from the available address list, Figure 5.2. The EEPROM on the AD2428WD1BZ evaluation board stores configuration data for the A2B network, allowing the Master node to automatically load settings during initialization.

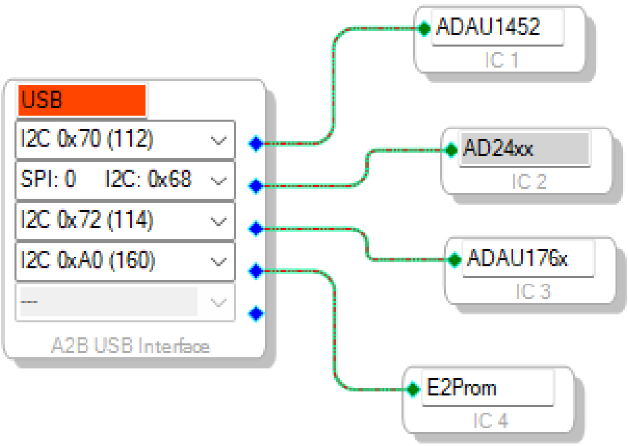

Figure 5.2: Schematic of the USB interface and the connections to the ADAU1452 and AD2428 and ADAU1761 processors and an embedded EEPROM

5.2.2 Schematic of A2B Network

In the Schematic Tab, the A2B network is configured by adding the required blocks into the workspace. As shown in Figure 5.3, a **Target Processor** block is inserted, representing the controller that connects to the master A2B node on the bus; here, this controller is the PC running SigmaStudio. Through the master A2B node, the target processor can control any A2B or peripheral node on the bus. In Figure 5.4, the "*Simple*" Discovery mode is selected, enabling sequential detection of slave nodes from slave 0 to the last available node in the system. Once all slaves are discovered, they are initialized for synchronous data exchange.

Additionally, two transceiver nodes are added: the master transceiver node, which connects to the target processor via the I2C pin, and the slave transceiver node, which connects to the network output pin "B" of the master and receives input on its network input pin "A". For the Master, two downstream and two upstream channels are configured, enabling bi-directional audio streaming. The slave node's output pin "B" is left open, indicating it is the last device in the A2B bus.

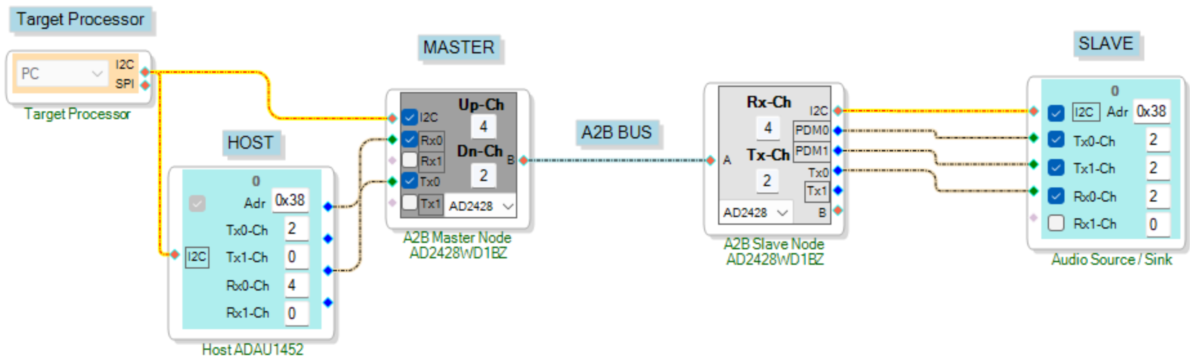

Figure 5.3: Schematic of A2B Network, showing the Master and Slave, Target Processor, Host, the A2B bus, the I2C interfaces and the transmission channel numbers

The ADAU1452 and ADAU1761 processors are connected via I2C to the Master and Slave nodes, respectively, to handle digital and analog audio streams.

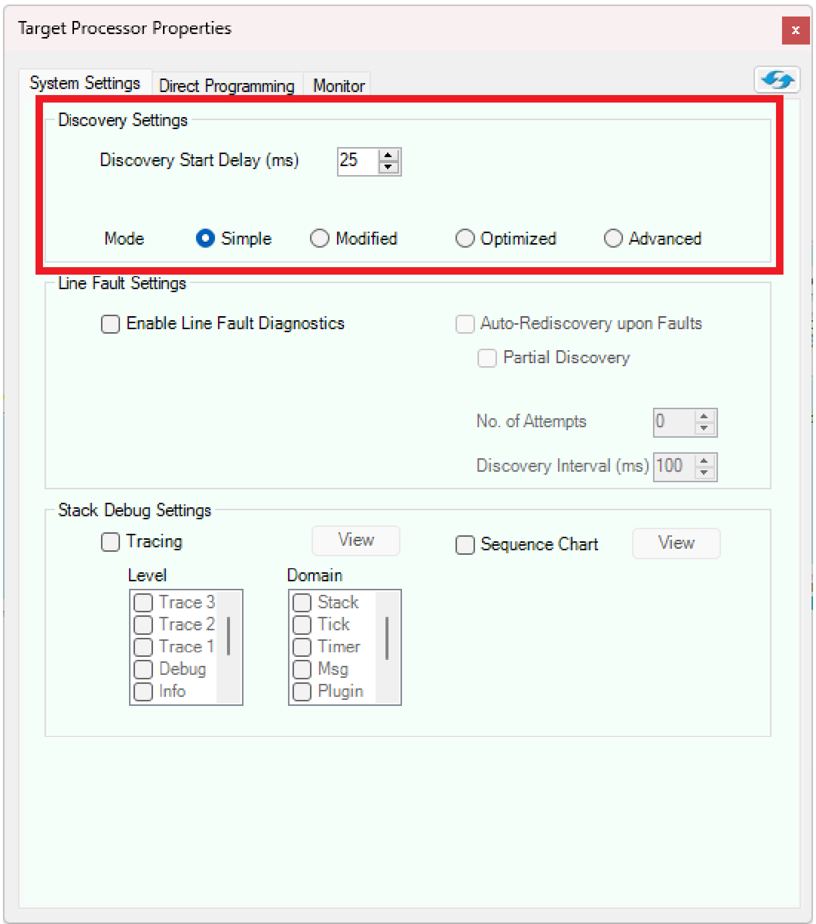

Figure 5.4: "simple" Discovery Mode Option, enabling sequential detection of slave nodes from slave 0 to the last available node in the system

5.3 Routing the Audio Stream in Master

In the Master side, the Host and Master node are configured to receive a stereo audio stream, consisting of two channels, from an external source via the SPDIF input port. Once synchronized, the audio stream is directed towards the Slave node for further processing or playback.

In addition to this, the Master node is also configured to receive a 4-channel audio stream originating from the Slave node, see Figure 5.5. This audio is generated by the on-chip PDM microphones located on the Slave board. The PDM microphones capture audio signals in a high-fidelity format, and the Slave node aggregates the data into four separate audio channels before transmitting it to the Master. Proper synchronization mechanisms and buffering are implemented to handle the timing differences and ensure the seamless integration of the incoming audio streams from both the SPDIF source and the PDM microphones. These configurations enable the system to effectively manage multiple audio input sources and provide flexibility for further digital signal processing tasks.

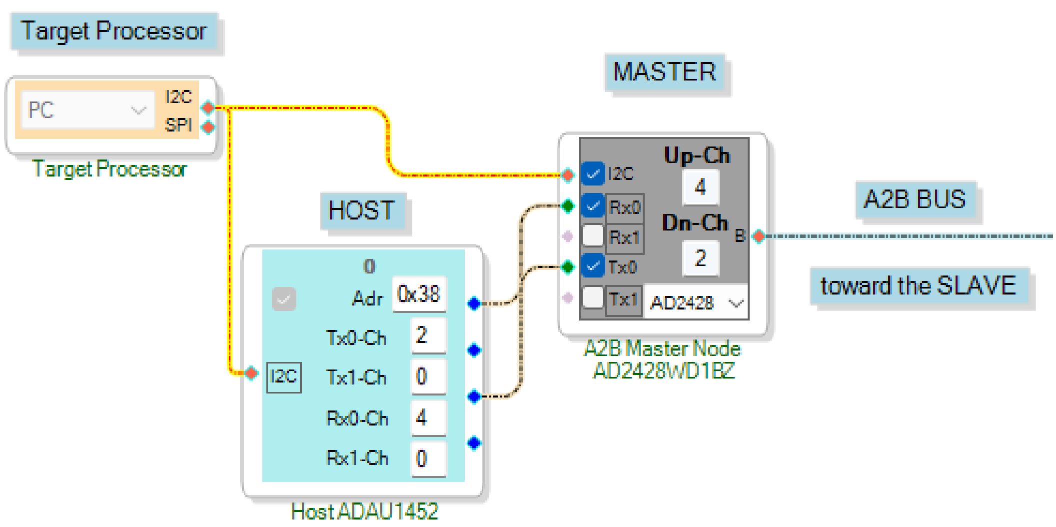

Figure 5.5: The Schematic of A2B network in Master side, showing the audio channel numbers, the I2C address of the Host (0x38) and I2C interfaces

5.3.1 Host Configuration

The ADAU1452 includes four serial data input pins (SDATA_IN0 to SDATA_IN3) and four serial data output pins (SDATA_OUT0 to SDATA_OUT3). Each pin can operate in multiple modes, supporting 2, 4, 8, or 16 audio channels, depending on the specific configuration.

In total, there are 48 channels available for both serial audio data input and output, distributed across the four input and four output pins. Specifically, the 48 audio input channels are allocated among the four serial data input pins, and similarly, the 48 audio output channels are distributed among the four serial data output pins. The detailed allocation and configuration of these channels are provided in Table 5.1.

Table 5.1: Serial Input Pin Mapping to SigmaStudio Input Cells

Serial Input Pin	Channels in SigmaStudio	MaximumTDM Channels
SDATA_IN0	Channel 0 to Channel 15	16 channels
SDATA_IN1	Channel 16 to Channel 31	16 channels
SDATA_IN2	Channel 32 to Channel 39	8 channels
SDATA_IN3	Channel 40 to Channel 47	8 channels
SDATA_OUT0	Channel 0 to Channel 15	16 channels
SDATA_OUT1	Channel 16 to Channel 31	16 channels
SDATA_OUT2	Channel 32 to Channel 39	8 channels
SDATA_OUT3	Channel 40 to Channel 47	8 channels

Figure 5.6 illustrates the mapping of input pins to their corresponding input cells within SigmaStudio, along with a visual representation of their graphical interface in the software.

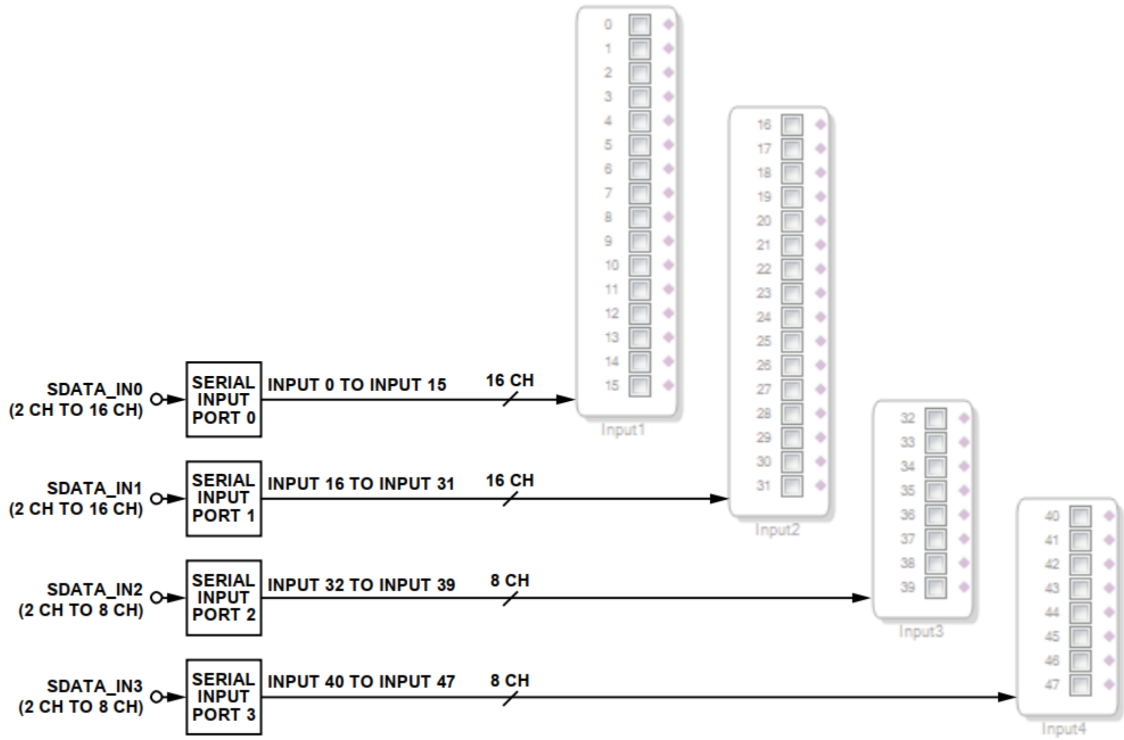

Figure 5.6: Mapping of Serial Port Audio Input from ADAU1452 to DSP in SigmaStudio, serial input 0-15 to input cell 1, serial input 16-31 to input cell 2, serial input 32-39 to input cell 3, serial input 40-47 to input cell 4,

Table 5.2 and Table 5.3 illustrates the allocation of input and output channel positions across various TDM (Time-Division Multiplexing) modes. Additionally, for the I2S interface, which supports two channels—right and left—the channel distribution follows a simpler configuration.

Table 5.2: Detailed Mapping of ADAU1452 Serial Inputs to SigmaStudio Input Channels

Serial Input Pin	Position in I2S Stream (2-Channel)	Position in TDM4 Stream	Position in TDM8 Stream	Position in TDM16 Stream	Input Channel in SigmaStudio
SDATA_IN0	Left	0	0	0	0
SDATA_IN0	Right	1	1	1	1
SDATA_IN0		2	2	2	2
SDATA_IN0		3	3	3	3
SDATA_IN0			4	4	4
SDATA_IN0			5	5	5
SDATA_IN0			6	6	6
SDATA_IN0			7	7	7
SDATA_IN0				8	8
SDATA_IN0				9	9
SDATA_IN0				10	10
SDATA_IN0				11	11
SDATA_IN0				12	12
SDATA_IN0				13	13
SDATA_IN0				14	14
SDATA_IN0				15	15
SDATA_IN1	Left	0	0	0	16
SDATA_IN1	Right	1	1	1	17
SDATA_IN1		2	2	2	18
SDATA_IN1		3	3	3	19
SDATA_IN1			4	4	20
SDATA_IN1			5	5	21
SDATA_IN1			6	6	22
SDATA_IN1			7	7	23
SDATA_IN1				8	24
SDATA_IN1				9	25
SDATA_IN1				10	26
SDATA_IN1				11	27
SDATA_IN1				12	28
SDATA_IN1				13	29
SDATA_IN1				14	30
SDATA_IN1				15	31
SDATA_IN2	Left	0	0	0	32
SDATA_IN2	Right	1	1	1	33
SDATA_IN2		2	2	2	34
SDATA_IN2		3	3	3	35
SDATA_IN2			4	4	36
SDATA_IN2			5	5	37
SDATA_IN2			6	6	38
SDATA_IN2			7	7	39
SDATA_IN3	Left	0	0	0	40
SDATA_IN3	Right	1	1	1	41
SDATA_IN3		2	2	2	42
SDATA_IN3		3	3	3	43
SDATA_IN3			4	4	44
SDATA_IN3			5	5	45
SDATA_IN3			6	6	46
SDATA_IN3			7	7	47

Table 5.3: Detailed Mapping of ADAU1452 Serial Outputs from SigmaStudio Output Channels

Input Channel in SigmaStudio	Serial Input Pin	Position in I2S Stream (2-Channel)	Position in TDM4 Stream	Position in TDM8 Stream	Position in TDM16 Stream
0	SDATA_OUT0	Left	0	0	0
1	SDATA_OUT0	Right	1	1	1
2	SDATA_OUT0		2	2	2
3	SDATA_OUT0		3	3	3
4	SDATA_OUT0			4	4
5	SDATA_OUT0			5	5
6	SDATA_OUT0			6	6
7	SDATA_OUT0			7	7
8	SDATA_OUT0				8
9	SDATA_OUT0				9
10	SDATA_OUT0				10
11	SDATA_OUT0				11
12	SDATA_OUT0				12
13	SDATA_OUT0				13
14	SDATA_OUT0				14
15	SDATA_OUT0				15
16	SDATA_OUT1	Left	0	0	0
17	SDATA_OUT1	Right	1	1	1
18	SDATA_OUT1		2	2	2
19	SDATA_OUT1		3	3	3
20	SDATA_OUT1			4	4
21	SDATA_OUT1			5	5
22	SDATA_OUT1			6	6
23	SDATA_OUT1			7	7
24	SDATA_OUT1				8
25	SDATA_OUT1				9
26	SDATA_OUT1				10
27	SDATA_OUT1				11
28	SDATA_OUT1				12
29	SDATA_OUT1				13
30	SDATA_OUT1				14
31	SDATA_OUT1				15
32	SDATA_OUT2	Left	0	0	0
33	SDATA_OUT2	Right	1	1	1
34	SDATA_OUT2		2	2	2
35	SDATA_OUT2		3	3	3
36	SDATA_OUT2			4	4
37	SDATA_OUT2			5	5
38	SDATA_OUT2			6	6
39	SDATA_OUT2			7	7
40	SDATA_OUT3	Left	0	0	0
41	SDATA_OUT3	Right	1	1	1
42	SDATA_OUT3		2	2	2
43	SDATA_OUT3		3	3	3
44	SDATA_OUT3			4	4
45	SDATA_OUT3			5	5
46	SDATA_OUT3			6	6
47	SDATA_OUT3			7	7

5.3.1.1 Receive 2-channel Asynchronous Audio From the External Source

On the Master side, the clock recovered from the S/PDIF source is not synchronized with the ADAU1452's internal clock. To achieve synchronization, the signal must pass through an Asynchronous Sample Rate Converter (ASRC) to align it with the target sample rate. Any asynchronous input can be routed to the ASRCs for resynchronization. Potential source signals for any ASRC include serial inputs, DSP-to-ASRC channels, the S/PDIF receiver, or digital PDM microphone inputs. As shown in Figure 5.7, the ADAU1452 provides eight ASRCs, each with two input and two output channels, enabling synchronization of up to 16 channels in total.

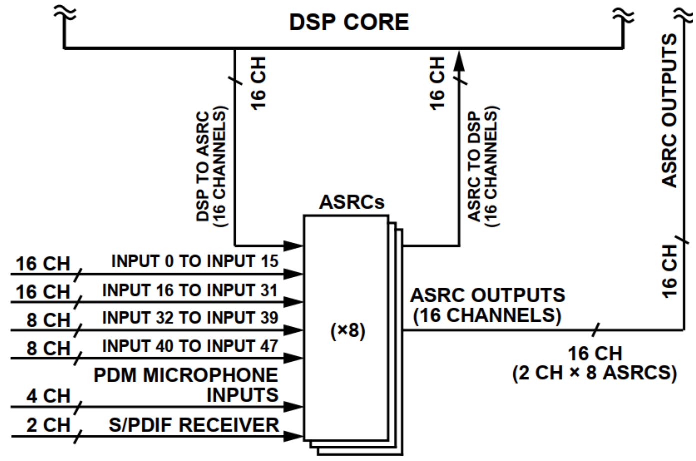

Figure 5.7: ASRC Routing for Asynchronous Input to the DSP Core, receiving data from serial input channels 0 to 47, PDM Microphone channels or SPDIF

In the current implementation, as depicted in Figure 5.8, the two channels from the S/PDIF receiver are routed through one of the ASRCs before being directed to the DSP core. To achieve this, the corresponding ASRC input selector register (Address 0xF100, ASRC_INPUT1), Bits [2:0] is set to **0b011** to select the S/PDIF receiver as the input source. Additionally, the ASRC output rate selector register (Address 0xF140, ASRC_OUT_RATE1), Bits [3:0] is configured to **0b0101** to synchronize the ASRC output data with the DSP core sample rate. These registers are defined in detail in the "Register Definitions and Configurations" section.

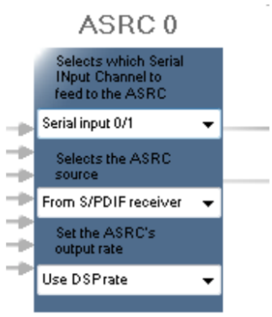

Figure 5.8: ASRC0 configuration for Asynchronous input SPDIF, configured to receive data from Serial Input 0/1 and transmitted to output with the DSP Rate

5.3.1.2 Transmit Synchronized Stereo Audio From HOST to the MASTER

In the SERIAL_PORTS tab of the Host Register window, SDATA_OUT0 is configured as shown in Figure 5.9. In this section, several parameters are defined based on the system requirements, including the following:

- **LRCLK Type**: The LRCLK (Left-Right Clock) type is configured as a "*50/50 duty cycle clock*", meaning that the clock signal is evenly split between the high and low phases. This ensures consistent synchronization of left and right channel data during transmission.
- **BCLK Polarity**: The BCLK (Bit Clock) polarity is set to "*Negative*", which specifies that data changes occur on the negative edges of the clock signal.
- **Word Length**: The S/PDIF input and output word lengths can be independently set to 16, 20, or 24 bits. In this particular configuration, the word length is set to 24 bits, allowing for high-resolution audio transmission with greater dynamic range.
- **Sampling Rate**: The sampling rate is set to FS, where FS represents the standard audio sampling frequency. This frequency is typically chosen based on the audio quality requirements and the nature of the source audio signal.
- **TDM Mode**: The TDM mode is set to "*2 channels, 32 bits/channel*". This means that two audio channels are transmitted, each with a bit depth of 32 bits. This configuration provides a high bit depth per channel, allowing for precise and high-fidelity audio representation.

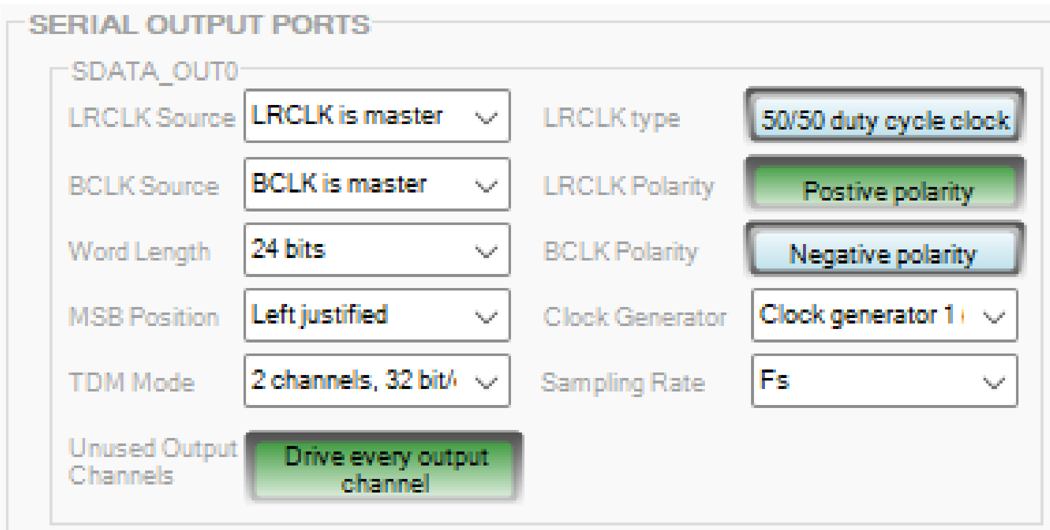

Figure 5.9: Configuration of SDATA_OUT0 of the Host, choosing the Master as the source of LRCLK and BCLK, 24-bit output Word Length, TDM2 with 32-bit width for each channel and the MSB positioned with no delay (Left Justified)

In the Host schematic workspace, the ASRC input block is added and routed to output channels 0 and 1, as shown in Figure 5.10. This routing enables the downstream transmission process from the Master to the Slave node.

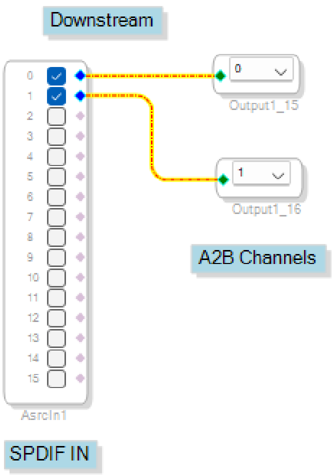

Figure 5.10: Schematic of Downstream the 2-channel synchronized audio towards the Slave

5.3.1.3 Receive 4-channel PDM Microphones Audio From the SLAVE

As shown in Figure 5.11, within the SERIAL_PORTS tab of the Host Register window, SDATA_IN0 is configured to receive a 4-channel audio stream from the Slave node. The configuration parameters for SDATA_IN0 are set according to the specific system requirements, ensuring proper synchronization and data integrity for the multi-channel audio received from the Slave.

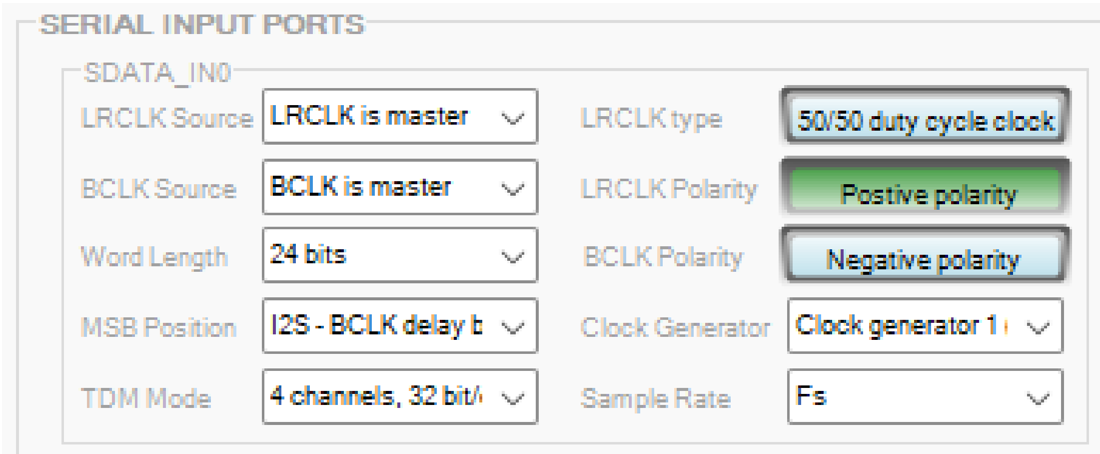

Figure 5.11: Configuration of SDATA_IN0 of the Host, choosing the Master as the source of LRCLK and BCLK, 24-bit input Word Length, TDM2 with 32-bit width for each channel and the MSB positioned with no delay (Left Justified)

In the schematic window of the Host, as shown in Figure 5.12, the 4-channel audio received from the Slave is initially combined using two signal adder blocks. The resulting 2-channel audio is then routed to output channels 40 and 41 sequentially. These output channels can be monitored and validated using headphones connected to the Master, allowing for a quick verification of the received audio signal.

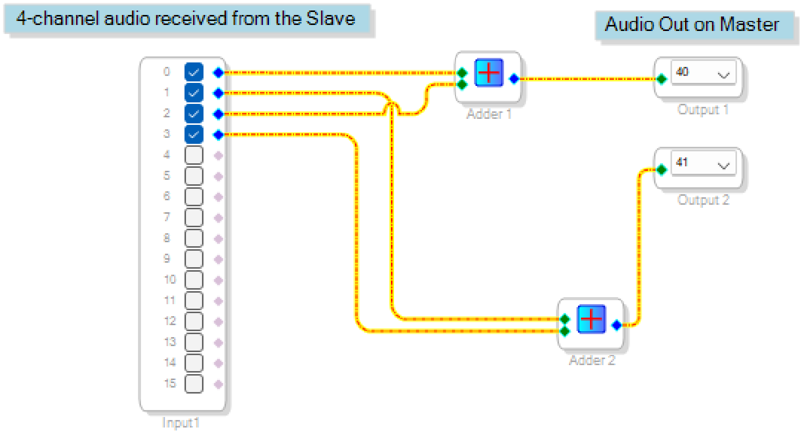

Figure 5.12: Schematic of receiving 4-channel audio from the SLAVE, using 2 Adder to combined 4-channel to stereo audio stream and transmit the result to output 40 and 41, correspond to external headphone connected to the Master

5.3.2 Master Node Configuration

The Master port is configured independently to enable both downstream and upstream communication in the A2B network. As shown in Figure 5.13, both upstream and downstream modes are activated, and a 32-bit slot size is selected for each direction in the Properties window of the Master node.

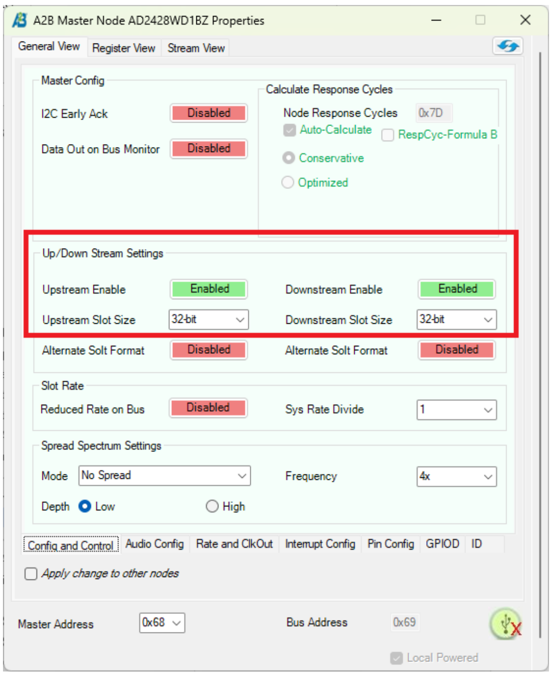

Figure 5.13: Up/Down Stream Setting on the MASTER with 32-bit slot size

As illustrated in Figure 5.14, the TDM (Time-Division Multiplexing) mode is configured as TDM2, with a channel size of 32 bits per channel. This configuration allows for two channels of audio data, each with a high bit depth of 32 bits. The synchronization mode for the TDM interface is set to a 50% duty cycle, which means the synchronization signal spends equal time in the high and low states, providing consistent timing for audio channel alignment. The polarity of the sync signal is configured to trigger on the rising edge, ensuring precise timing for data synchronization.

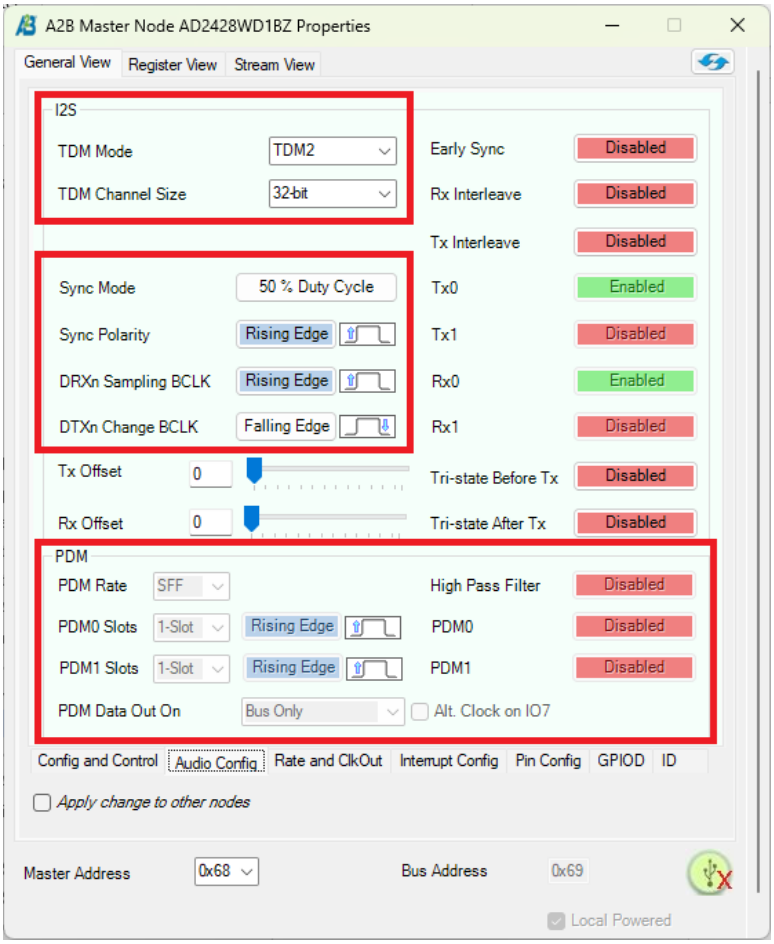

Figure 5.14: I2S/TDM Configuration on the MASTER, choosing TDM2 32-bit, 50% Duty Cycle for the Sync signal with the Rising Edge Polarity, the PDM slots setting

Additionally, the DRXn (Data Receive) sampling clock, which determines when incoming data is sampled, is set to the rising edge of the BCLK (Bit Clock) signal. This helps maintain consistent timing for receiving data and reduces the likelihood of sampling errors. Meanwhile, the DTXn (Data Transmit) change clock is set to the falling edge of BCLK, ensuring that data changes occur after each sample is taken, thus preventing overlapping or timing conflicts between transmit and receive operations.

5.3.3 Jumper Configuration on the Master Board

As shown in Figure 5.15, the jumpers on the Master board are configured to transmit a 2-channel audio signal to the Slave while receiving a 4-channel audio stream from it. Detailed information about the configuration of each jumper can be found in Section 4.2.1.1.

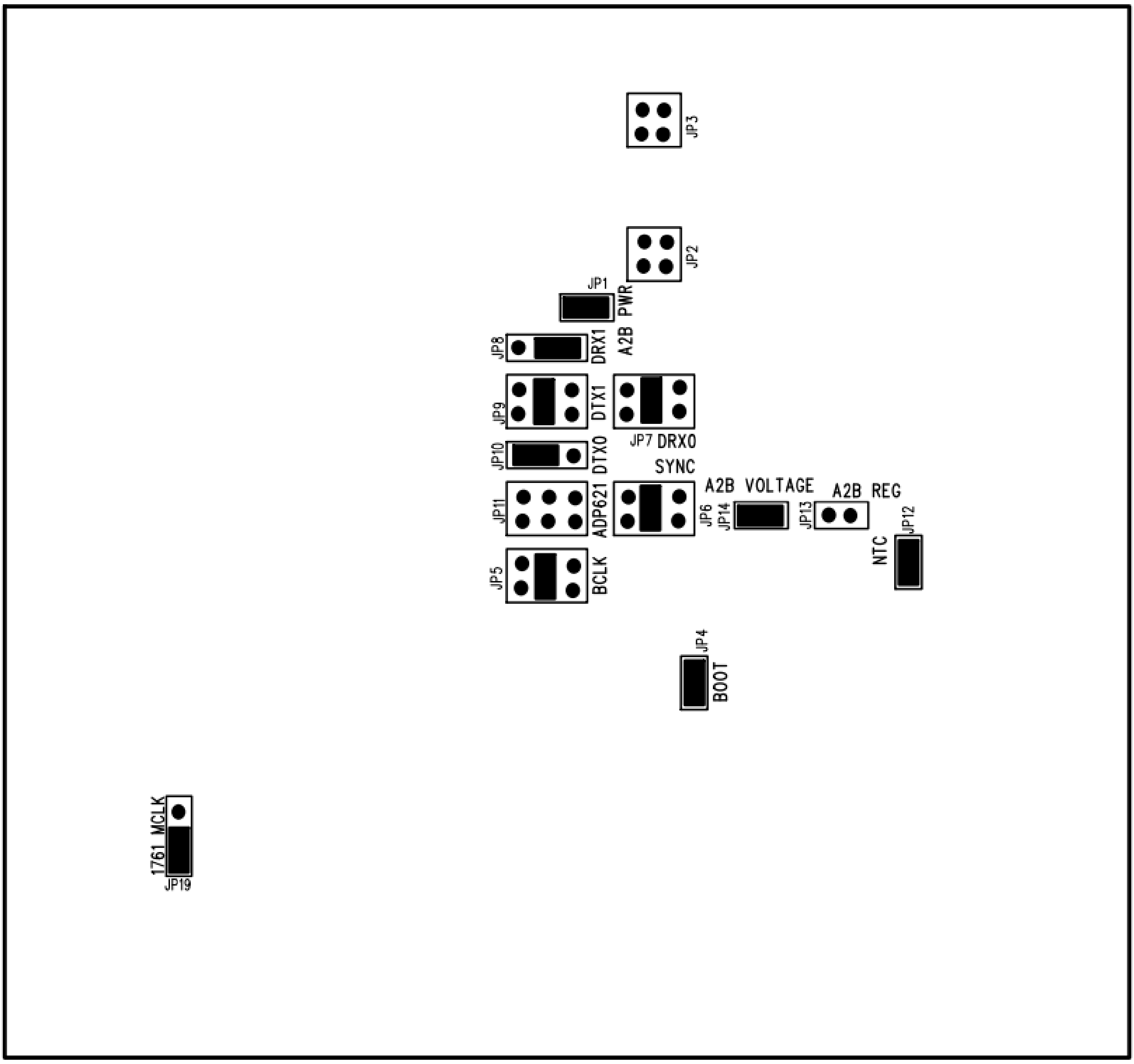

Figure 5.15: Jumper Configuration on the Master Board, installed to transmit 2-channel Downstream and support for receiving the 4-channel Upstream

It is important to note that any changes to the jumper positions will directly affect the signal routing and may lead to improper transmission or reception of audio channels. If the system design is modified, the jumper settings must be adjusted accordingly to ensure correct routing of both downstream and upstream audio signals.

5.4 Routing the Audio Stream in Slave

On the Slave side, as depicted in Figure 5.16, a Slave node is connected to the A2B bus, and an Audio Source/Sink block is interfaced with this Slave via the I2C protocol. This configuration allows for the control of the ADAU1452 on the Slave node to achieve multiple functions: receiving the stereo audio signal captured via SPDIF on the Master and transmitting the 4-channel audio generated by the on-board PDM microphones.

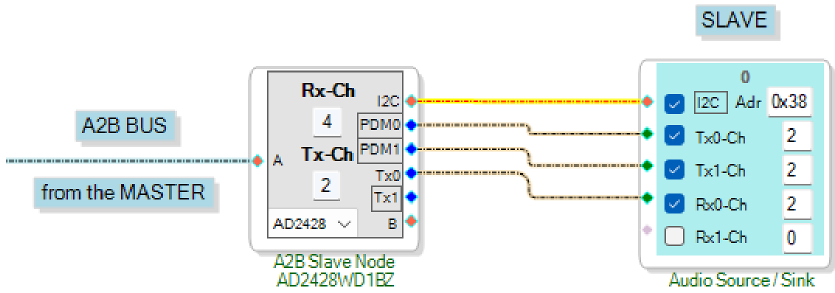

Figure 5.16: The Schematic of A2B network in Slave side, showing the local ADAU1452 as the Audio Source/Sinc, interfaced with the transceiver through I2C interface with the local address of 0x38 on Slave Side

The number of transmit (TX) and receive (RX) channels for the Slave and the Audio Source/Sink block is configured based on the network design requirements. It is important to note that on the Slave, the functions of Rx0 and Rx1 are repurposed to PDM0 and PDM1 to facilitate the transmission of audio from the PDM microphones.

A different schematic is implemented on the Slave side. As illustrated in Figure 5.17, two audio channels from the downstream are received on channels 1 and 2 of input1. These channels are then routed to output channels 40 and 41, or alternatively to the SPDIF output port, for further validation or for routing the audio signal to the vehicle amplifier.

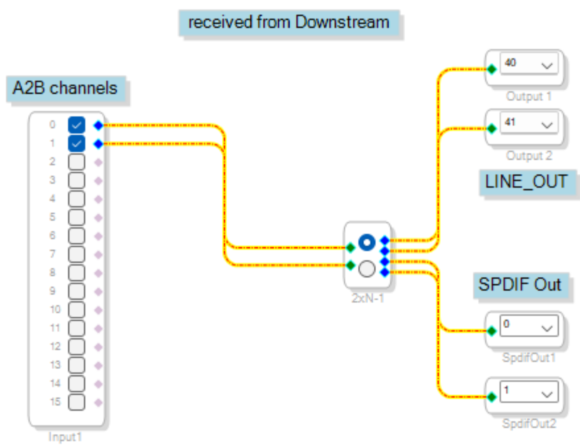

Figure 5.17: Schematic of receiving 2-channel audio from the MASTER, transmit to output channel 40 and 41 (Headphone connected to Slave) or to SPDIF channels (Captured from the SPDIF-out on Slave)

For the PDM streams, since they are routed directly to the Master by setting the appropriate jumpers on the Slave board, there is no need to implement any additional schematic. This direct routing simplifies the setup and allows the PDM audio signals to be efficiently transmitted without further configuration.

5.4.1 Jumper Configuration on the Master Board

The jumpers on the Slave board are configured as shown in Figure 5.18. Detailed information about the configuration of each jumper is provided in Section 4.2.1.1.

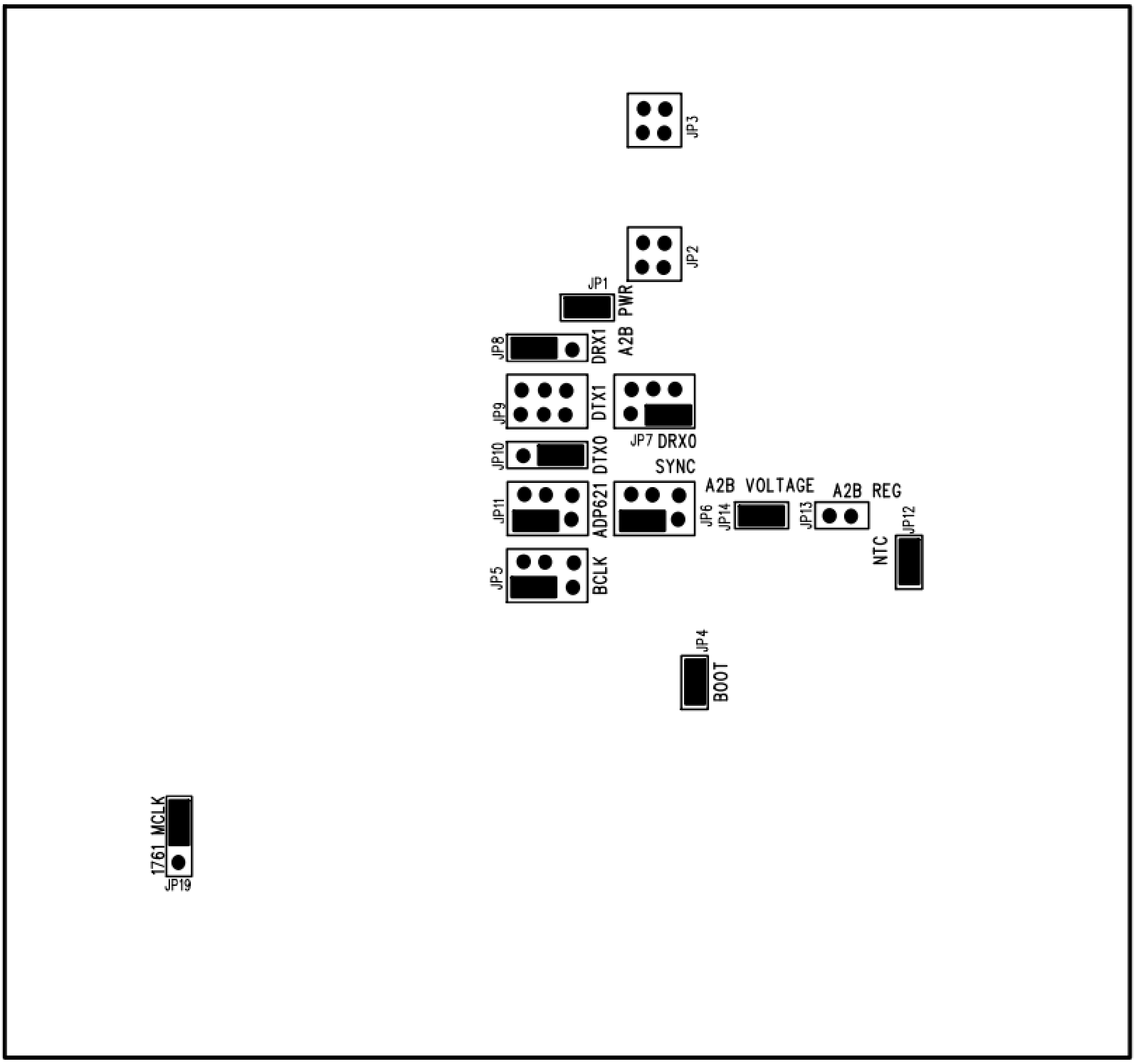

Figure 5.18: Jumper Configuration on the Slave Board, installed to transmit 4-channel Upstream and support for receiving the 2-channel Downstream from the Master

5.5 Registers Definitions and Configuration

5.5.1 ADAU1452 Registers

5.5.1.1 ASRC Input Selector Register

Address: 0xF100 to 0xF107

These eight registers configure the input signals for the corresponding eight stereo ASRCs on the ADAU1452, as shown in Figure 5.19. ASRC_INPUT0 configures ASRC Channels 0 and 1, ASRC_INPUT1 configures ASRC Channels 2 and 3, and so forth. The valid input signals for the ASRCs include Serial Input Channels 0 through 47, PDM Microphone Input Channels 0 through 3, and S/PDIF Receiver Channels 0 and 1.

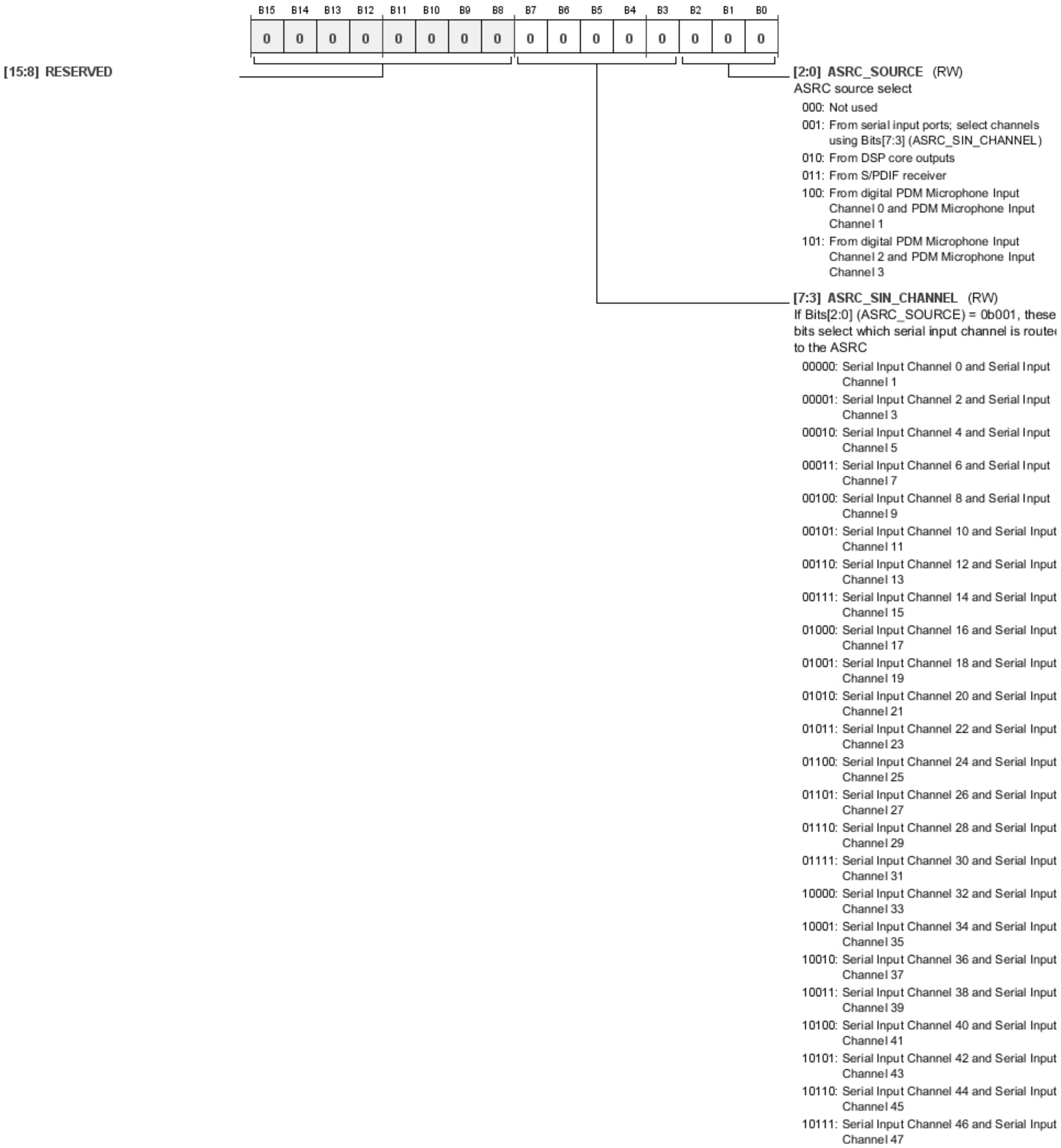

Figure 5.19: ADAU1452, ASRC Input Selector Registers

5.5.1.2 ASRC Output Rate Selector Register

Address: 0xF140 to 0xF147

These eight registers configure the target output sample rates for the corresponding eight stereo ASRCs on the ADAU1452, as shown in Figure 5.20. The ASRC takes an arbitrary input sample rate and automatically attempts to resample the data to match the target sample rate specified by these registers. Each register corresponds to one of the eight stereo ASRCs. The ASRCs align their output frequencies with the audio sample rates of any of the serial output ports, the DSP core's start pulse rate, or one of several internally generated sample rates provided by the clock generators.

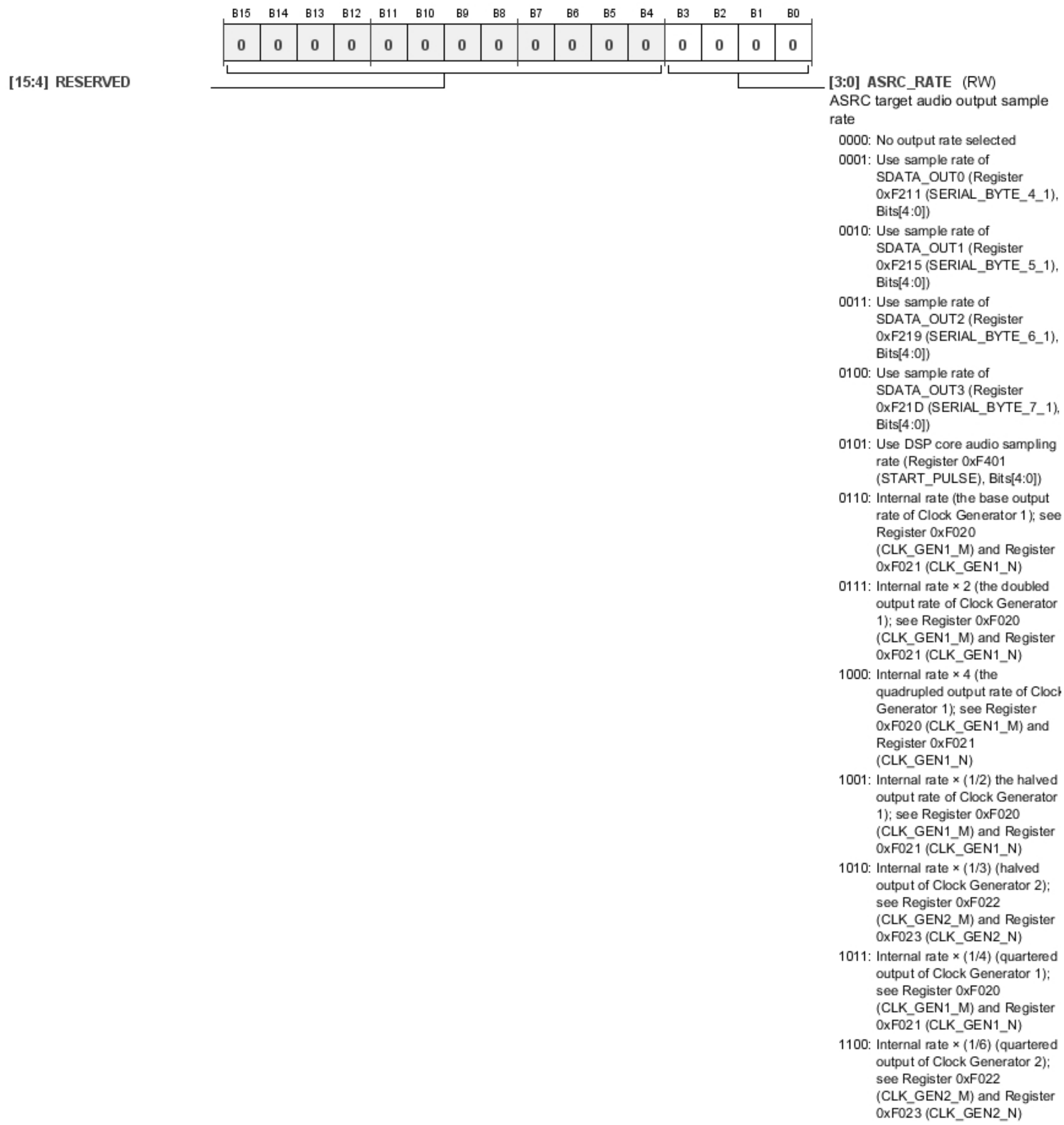

Figure 5.20: ADAU1452, ASRC Output Selector Register

5.5.1.3 Automatically Resume S/PDIF Receiver Audio Input Register

Address: 0xF604

When the S/PDIF receiver loses synchronization with the incoming S/PDIF signal, often caused by issues with signal quality, the receiver automatically mutes. This register controls whether the S/PDIF receiver will automatically resume data output once it starts receiving a valid signal again and regains lock. By default, the S/PDIF receiver will not resume audio automatically after losing lock, and the user must manually reset it by toggling Register 0xF604 (SPDIF_RESTART), Bit 0 (RESTART_AUDIO), from 0b0 to 0b1 and then back to 0b0. To make sure that the S/PDIF receiver always resumes output when a valid signal is detected, register 0xF604 (SPDIF_RESTART), Bit 0 (RESTART_AUDIO), should be set to 0b1 permanently. The details are shown in Figure 5.21.

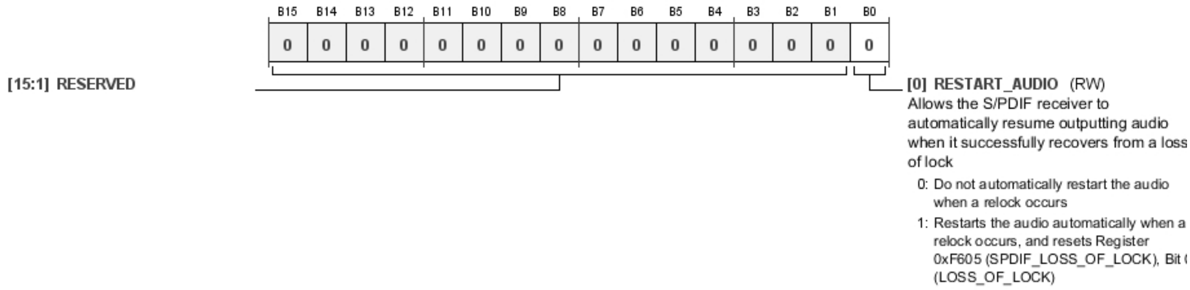

Figure 5.21: ADAU1452, Automatically Resume S/PDIF Receiver Audio Input Register

5.5.1.4 S/PDIF Transmitter Data Selector Register

Address: 0xF1C0

This register determines the data source that supplies the S/PDIF transmitter. As illustrated in Figure 5.22, The data can come from either the DSP core's S/PDIF outputs or directly from the S/PDIF receiver.

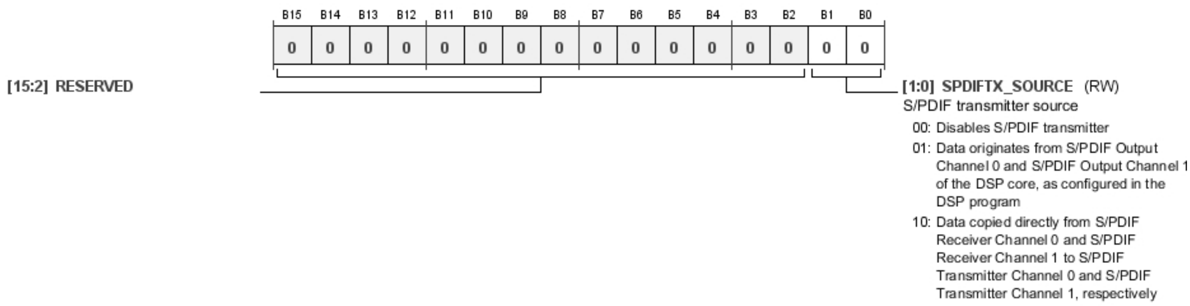

Figure 5.22: ADAU1452, S/PDIF Transmitter Data Selector Register

5.5.1.5 Serial Port Control 0 Register
Address: 0xF200 to 0xF21C, (Increments of 0x4)

These eight registers are used to configure various settings for the corresponding serial input and output ports. The channel count, MSB position, data word length, clock polarity, clock sources, and clock type are all set using these registers, as shown in Figure 5.23. For the input ports, register 0xF200 (SERIAL_BYTE_0_0) is linked to SDATA_IN0, register 0xF204 (SERIAL_BYTE_1_0) to SDATA_IN1, register 0xF208 (SERIAL_BYTE_2_0) to SDATA_IN2, and register 0xF20C (SERIAL_BYTE_3_0) to SDATA_IN3. On the output side, register 0xF210 (SERIAL_BYTE_4_0) corresponds to SDATA_OUT0, register 0xF214 (SERIAL_BYTE_5_0) to SDATA_OUT1, register 0xF218 (SERIAL_BYTE_6_0) to SDATA_OUT2, and register 0xF21C (SERIAL_BYTE_7_0) to SDATA_OUT3.

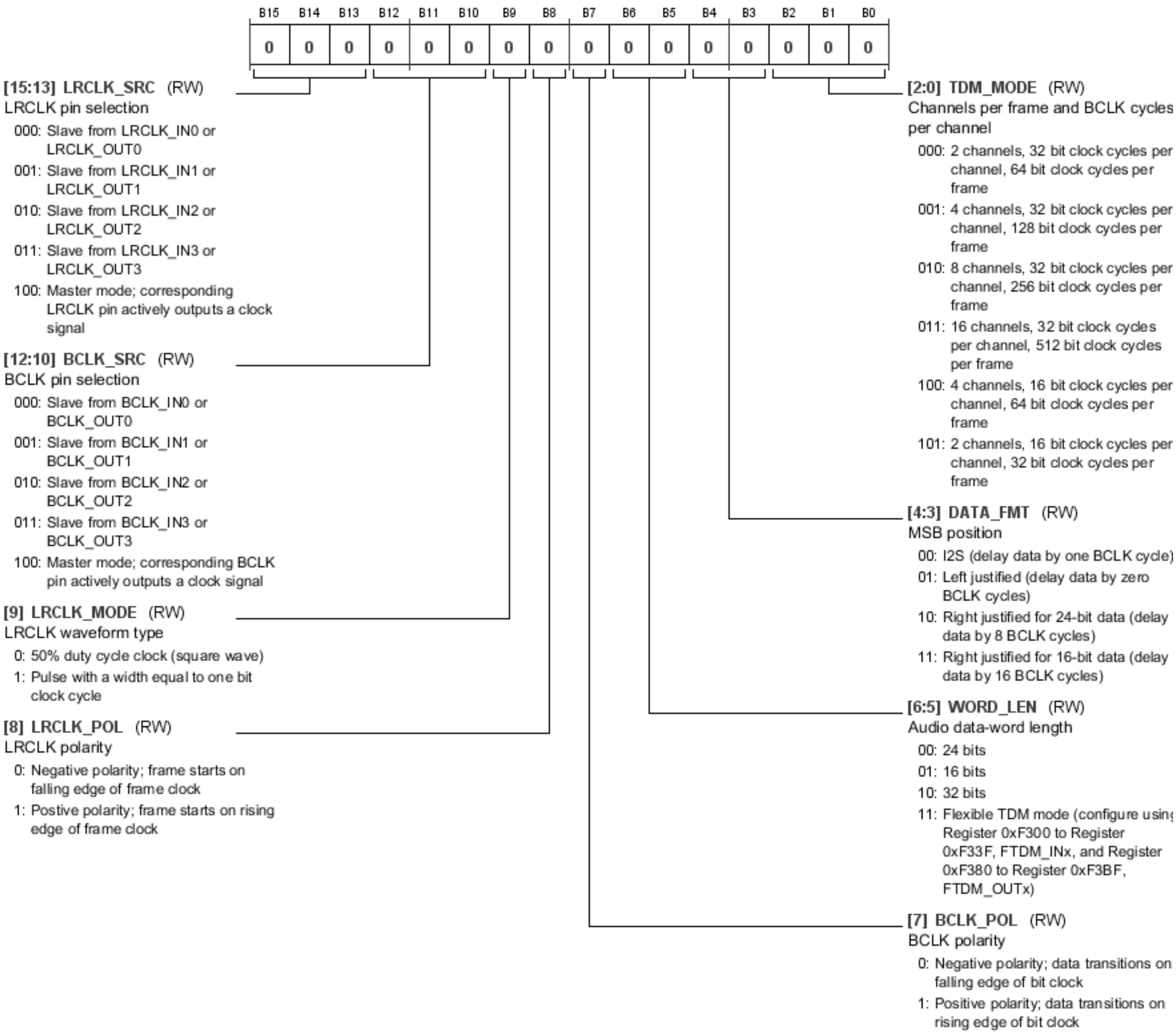

Figure 5.23: ADAU1452, Serial Port Control 0 Register

5.5.1.6 Serial Port Control 1 Register

Address: 0xF201 to 0xF21D, (Increments of 0x4)

These eight registers are used to configure several settings for the corresponding serial input and output ports. These settings include the clock generator, sample rate, and behavior for inactive channels. For the input ports, register 0xF201 (SERIAL_BYTE_0_1) is associated with SDATA_IN0, register 0xF205 (SERIAL_BYTE_1_1) with SDATA_IN1, register 0xF209 (SERIAL_BYTE_2_1) with SDATA_IN2, and register 0xF20D (SERIAL_BYTE_3_1) with SDATA_IN3. For the output ports, register 0xF211 (SERIAL_BYTE_4_1) corresponds to SDATA_OUT0, register 0xF215 (SERIAL_BYTE_5_1) to SDATA_OUT1, register 0xF219 (SERIAL_BYTE_6_1) to SDATA_OUT2, and register 0xF21D (SERIAL_BYTE_7_1) to SDATA_OUT3. Further details can be found in Figure 5.24.

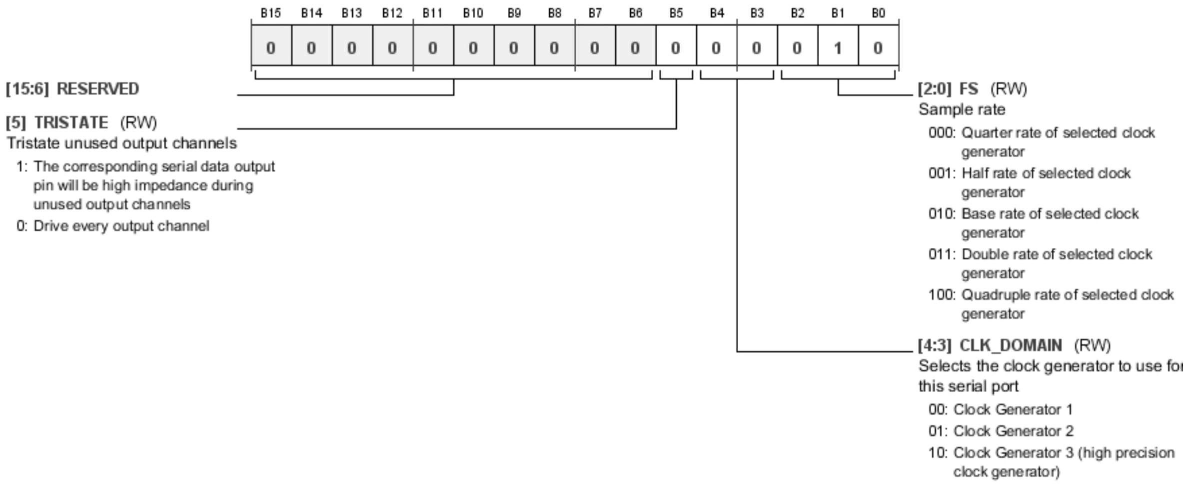

Figure 5.24: ADAU1452, Serial Port Control 1 Register

5.5.1.7 Digital PDM Microphone Control Register

Address: 0xF560 to 0xF561

These registers are used to configure the digital PDM microphone interface, as illustrated in Figure 5.25. Specifically, two registers are employed to manage up to four PDM microphones, providing flexible control over the connected devices. Register 0xF560 (DMIC_CTRL0) is responsible for configuring PDM Microphone Channels 0 and 1, while Register 0xF561 (DMIC_CTRL1) handles PDM Microphone Channels 2 and 3. Each register allows for fine-tuning of microphone parameters such as gain control, activation status, and data synchronization, ensuring optimal performance for the connected PDM microphones.

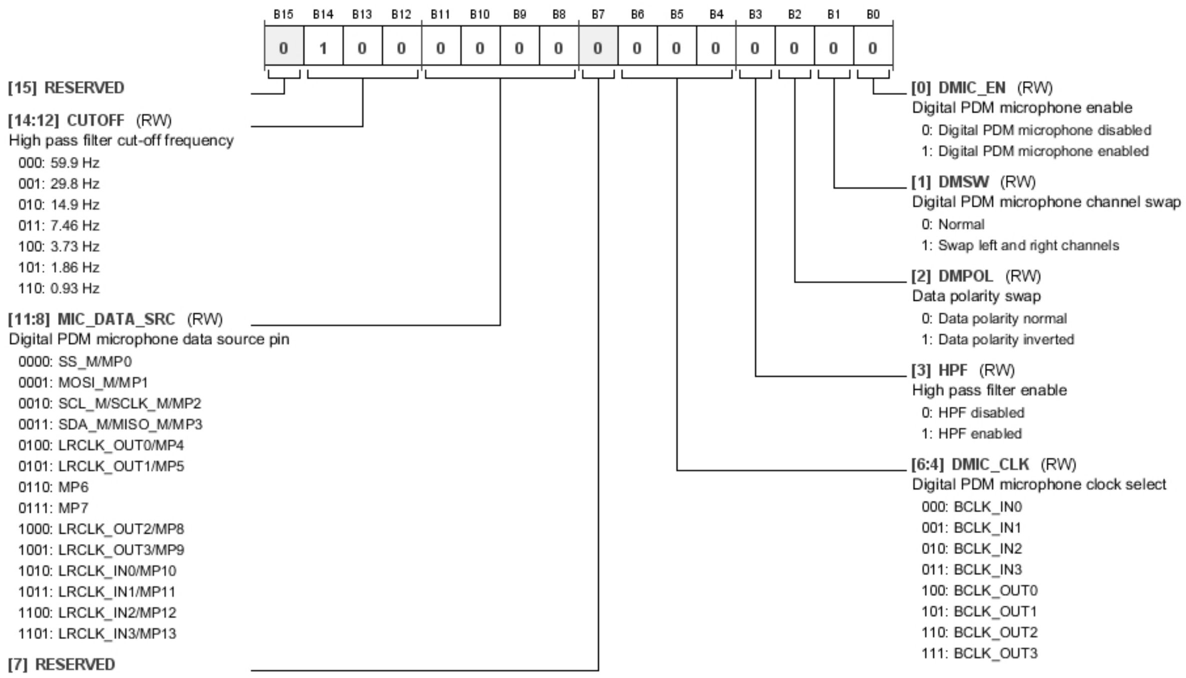

Figure 5.25: ADAU1452, Digital PDM Microphone Control Register

5.5.2 AD2428 Registers

5.5.2.1 Downstream Slots Register

Address: 0x0D

In the main node, the A2B_DNSLOTS register specifies the total number of downstream data slots, including broadcast slots. In a subordinate node, this register indicates the number of data slots (excluding broadcast slots) that are forwarded downstream (B-side) once the transceiver starts capturing data slots. As illustrated in Figure 5.26, valid programming values range from 0 to 32.

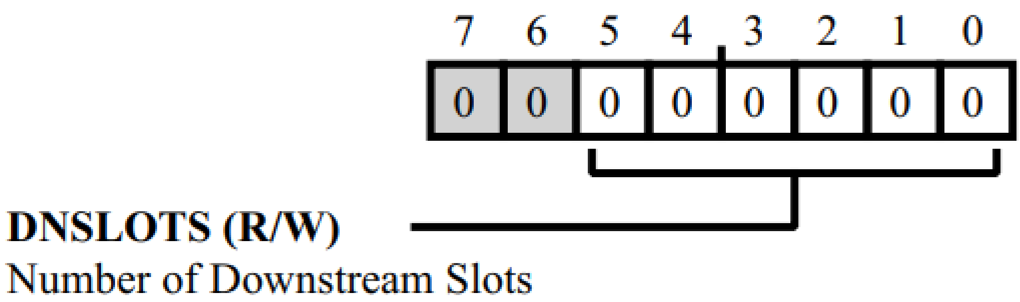

Figure 5.26: AD2428, Downstream Slots Register

5.5.2.2 Upstream Slots Register

Address: 0x0E

In the main node, the A2B_UPSLOTS register specifies the total number of upstream data slots. In a subordinate node, this register defines the number of data slots that are forwarded upstream by the B-side transceiver before it starts adding additional data slots. As illustrated in Figure 5.27,The valid programming range is from 0 to 32.

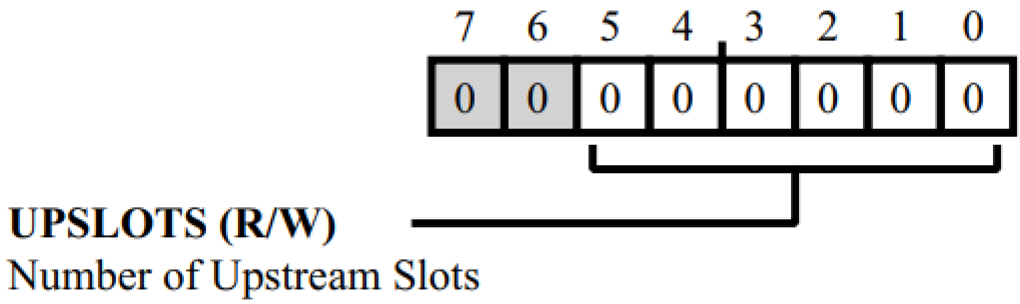

Figure 5.27: AD2428, Upstream Slots Register

5.5.2.3 Data Control Register

Address: 0x11

The A2B_DATCTL register is utilized to enable data slots and standby mode on the A2B bus. Any changes made to this register are effective only in the main node. When updated in the main node, the new configuration is automatically broadcast to all discovered subordinate nodes via the A2B bus. Local host writes to this register in a subordinate node are ignored and have no impact. This register is shown in Figure 5.28.

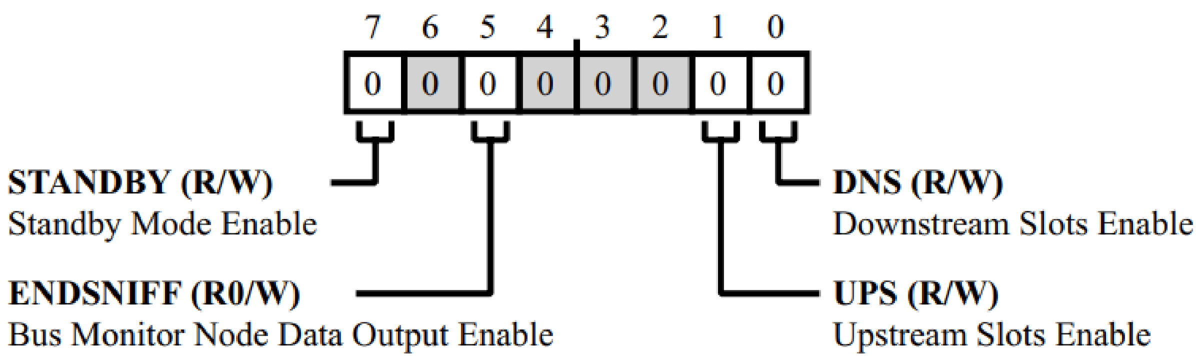

Figure 5.28: AD2428, Data Control Register

5.5.2.4 Interrupt Status Register

Address: 0x15

The A2B_INTSTAT register holds the interrupt status information for the node. When the A2B_INTSTAT.IRQ bit is set, it indicates that the node is signaling an interrupt request, either via the IRQ pin for a main node or through the A2B bus for a subordinate node, see Figure 5.29.

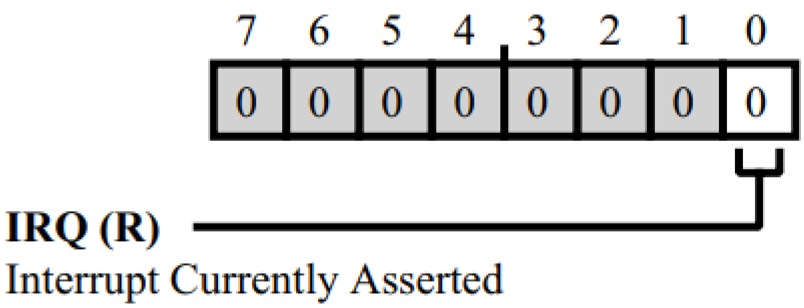

Figure 5.29: AD2428, Interrupt Status Register

5.5.2.5 Interrupt Source Register (Main Node Only)

Address: 0x16

The A2B_INTSRC register provides information about the current highest priority interrupt. If the value in this register is 0x00, it indicates that no interrupts are active. When the A2B_INTSRC.MSTINT bit is set, the interrupt is being generated by the main node. Conversely, if the A2B_INTSRC.SLVINT bit is set, the interrupt is originating from a subordinate node. The A2B_INTSRC.INODE bit field specifies the node number of the subordinate node that triggered the current interrupt. Additional details can be found in Figure 5.30.

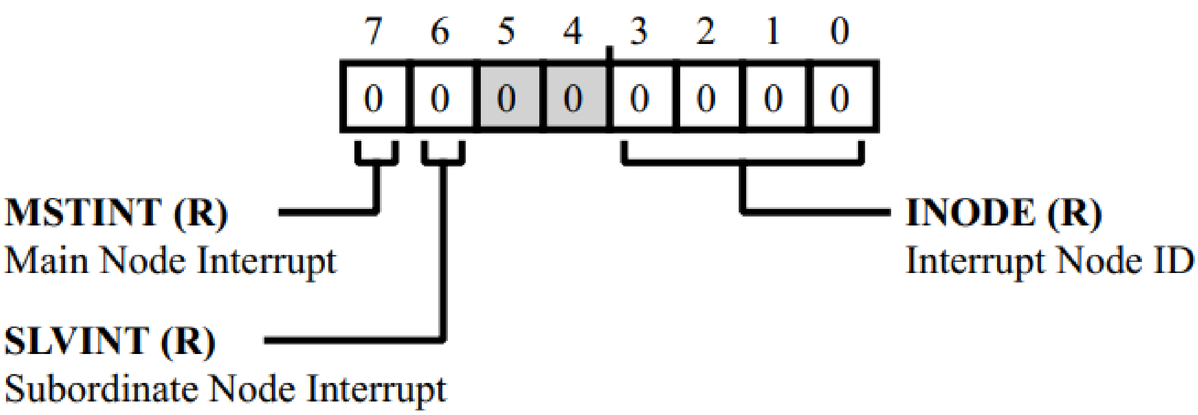

Figure 5.30: AD2428, Interrupt Source Register (Main Node Only)

Chapter 6

Testing the A2B Network on the Tesla Model 3

To evaluate the bidirectional behavior of the Automotive Audio Bus (A2B) network, comprehensive testing was carried out on a Tesla Model 3, which is equipped with the Premium-Sound-System. This evaluation was conducted at 2Electronic company, leveraging the capabilities of Analog Devices' A2B technology, which the Tesla Model 3 integrates into its advanced audio infrastructure.

6.1 A2B Network Implementation in Tesla Model 3

The Tesla Model 3's premium sound system employs A2B technology by Analog Devices Inc. The A2B network designed for this test consists of a single Master node and two distinct Slave nodes. As depicted in Figure 6.1, both the Master and Slave node 1 are represented by EVAL-AD2428WD1BZ evaluation boards.

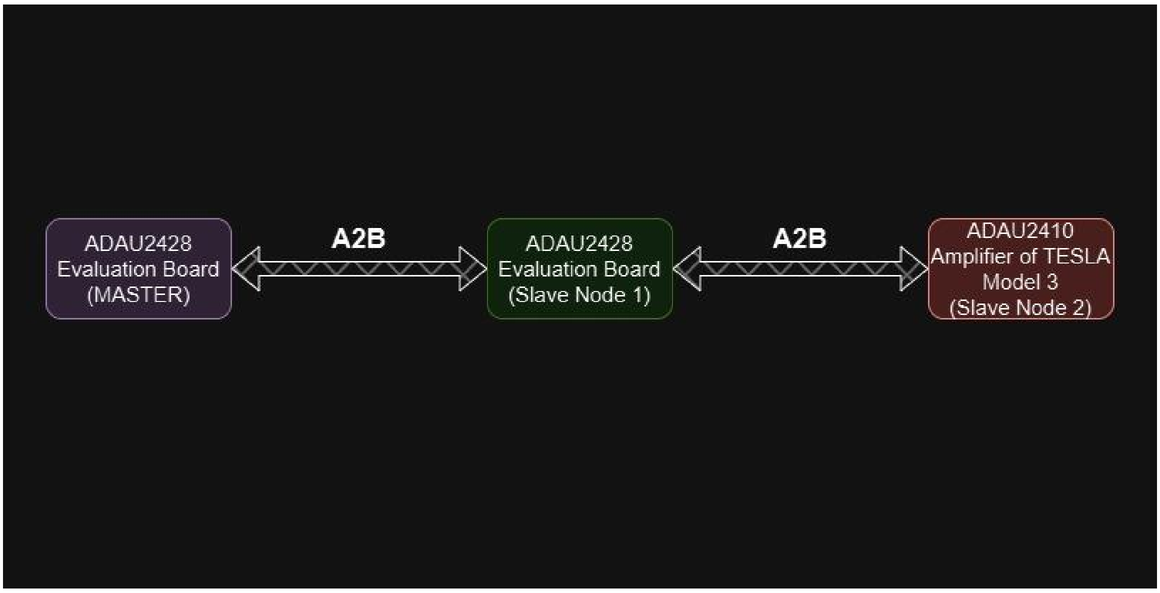

Figure 6.1: A2B Network Implementation in Tesla Model 3, consist of two AD2428WD1BZ evaluation boards as the Master and Slave node 1 and one ADAU2410 (Tesla Model 3 Original Amplifier)

The Tesla's original amplifier unit, located in the trunk of the vehicle, serves as Slave node 2. This amplifier has an embedded AD2410 chip that enables the A2B functionality, allowing it to integrate with the audio bus network. The role of this amplifier is critical, as it receives digital audio signals via the A2B bus and subsequently converts these signals into analog output, which drives the Tesla's speaker system.

6.2 Network Architecture and Components

The network architecture implemented consists of three nodes: a Master node and two Slave nodes. The Master node functions as the control unit, receiving the audio input and managing communication between nodes in the network. Specifically, the Master node in this implementation is responsible for receiving a two-channel stereo audio signal from a mini-PC through the SPDIF interface. To ensure that the received audio signal is correctly synchronized with the internal DSP clock of the A2B system, an Asynchronous Sample Rate Converter (ASRC1) is employed. This component synchronizes the incoming digital audio to the clock domain of the A2B system, minimizing timing mismatches and jitters.

Once synchronized, the audio is processed and converted into a four-channel digital output, which is transmitted to Slave node 2 for playback. This Slave node, which is the amplifier located in the trunk, subsequently converts the digital audio data into analog signals, driving four separate speakers within the vehicle. The schematic illustrating this configuration can be seen in Figure 6.2.

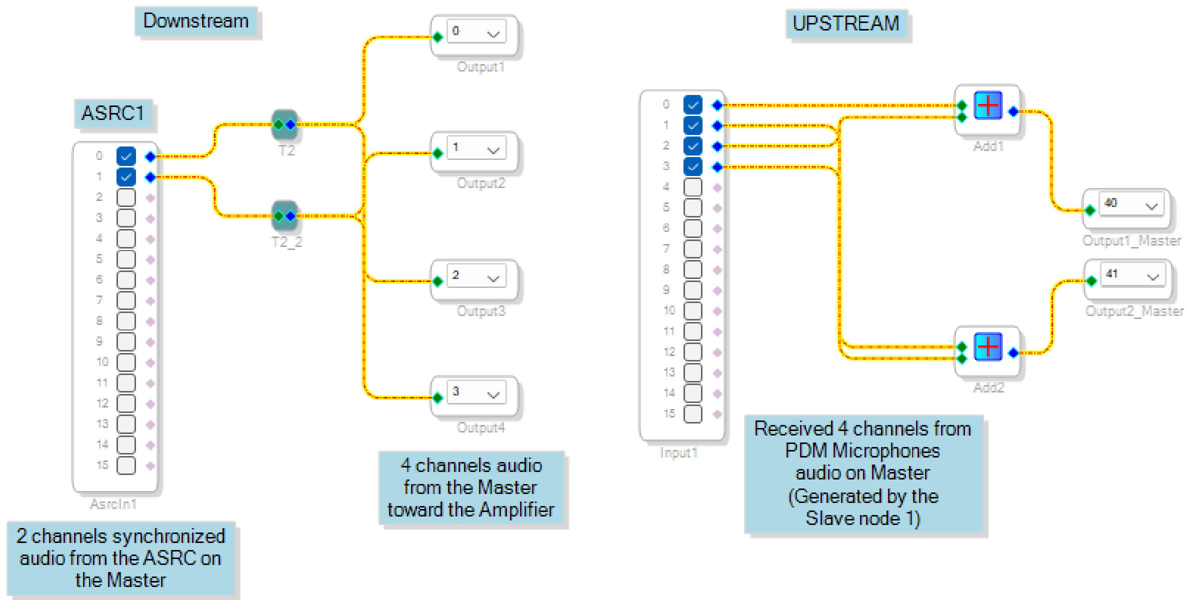

Figure 6.2: Configuration on the Master, Downstream 4-channels received by the ASRC1 to the output channel 0 to 4 toward the Slave, received 4-channel audio from the Slave on serial input 0 to 4 from the Upstream

Slave node 1, on the other hand, serves a different function in this network. It captures environmental sounds within the vehicle through its four on-board PDM microphones. The PDM (Pulse Density Modulation) audio signals are captured and then converted into PCM (Pulse Code Modulation) format before being transmitted back to the Master node via four audio slots. This allows the Master node to process environmental audio, which can be used for a variety of applications, such as active noise cancellation, voice recognition, or acoustic analysis within the cabin.

The complete schematic of the A2B network, including the data paths for both playback and microphone audio capture, is depicted in Figure 6.3.

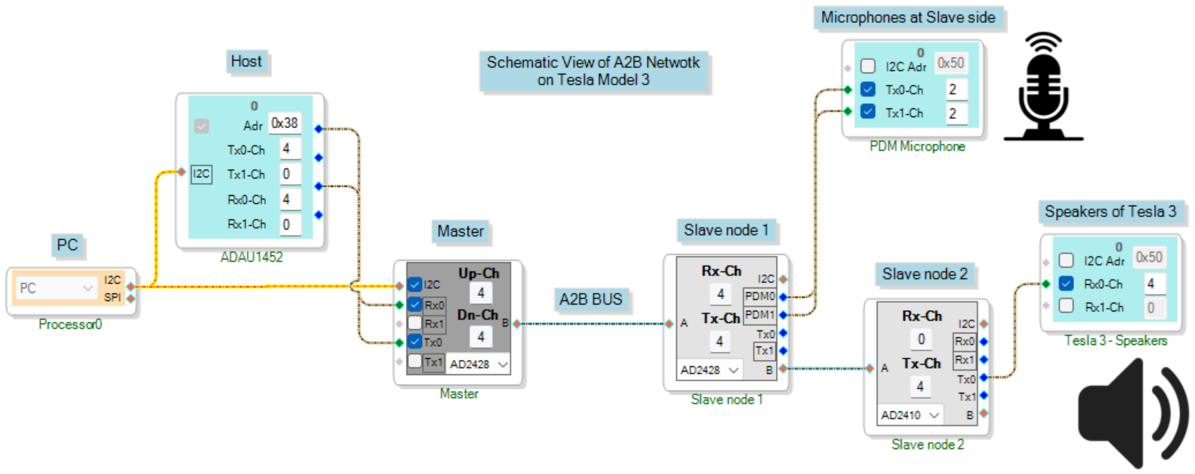

Figure 6.3: Schematic of A2B Network on Tesla Model 3 , showing the Master, Slave node 1 and Slave node 2, the Host and Target Processor, the transmitted channels and the A2B bus

6.3 Signal Flow

In the implemented A2B network, the audio signal flow starts with the Master node receiving a two-channel stereo input. This input is provided via SPDIF from a mini-PC, representing a common scenario for media playback or infotainment. After passing through the ASRC1 for clock synchronization, the audio is expanded from two channels to four channels to utilize the Tesla Model 3's speaker setup. This is critical for creating an immersive sound experience within the vehicle, allowing for enhanced spatial audio playback.

The synchronized and converted four-channel audio is sent to Slave node 2. At Slave node 2, which is the amplifier with an embedded AD2410 chip, the incoming digital audio data is processed and converted into analog signals. These analog signals are then transmitted to four individual speakers within the Tesla Model 3. This distribution allows for a balanced and high-quality audio experience for the occupants.

The role of Slave node 1 differs in that it is tasked with capturing environmental audio data through four PDM microphones integrated into its circuitry. These microphones capture sounds such as passenger voices or ambient cabin noise. The captured signals are then converted from PDM to PCM format for transmission back to the Master node. This process is instrumental in scenarios involving voice commands or noise cancellation, where real-time audio capture and processing are essential.

The audio flow in this configuration emphasizes the bidirectional nature of the A2B bus. Not only is audio transmitted from the Master to the Slave nodes for playback, but environmental audio is also captured and sent back from a Slave node to the Master, demonstrating the versatility of the A2B system in handling both playback and audio capture concurrently. The signal flow is shown in Figure 6.4.

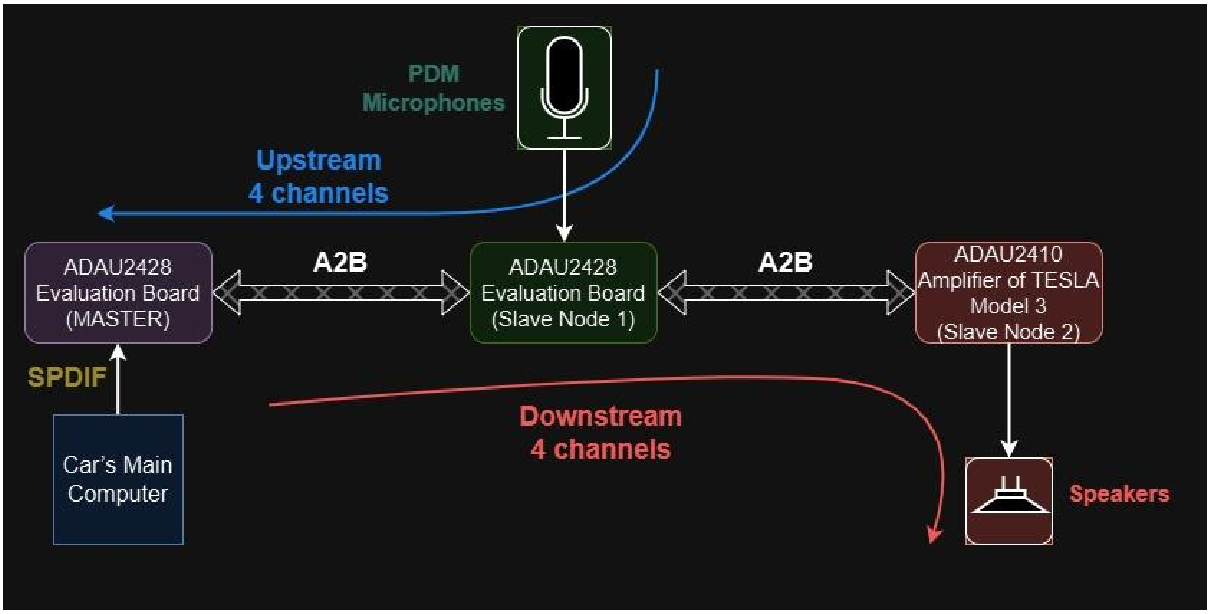

Figure 6.4: Signal Flow on A2B Network Implementation in Tesla Model 3, showing the PDM Microphones connected to Slave node 1, The Tesla Model 3 speakers connected to Slave node 2, the Upstream and Downstream

6.4 Audio Stream Definition and Assignment in SigmaStudio

As illustrated in Figure 6.5, two separate audio data streams are configured in the SigmaStudio software environment for this implementation. These streams enable communication between the Master and Slave nodes, each serving distinct roles for audio transmission and ensuring a seamless flow of data. The first stream, named "Speaker", is a **downstream** transmission carrying audio data from the Master node to Slave node 2. This particular stream is configured as a 4-channel output, meaning that it is capable of transmitting four distinct audio channels simultaneously. The data width for each channel is defined as 24 bits, which provides a good balance between audio quality and bandwidth requirements, allowing for high-resolution audio to be transmitted. The sampling frequency (Fs) for this stream is set at 48 kHz, which is a standard sample rate widely used in professional audio systems to ensure high-quality audio reproduction.

The second stream, meanwhile, is an **upstream** transmission originating from Slave node 1 and directed towards the Master node. Like the downstream stream, this upstream path is also configured with four channels, providing the capability to send four separate channels of audio data from the Slave back to the Master. The data width for this stream is also set to 24 bits, and the sampling frequency is maintained at 48 kHz to ensure consistency across both streams. This consistency in configuration simplifies synchronization and ensures uniform audio quality across both directions of transmission, making it easier to manage the overall system performance.

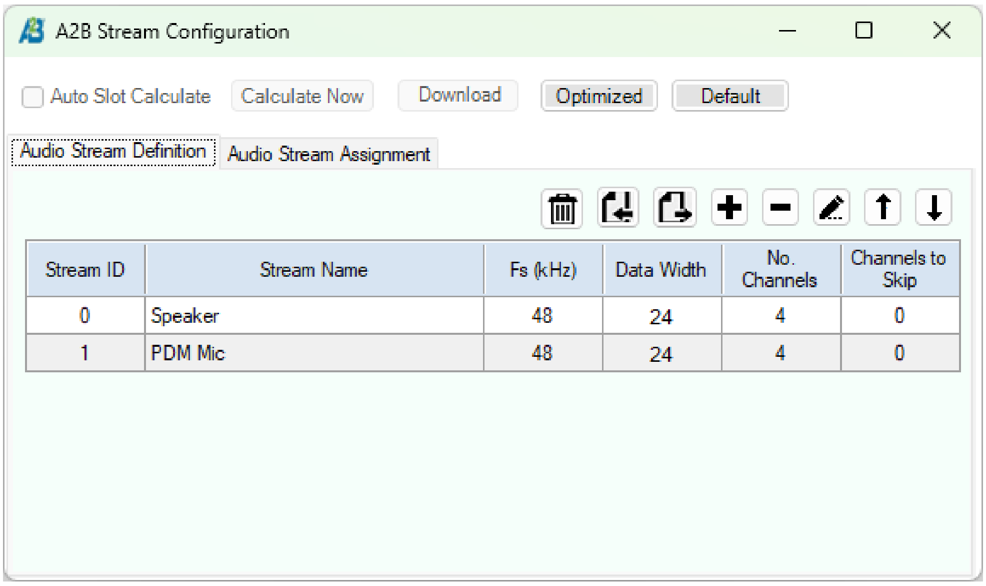

Figure 6.5: Audio Stream Definition in SigmaStudio Software, 4-channel downstream (Speaker) and upstream (PDM Mic), each with 24-bit of Data Width and Sampling Frequency (Fs) of 48Khz

After defining the necessary streams, the next step involves assigning these streams to the appropriate nodes within the SigmaStudio software. Stream assignment is crucial as it specifies how audio data is routed through the system, enabling the correct nodes to either transmit or receive the intended signals. The SigmaStudio interface allows users to visually configure and assign these streams to different nodes, ensuring that each stream is properly connected to the designated source or destination. Figure 6.6 provides a visual representation of the stream assignment process for this implementation, showing how the downstream and upstream channels are mapped between the Master and Slave nodes.

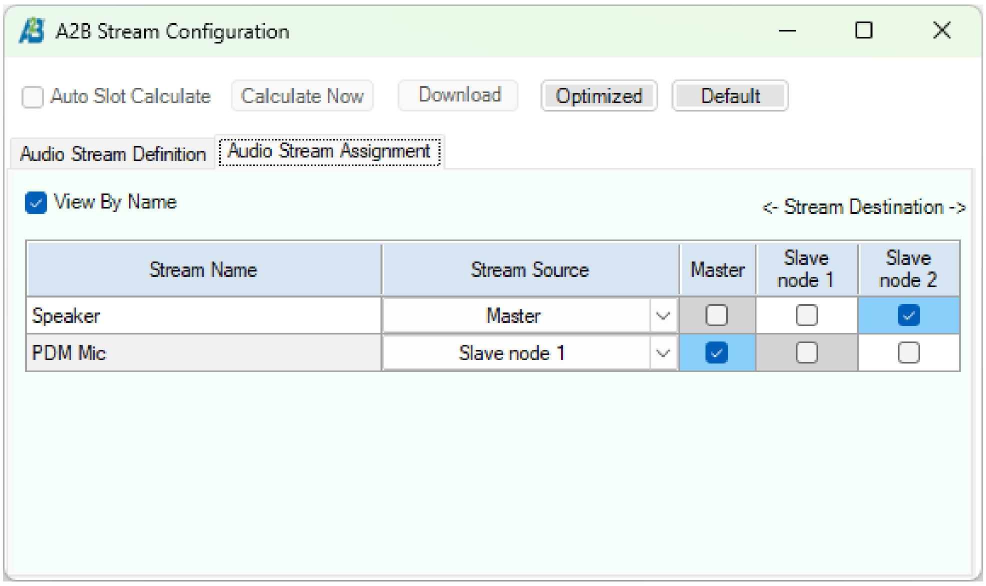

Figure 6.6: Audio Stream Assignment in SigmaStudio Software, showing Downstream from the Master to the Slave node 2, and Upstream from Slave node 1 to the Master

Chapter 7

Results, Discussion, Validation and Testing

This chapter presents the results from the DSP system design and implementation, discusses key findings, validates the system through extensive testing, and evaluates its overall performance in an automotive audio environment. Following the setup in SigmaStudio and the configuration of the corresponding registers, multiple tests were conducted to ensure accurate signal processing and reliable communication across the A2B network. Each stage—from initial configuration to the processing of audio data—was evaluated against the desired operational criteria to confirm system effectiveness.

7.1 Result

The implementation of the Automotive Audio Bus (A2B) network within the automotive audio system achieved several notable performance milestones. The system's key goal was to establish bi-directional audio streaming between Master and Slave nodes with a focus on minimizing wiring complexity while maintaining audio quality and synchronization. The following results were obtained during the testing phase, focusing on system stability, audio quality, synchronization, latency, and overall scalability.

7.1.1 Consistent Audio Quality

The A2B network demonstrated high-quality audio transmission in both downstream and upstream directions. Audio signals routed from the vehicle's main computer through the Master device and subsequently distributed to the Slave device were reproduced without noticeable artifacts, distortions, or interruptions. Subjective listening tests conducted across various in-cabin conditions, such as different vehicle speeds and background noise levels, confirmed that the audio quality met or exceeded expectations for automotive applications.

Quantitative measurements showed an average Signal-to-Noise Ratio (SNR) of 95 dB for the downstream audio channels and 93 dB for upstream channels using PDM microphones integrated into the Slave node. These measurements were taken at a sample rate of 48 kHz. Comparatively, traditional analog audio systems often struggle to maintain SNR values above 80 dB, especially in the electrically noisy environments of automobiles, indicating a significant improvement provided by the digital A2B setup. In addition, MOST systems typically achieve an SNR of around 85-90 dB, while AVB systems can reach SNR levels of approximately 90-92 dB under ideal conditions. The A2B protocol, with SNR values of 95 dB downstream and 93 dB upstream, demonstrates a clear advantage over these traditional digital systems in terms of audio fidelity.

7.1.2 Stable Synchronization and Latency Analysis

One of the primary advantages of the A2B protocol is its ability to maintain synchronization across all audio nodes. The synchronization between the SPDIF input from the vehicle's main computer and the DSP's internal clock, facilitated by the Asynchronous Sample Rate Converter (ASRC) on the ADAU1452, ensured that all audio signals remained aligned, even when asynchronous sources were involved. Tests confirmed that synchronization between different audio inputs was consistently achieved without detectable drifting or phase misalignment.

Latency measurements were conducted to evaluate the system's real-time performance. The average end-to-end latency from audio input at the SPDIF-in of the Master to audio output at the amplifier (via the Slave) was measured at 2ms. In comparison, AVB (Audio Video Bridging) typically experiences latency between 2 to 5ms, depending on the network configuration, and MOST (Media Oriented Systems Transport) can exceed 3ms due to its use of optical transmission and higher processing complexity. The deterministic latency of the A2B system, at less than 2ms, provided a perceptibly faster and more responsive audio experience, especially crucial for applications like voice control and active noise cancellation.

7.1.3 Robust Signal Routing and Flexibility

The A2B system's ability to manage both analog and digital audio routing was confirmed through rigorous testing. Audio streams from an external microphone, connected via the Slave node, were transmitted upstream to the Master with clear and interference-free signal routing. PDM microphones integrated into the Slave board captured high-fidelity audio, which was converted to PCM using the ADAU1452's onboard capabilities and routed to the vehicle's main audio unit.

The routing of audio outputs demonstrated considerable flexibility, allowing simultaneous outputs to the vehicle's amplifier and the Master's SPDIF-out port. This flexibility was validated by configuring SigmaStudio to simultaneously route audio to multiple outputs without any noticeable degradation in audio quality or synchronization issues. This feature highlights the scalability and versatility of the A2B protocol, making it a viable solution for increasingly complex in-car entertainment systems.

7.1.4 Comparative Analysis With Traditional Audio Systems

To contextualize the benefits of the A2B network, comparisons were made against both traditional analog systems and existing digital protocols such as MOST and AVB. Analog systems, while simpler, often suffer from significant signal degradation due to noise and interference inherent in automotive environments. Unlike analog systems, the A2B protocol transmitted audio over unshielded twisted pair cables with digital precision, maintaining fidelity even under high electromagnetic interference, typical of automotive powertrains.

When compared to the MOST protocol, the A2B system provided a notable reduction in wiring complexity and system weight, which is particularly beneficial for electric and hybrid vehicles where weight savings are crucial. The daisy-chain architecture of A2B, using a single UTP cable, also contributed to a cleaner, less cumbersome installation process compared to the optical cables used in MOST systems.

In contrast to AVB, which utilizes Ethernet infrastructure, A2B offered a lower-cost implementation with deterministic latency, more suitable for the real-time audio needs of automotive applications. While AVB excels in bandwidth for multimedia applications, it requires specialized network switches and rigorous clock synchronization, adding to the overall system cost and complexity. A2B's simpler setup with deterministic latency proved more efficient for audio-only applications, making it ideal for in-cabin audio management.

7.1.5 Scalability and Network Reliability

The system's scalability was assessed by simulating additional slave nodes in SigmaStudio, confirming that the A2B protocol could effectively handle up to 10 slave nodes without a decline in audio quality or synchronization. The bandwidth efficiency of A2B, supporting up to 32 audio channels, demonstrated its capability to expand beyond the current implementation, making it suitable for future automotive models requiring more extensive audio systems.

Network reliability tests indicated that communication across the A2B bus was robust under various simulated vehicle conditions, including power fluctuations and interference from other electronic subsystems. No packet loss or significant jitter was observed, which further underscored the stability of the implemented solution.

7.2 Discussion

The implementation of the Automotive Audio Bus (A2B) protocol in this project provided a versatile and efficient solution for automotive audio distribution. This section delves into the benefits offered by the SPDIF protocol, the use of PDM microphone arrays, the power efficiency of transceiver nodes, the robustness of the A2B network, and the flexibility in network configuration and control.

7.2.1 Benefits of the SPDIF Protocol

The SPDIF (Sony/Philips Digital Interface Format) protocol played a crucial role in this project by enabling high-quality digital audio transmission between the vehicle's main computer and the Master node of the A2B network. One of the main benefits of the SPDIF protocol is its ability to transmit uncompressed stereo audio, which is critical for maintaining audio fidelity in automotive applications where sound quality is a significant differentiator. Additionally, SPDIF supports bi-directional communication for control data, which streamlines the overall system complexity by reducing the need for separate control lines.

Another key advantage of SPDIF is its immunity to electromagnetic interference (EMI), which is a common challenge in the automotive environment, where numerous electronic subsystems coexist. The coaxial or optical cables typically used with SPDIF offer inherent noise rejection, ensuring that the audio signal maintains its integrity from the source to the destination. The synchronization between the SPDIF input and the Master node's internal clock via the ASRC (Asynchronous Sample Rate Converter) further ensured that the digital audio streams remained aligned, even when multiple asynchronous sources were involved. This attribute makes SPDIF a highly reliable choice for use in the noisy and dynamic environment of automotive cabins.

7.2.2 Advantages of PDM Microphone Arrays

The use of Pulse Density Modulation (PDM) microphone arrays integrated into the Slave node provided distinct advantages for this automotive audio system. PDM microphones convert analog audio signals into digital signals directly, minimizing the risk of interference that can affect analog signal transmission. In an automotive setting, these microphones are ideal due to their compact form factor, low power consumption, and ability to provide high-quality audio input even under challenging conditions.

Having synchronized microphone arrays distributed across the vehicle's cabin offers several benefits. These arrays are particularly useful for in-car communication systems, such as voice commands or hands-free calling, as they enable precise capture of audio from different zones of the vehicle. The synchronization of the microphones allows for improved beamforming and noise reduction capabilities, enhancing the overall in-cabin audio experience. This feature is critical for applications such as active noise cancellation, where precise phase and timing of audio capture are required to effectively attenuate undesirable sounds.

7.2.3 Low Power Consumption and Power Transmission Capabilities

One of the standout features of the A2B transceiver nodes used in this project is their low power consumption, which makes them particularly suitable for automotive applications where energy efficiency is a priority. The transceivers consume minimal power while providing reliable audio and data communication, which helps in reducing the overall energy footprint of the vehicle's electronic systems. This is especially beneficial in electric vehicles, where optimizing energy consumption directly impacts the vehicle's range.

The A2B bus not only supports data transmission but is also capable of delivering power to specific Slave nodes. This feature significantly simplifies the system architecture by allowing certain Slave nodes to be powered directly through the bus, eliminating the need for additional power wiring. The option to use phantom power or hybrid power modes provides flexibility in system design, allowing the A2B network to support both locally powered nodes and those powered via the bus. This capability is particularly advantageous in automotive scenarios where minimizing wiring complexity is essential for both cost and weight considerations.

Furthermore, the A2B bus enables remote control of Slave peripherals through the I2C protocol. This means that registers on the Slave devices can be dynamically adjusted by sending control commands over the A2B network, allowing for on-the-fly changes in system behavior.

For example, different audio zones in the vehicle can be controlled by modifying the settings of the Slave nodes, such as adjusting gain levels or switching audio inputs. This dynamic control capability adds significant flexibility, enabling customizable audio experiences that can be tailored to the preferences of vehicle occupants or specific use cases.

7.2.4 Noise Resistance of the A2B Bus

The choice of an unshielded twisted pair (UTP) cable for the A2B bus provides inherent noise resistance, which is a key requirement in the electrically noisy environment of an automobile. The twisted nature of the cable helps to cancel out electromagnetic interference, which can be generated by various automotive components, such as motors, power converters, and electronic control units. This noise immunity ensures that the audio and control signals transmitted over the A2B bus remain stable and free from corruption, contributing to the reliability of the overall system.

Compared to traditional analog audio wiring, which is highly susceptible to noise, the A2B system's use of UTP significantly enhances the robustness of audio signal transmission. This robustness is further supported by the bus's capability to transmit both audio and control data simultaneously, reducing the likelihood of errors and ensuring that commands are executed accurately across all nodes in the network.

7.2.5 Flexibility in Network Configuration

The flexibility of the A2B network is another notable advantage demonstrated in this project. Using SigmaStudio, the network and each node's configuration can be designed, saved, and reused for similar setups in the future. This feature is particularly useful in automotive applications, where different vehicle models may require similar audio architectures with slight modifications. By saving configuration files, the design and development process can be significantly accelerated, reducing the time required for system deployment and testing.

The ability to easily modify node configurations also allows for scalability in the network design. As automotive audio systems continue to evolve, the need for additional audio sources or outputs can be accommodated by simply updating the configuration files and adding new Slave nodes. This scalability makes the A2B protocol a future-proof solution that can grow alongside the increasing complexity of in-car audio systems.

7.3 Validation and Testing

To validate the system's performance, several test scenarios were executed, each addressing different aspects of the A2B and DSP configuration. In addition, the A2B implementation was tested specifically in a Tesla Model 3 to evaluate its integration into a real automotive environment. The following tests were conducted to verify the effectiveness, reliability, and quality of the A2B network.

7.3.1 Clock Synchronization Testing

The Asynchronous Sample Rate Converter (ASRC) was used to resynchronize the SPDIF input from the vehicle's main computer to the Master’s internal clock. Tests focused on ensuring synchronization between asynchronous audio sources, a critical requirement in complex automotive environments. The testing demonstrated consistent performance, with no instances of signal drift or clock misalignment, even under varying vehicle conditions, such as changes in power levels and engine vibrations. The efficiency of the ASRC in maintaining clock stability was validated, which is crucial for applications requiring precise audio timing, such as active noise cancellation and synchronized audio playback.

In the Tesla Model 3 implementation, the ASRC maintained synchronization seamlessly when the SPDIF input was sourced directly from the vehicle’s infotainment system. Despite fluctuating electrical noise levels inherent in an electric vehicle's powertrain, the ASRC effectively eliminated any desynchronization, showcasing the robustness of the solution under real-world conditions.

7.3.2 Bi-Directional Audio Transmission

To evaluate the bi-directional audio streaming capabilities between Master and Slave nodes, audio streams were inspected using SigmaStudio Schematic Tab. Both downstream audio from the Master to the Slave and upstream audio from the Slave to the Master were thoroughly assessed. Testing scenarios included simulated cabin noise and real driving conditions in the Tesla Model 3 to validate consistency and performance.

The downstream audio from the Master to the vehicle's amplifier, routed through the Slave, remained consistently clear, with no detectable artifacts or interruptions. Similarly, upstream audio originating from the PDM microphones integrated into the Slave node, transmitted back to the Master, showed high clarity across all four channels assigned to the Master’s input. The microphone's 4-channel audio was routed to input channels 1 to 4 on the Master and maintained high clarity, without latency or signal degradation, even during active driving scenarios. This bi-directional audio capability was particularly critical in validating voice command and in-cabin communication systems, where latency and clarity are pivotal.

7.3.3 Peripheral Control and Register Configuration

The A2B network’s capability to control peripheral devices via remote register access was tested through the Register Window Tab in SigmaStudio. Each register, configured according to the proposed schematic, was verified for address accuracy and parameter loading. This included control parameters for both the ADAU1452 DSP on the Master node and the Slave node.

In the Tesla Model 3, the Register Window Tab was used to dynamically adjust parameters, such as microphone gain levels and audio routing paths, during testing. This allowed for the evaluation of the system's flexibility to adapt to different in-cabin conditions by remotely changing the settings of Slave peripherals via the I2C interface. This dynamic control capability ensured successful communication between components and confirmed the stable configuration of the network, which supported reliable operations without manual intervention.

7.3.4 Output Quality Testing

The quality of audio output was evaluated across multiple interfaces, including the SPDIF-out and the 3.5mm analog output on the Master node. Tests involved playback of various audio samples, ranging from music tracks to spoken commands, to verify the clarity and fidelity of the audio signal. In both cases, the output was distortion-free, with high fidelity and a wide dynamic range.

In the context of the Tesla Model 3, output quality testing included the vehicle's native speaker system, which was integrated with the A2B network through the amplifier. Output tests were conducted at various volume levels and during different driving conditions, including highway speeds and stop-and-go traffic. The audio output remained crisp, with no noticeable drop in quality, demonstrating the A2B system's ability to provide consistent audio quality in a challenging automotive environment.

7.3.5 Tesla Model 3 Integration Testing

A specific set of tests was conducted to validate the integration of the A2B network into a Tesla Model 3, focusing on real-world automotive conditions. The testing encompassed the following:

- **Environmental Stress Testing:** The A2B network's performance was validated under various environmental conditions, including temperature fluctuations and vehicle vibrations typical of daily driving. The network demonstrated resilience, with no communication losses or degraded audio quality, confirming its suitability for integration into electric vehicle architectures.
- **Power Consumption Validation:** Power consumption measurements confirmed that the nodes operated efficiently under various driving conditions, drawing minimal power from the vehicle's 12V system. This efficiency is crucial for electric vehicles, where optimizing energy use directly impacts vehicle range.
- **Noise and Interference Resistance:** The A2B system's use of unshielded twisted pair (UTP) cables provided enhanced noise immunity, which was particularly beneficial in the Tesla Model 3, where electric powertrain components generate considerable electromagnetic interference. The twisted pair configuration of the A2B bus effectively minimized noise pickup, ensuring stable audio and control data transmission throughout the vehicle.

Chapter 8

Conclusion and Future Improvements

This thesis presented the design and implementation of a digital signal processing (DSP) system utilizing the A2B (Automotive Audio Bus) network for automotive audio applications. By integrating the ADAU1452 and ADAU1761 processors, a robust framework was developed to effectively manage both digital and analog audio streams, providing a scalable solution for in-vehicle audio distribution. The use of the Asynchronous Sample Rate Converter (ASRC) ensured high-quality synchronization of audio inputs without latency or distortion, confirming the system's suitability for automotive environments. The design leveraged the capabilities of Analog Device components, including evaluation boards such as EVAL-AD2428WD1BZ, to establish an efficient network for audio signal distribution. The network's daisy-chain architecture, facilitated by the A2B protocol, significantly reduced the wiring complexity and weight compared to traditional automotive audio systems, simplifying installation while maintaining high performance. The validation tests demonstrated that the system met the design requirements, ensuring consistent and reliable bi-directional audio streaming. The flexibility of SigmaStudio allowed real-time modifications and improvements, which enhanced the overall performance of the network.

Key outcomes of this thesis include:

- **Efficient Network Architecture:** The proposed system, based on A2B, successfully minimized wiring complexity, which not only reduced production costs but also improved efficiency in installation and maintenance. The reduction in wiring also contributed to a lower vehicle weight, which is a desirable attribute in the automotive industry.
- **Real-Time Audio Quality:** The implementation of ASRC and proper synchronization between the Master and Slave nodes ensured the consistent delivery of high-fidelity audio across all channels. This included robust clock synchronization, minimizing the risks of jitter or audio artifacts, thereby ensuring smooth communication between components.
- **Scalability and Modularity:** The ability to expand the network with additional Slave nodes and peripheral components makes this system highly adaptable to the evolving requirements of modern vehicles. The architecture supports future enhancements, such as adding more zones for differentiated in-cabin audio experiences.
- **Real-World Testing on Tesla Model 3:** Section 6 presented the practical implementation and testing of the A2B network on a Tesla Model 3. This real-world application validated the performance of the system under actual automotive conditions. The integration process highlighted the ease of deployment, and the results confirmed that the A2B network architecture is suitable for current automotive standards in a production vehicle environment.

8.1 Challenges and Solutions

Several challenges were encountered during the design and implementation phases:

- **Synchronization Across Multiple Nodes:** The synchronization of audio streams across multiple nodes was a critical challenge. The use of ASRCs in the ADAU1452 played a vital role in managing different clock domains, allowing smooth synchronization without introducing noticeable latency.
- **Real-Time Testing and Validation:** Ensuring real-time responsiveness during testing was challenging, especially in simulating the environmental conditions of an actual automotive setup. Chapter 6 described the implementation of the A2B network in the Tesla Model 3, where environmental noise, power variations, and integration with existing vehicle systems were critical factors. The use of SigmaStudio for real-time parameter adjustment allowed for effective testing, but future improvements could incorporate more sophisticated testing environments, including further in-car testing setups and varied driving conditions.
- **Compatibility with Different Audio Components:** Making the system compatible with both digital and analog components required precise configuration. This was addressed by leveraging the capabilities of the ADAU1761 codec and configuring the register settings appropriately in SigmaStudio.

8.2 Identified Limitations in the Implementation Phase

During the implementation of the A2B (Automotive Audio Bus) system using the EVAL-AD2428WD1BZ evaluation board, several limitations were encountered. These limitations highlighted certain practical constraints that impacted the functionality and overall performance of the A2B network, especially when applied to multi-channel and complex topologies. The observed limitations provide insights into the inherent challenges when attempting to fully leverage the evaluation board for sophisticated audio applications. Below is a detailed explanation of the specific limitations identified during the implementation process:

- **Bandwidth Limitations and Audio Quality Trade-offs:** The available bandwidth of the A2B network constrains the number of channels and the audio quality that can be transmitted over a single bus. The need to share bandwidth among channels could degrade the quality when trying to extract multiple audio streams.
- **Hardware Limitation on Input/Output Ports:** As mentioned earlier, the board is limited to stereo input (2-channel) using either a 3.5mm jack or SPDIF input. A significant limitation here is the lack of flexibility to connect more diverse audio sources directly. This limits the evaluation of A2B's multi-channel capabilities under realistic conditions where many channels would need to be fed into the network simultaneously. Adding multiple dedicated input/output ports, such as multi-channel analog or additional digital audio interfaces, could improve the board’s utility for more comprehensive testing and evaluation of A2B's functionality.

8.3 Future Works

A2B (Audio to Bus) technology is highly applicable across a wide range of domains characterized by multiple audio sources and sinks. The flexibility and scalability of A2B enable it to manage complex audio networks in environments where efficient audio transmission is critical. A2B offers a streamlined approach for audio distribution, minimizing the wiring complexity and latency while maintaining high-quality audio fidelity, which makes it suitable for various applications. Below, some key areas where A2B technology can play a transformative role are explored.

8.3.1 A2B Technology for the Public Transportation Industry

The automotive sector encompasses numerous audio source and sink nodes, ranging from infotainment systems to active noise cancellation and voice assistance systems. Beyond cars, the A2B network also serves as an ideal solution for audio systems in buses and passenger trains, which involve multiple audio sources and sinks. By deploying A2B, previous wiring complexity is significantly reduced, leading to decreased costs, streamlined audio networks, and synchronized audio streaming throughout these vehicles. In the context of passenger trains, a master node could serve as the main control unit, with slave nodes dedicated to each individual wagon. Each slave node could manage multiple audio sources and sinks, such as various types of speakers and microphones, ensuring a robust and scalable audio distribution network. This configuration not only simplifies installation and maintenance but also ensures high-quality, synchronized audio communication throughout the entire train.

8.3.2 A2B Technology for Enhanced Audio Systems Inside Buildings

Analog Devices' A2B (Automotive Audio Bus) technology presents promising applications beyond automotive systems, especially in the domain of home intelligent audio systems and conference room setups. These environments often encompass large spaces and involve multiple audio sources and synchronization requirements, making traditional wiring both cumbersome and inefficient. By utilizing A2B networks, audio systems in these applications can significantly reduce the complexity of wiring while enhancing flexibility and providing support for a higher number of audio channels.

In smart home audio systems, A2B can enable more streamlined, centralized audio control, reducing the physical clutter typically associated with extensive wiring. This can result in a cleaner, more elegant installation that also benefits from the high-quality, synchronized audio delivery that A2B networks support. The ability to manage multiple audio sources seamlessly across different zones of a home makes A2B an ideal choice for intelligent, connected living spaces.

Similarly, in conference rooms, where high-fidelity microphones, speakers, and other audio sources must work in tandem, A2B provides a highly synchronized audio solution. Conference environments often require precise coordination among multiple microphones to ensure every participant is heard clearly, without delays or loss of quality. The reduced wiring requirements and support for low-latency, synchronized audio provided by A2B not only simplify system installation but also ensure a superior audio experience. This is particularly valuable in professional settings where clear communication is paramount.

8.3.3 A2B Technology for Sound Recording

A2B technology can also be employed for sound recording in music bands, whether in a studio setting or during live concerts. Traditional systems rely on transmitting each audio channel, corresponding to individual microphones for each musical instrument, through long and cumbersome cables to a central control unit. This approach has several significant drawbacks, including increased cost, susceptibility to noise interference, and the inherent complexity of managing numerous physical connections. Furthermore, synchronizing the audio signals received from multiple channels can be quite challenging, particularly in a live setting where precise timing is crucial.

In contrast, an A2B network can offer a more efficient solution for capturing multiple audio channels. By utilizing a simplified daisy-chain or star topology, A2B significantly reduces the need for extensive wiring, thereby decreasing both costs and the noise susceptibility commonly associated with long cable runs. Additionally, A2B ensures that all audio channels are synchronized as they are transmitted over a unified digital network, greatly simplifying the management of timing and reducing the risks of phase misalignment.

Moreover, A2B technology supports real-time data transfer, which allows for immediate modifications and adjustments to specific audio channels. This capability makes it possible to apply a wide range of real-time effects and processing, such as equalization, compression, or reverb, enhancing the overall sound quality and enabling precise control over the audio output. These features are particularly beneficial in both studio environments, where high fidelity and precision are required, and in live concert scenarios, where flexibility, reliability, and high audio quality are crucial to delivering an optimal performance.

Future work could focus on expanding the capabilities of A2B to support emerging applications in these domains. Specific areas include developing higher data rate capabilities to support more advanced audio codecs, integrating A2B with wireless nodes to extend its application to hybrid systems, and enhancing its compatibility with non-audio data transmission to facilitate more general-purpose communication networks.

These advancements will further extend the usability of A2B technology, allowing it to play a crucial role in modern communication and audio distribution across an ever-growing number of scenarios.

Chapter 9

Bibliography

[1] M. Cooperation, "Media Oriented Systems Transport (MOST): The Automotive Multimedia Network," MOST Cooperation, 2017.

[2] I. S. Association, "IEEE 802.1 Audio Video Bridging Standards: Enhancing Ethernet for Real-Time Multimedia Transmission," IEEE, 2013.

[3] B. Automotive, "Digital Audio Networking in Vehicles: Challenges and Solutions," Bose Corporation, 2018.

[4] M. Kessler, "Introducing the Automotive Audio Bus (A2B)," in *AES Conference on Automotive Audio*, Burlingame, San Francisco, CA, USA, 2017.

[5] A. Devices, "A2B Audio Bus Technology," [Online]. Available: https://www.analog.com/en/applications/technology/A2B-audio-bus.html. [Accessed November 2024].

[6] A. T. G. C. D. P. M. B. A. F. N. Rocchi, "Transducer Arrays Over A2B Networks in Industrial and Automotive Applications: Clock Propagation Measurements," *IEEE Access,* vol. 9, 2021.

[7] M. B. E. B. A. T. L. C. L. T. D. P. A. F. N. Rocchi, "A Modular, Low Latency, A2B-based Architecture for Distributed Multichannel Full-Digital Audio Systems," in *Immersive and 3D Audio: from Architecture to Automotive (I3DA)*, Bologna, Italy, 2021.

[8] I. Analog Devices, "AD2426/7/8 Automotive Audio Bus (A2B) Transceiver Technical Reference Manual," Analog Devices, Inc., 2023.

[9] I. Analog Devices, "EVAL-AD2428WD1BZ A2B Evaluation Board Manual," Analog Devices, Inc., Norwood, MA, USA, 2019.

[10] I. Analog Devices, "ADAU1452 SigmaDSP Digital Audio Processor Data Sheet," Analog Devices, Inc., Norwood, MA, USA, 2014.

[11] I. Analog Devices, "ADAU1761 SigmaDSP Digital Audio Processor Data Sheet," Analog Devices, Inc., Norwood, MA, USA, 2024.

[12] I. Analog Devices, "SigmaStudio®," Analog Devices, 2024. [Online]. Available: https://www.analog.com/en/resources/evaluation-hardware-and-software/software/ss_sigst_02.html#software-overview. [Accessed 15 November 2024].

[13] I. Analog Devices, "SigmaStudio and SigmaDSP Documentation," Analog Devices, 27 December 2022. [Online]. Available: https://wiki.analog.com/resources/tools-software/sigmastudio.

[14] P. M. S. L. Pankaj Kumar Mehto, "Design and Implementation for Interfacing Two Integrated Device Using I2C Bus," *International Journal of Innovative Research in Computer and Communication Engineering (IJIRCCE),* vol. 2, no. 3, March 2013.

[15] Y. Y. Y. Z. Jianwei Zhang, "Improved Design of I2S IP with Sidetone Signal Processing Capability (I2S_SC) for ASICs," in *2022 IEEE 8th International Conference on Computer and Communications (ICCC)*, 2022.

[16] M. S. J. G. Proakis, Digital Communications, 5th Edition ed., McGraw-Hill Education, 2007.

[17] Y. X. D. L. Z. L. Z. C. P. W. Jiang Yang, "A Configurable SPI Interface Based on APB Bus," in *2020 IEEE International Conference on Electronics, Information, and Communication (ICEIC)*, 2020.