

# Why A<sup>2</sup>B Enables New and More Complex Data and Audio Transport Systems

Roland Prager, Staff Field Applications Engineer

## Abstract

New enhancements of ADI's A<sup>2</sup>B® bus are enabling new fields in data and audio transport and distribution. Smart buildings, halls, rooms, or smart homes can benefit from the rich integration of new transceivers. This article highlights how new A<sup>2</sup>B enhancements with longer bus lengths of up to 300 m and higher bus power of up to 50 W can help solve more complex systems. It shows examples of applications in which an A<sup>2</sup>B bus can help to simplify wiring architecture with very low hardware and software effort.

## Introduction

A<sup>2</sup>B is a bidirectional, high bandwidth digital audio bus that is capable of transporting I<sup>2</sup>S/TDM/PDM data with I<sup>2</sup>C/SPI control information, along with clock and power, using a single, 2-wire unshielded twisted pair (UTP) cable over distances up to 30 m between nodes and 300 m over the entire bus length. This bus is well established in automotive applications as well as unified communications applications, and through constant further development, A<sup>2</sup>B can also be used for many commercial and industrial purposes.

Commercial, office, and public buildings require the continuous availability of public address systems (PAs). A PA is an electronic system consisting of microphones, amplifiers, loudspeakers, and related equipment. It increases the loudness of a human voice, musical instrument, or other sources of sound. As an example, such systems can replace classic alarms. The need for clear voice-based instructions in case of an emergency improves listener reaction when compared to traditional sirens, improving cognition and potentially avoiding catastrophe.

Furthermore, music distribution, voice alarm, broadcast, and intercom functionality can become part of that system and therefore increase their complexity. As systems evolve with increased needs for additional broadcast functionality, so does design complexity. A<sup>2</sup>B technology may help meet this demand.

In order to serve a high number of nodes, using Ethernet as a physical layer requires costly microcontrollers at each node. This method is most used with simple audio and control devices. Besides Ethernet, 100 V lines are also used and have the advantage of carrying power on cable; however, they can only transport a single audio signal in one direction. Data communication cannot be added, and the required transformers are very bulky.

This is where A<sup>2</sup>B comes into play. A<sup>2</sup>B enables locally related devices such as loudspeakers, microphones, intercom stations, control panels, and sensors to connect with a simple cabling in the daisy-chain format.

A<sup>2</sup>B is an ideal solution due to its ease of use and implementation. An A<sup>2</sup>B transceiver can be connected to a device and users will get 64 bidirectional audio channels (32 downstream and 32 upstream) that can be paired with an I<sup>2</sup>C/SPI/GPIO communication for peripheral devices located at the endpoint. There is no need to develop or use more complex time slot stacks.

A<sup>2</sup>B is a high speed bidirectional time synchronized bus that carries I<sup>2</sup>S/TDM along with I<sup>2</sup>C, SPI, and GPIO data with a data rate of 50 Mbps. The benefit of this high speed data is that the latency is as low as 50 µs between any two nodes.

Up to 17 nodes (including the main node) can be daisy chained on one bus with an unshielded twisted pair cable (UTP). The max bus length can go up to 300 m, whereas the distance between each node can go up to 30 m. Communication can go from node to node, which means every node can send to any other node 32 audio channels, if 48 kHz and 16 bit are selected. Data size can be established for different needs and data rates can be set between 1.5 kHz and 192 kHz. Lower data widths and data rates will lead to higher channel count availability.

Another achievement is bus power, which can be as high as 50 W throughout the entire bus. This allows nodes to be remotely powered over the bus without the need for extra localized power supplies. Speaker nodes with moderate output power can also be supplied over the A<sup>2</sup>B bus. However, higher power consumption needs will require external power supplies.

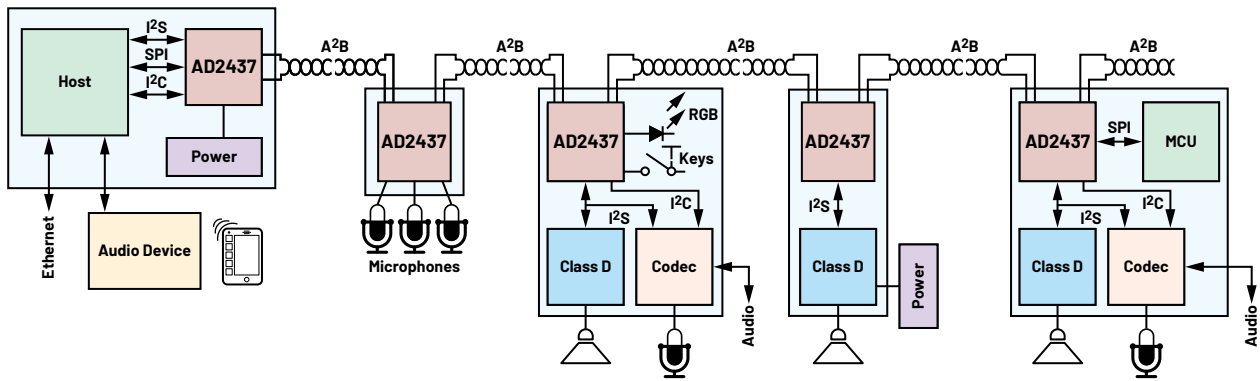


Figure 1. A setup example of an A<sup>2</sup>B network and its attached blocks.

Figure 1 illustrates the flexibility of the A<sup>2</sup>B bus, which eases system design and configuration. Since the transceiver already integrates many blocks and interfaces, it can often operate without the need for a separate microcontroller.

As shown, the simplest node is a microphone array featuring the AD2437 transceiver. Up to four PDM microphones can be attached. Such microphone arrays can be used to localize noise sources to perform a noise cancellation or to find audio's direction of arrival. In this case, it would be possible to extract this audio source from adjacent background noise. Due to the very low latency of the bus, this array does not necessarily need to be on one node. It can easily be distributed in different nodes in various locations within a room. Power will be taken from the bus, as this node will have very low power consumption. There is no need for a separate supply. This makes the solution very small and simple to install. We have one example where we build up this node in a size of 35 mm × 19 mm, including the wire connector and the bus power circuitry.

If more complexity is needed, a separate Class D or any other power amplifier can be attached to the AD2437 over the I<sup>2</sup>S output. Furthermore, an audio codec can be connected as well. As audio usually may not be interrupted at any time,

the I<sup>2</sup>C interface can be taken to set up the power amplifier or codec in parallel. With this configuration, a simple bus-powered intercom terminal can be created. The AD2437 also has GPIOs and some of them with pulse-width modulation (PWM) output. This can be taken as keyport input to interact with the host. The host will get an interrupt if a key is pressed, and communication can be established. PWM output can be used to indicate if communication is active, to drive LEDs, which indicate the status of the connection, and to indicate everything else that is needed. This simple but effective functionality does not require a separate microcontroller, which eases the full system's software effort.

Intercom terminals with higher complexity like a graphical user interface (GUI) can use a microcontroller that can get data over SPI. Note that the maximum SPI bus speed is 10 Mbps.

For pure speaker nodes, which might be in ceilings or other parts of the building that require higher output power, external power supplies will be sufficient. Power to the bus does not need to be inserted at the host node, it can be inserted somewhere within any middle node, to relieve current flow stress within the cables.

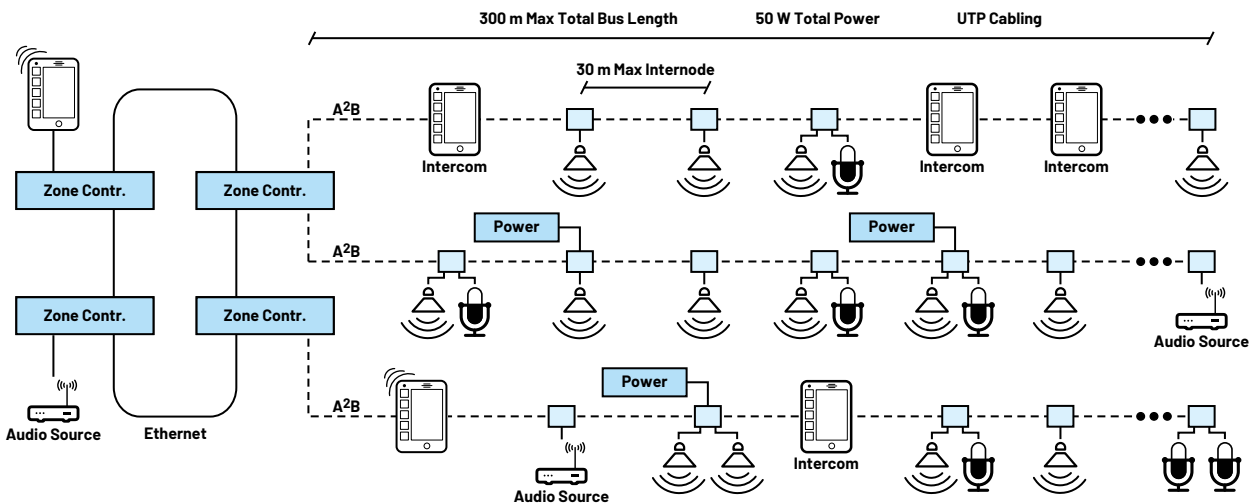


Figure 2. An example of a public address system.

There are many more applications where the feature rich A<sup>2</sup>B can be a differentiator to previously used technology. These applications include those that need communication and multichannel audio delivery.

A nurse calling/notification system features low complexity that consists of an Ethernet configuration. In such case, there will be a room controller connected via Ethernet, but the connection to the patient's terminals will run over A<sup>2</sup>B. This configuration is capable of delivering all necessary audio, data, and power over one UTP cable to up to 16 beds distributed in a room. Each terminal can be easily equipped by using a small microcontroller to provide each patient with a rich selection of different audio channels. A display can indicate channel selection, time, and status on alert. An alert button can be directly connected to one of the GPIO to trigger an interrupt on the room controller. Furthermore, if patients have constrained movements, a selected terminal can transfer microphone signals instantaneously to the room controller. A selected buzzword can be converted and trigger an alarm.

As the installation of this environment can be very dynamic, the system needs to support easy changes to its setup. Adding and removing terminals from the network will be handled from the plug and play stack provided in the host. To keep a daisy-chained configuration on removed terminals, a small adapter can bridge the missing node. If communication breaks, the diagnostic will report failure.

Counter intercom systems can also benefit from the very low latency and full synchronous behavior of the A<sup>2</sup>B bus. Multiple microphones can be located in different positions to use beamforming for a clear channel separation of the person who speaks in front of the counter. All background noise from adjacent persons will be suppressed. A crystal-clear conversation between any acoustical barriers like glass walls can be established. Typically, counter desks, isolation areas in hospitals, or clean room facilities can benefit from such solutions.

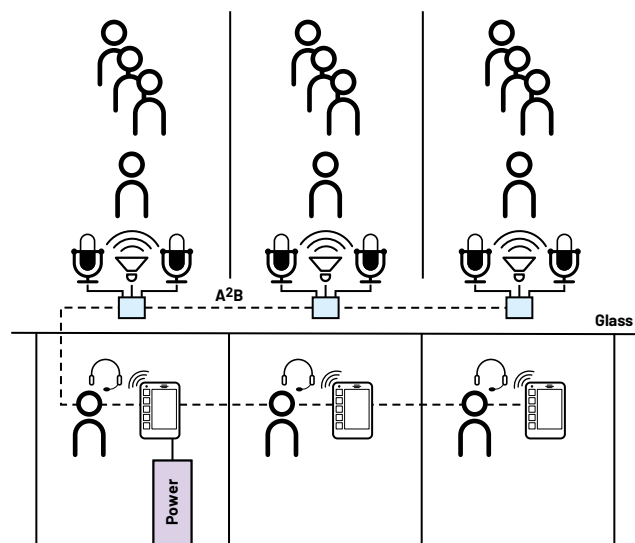


Figure 3. An example of a counter intercom system.

Similarly, prison intercom systems where all microphone data can be transferred to a host system, and many audio channels, like different radio channels, can be broadcast to each cell.

A further good example of use of beamforming is conference systems, where many microphones can be placed in a meeting room. Speech-to-text systems need a clean audio separation between different speakers. Due to the low latency, all microphone channels can simultaneously be transferred to the host controller or digital signal processor (DSP), which calculates the different beam locations.

An alternate approach would be a seat to seat table microphone or headset, which could be routed to an interpreter system. Here all participants can get personalized audio data, automatically translated to their native language. By using lower quality audio bandwidth, a significantly high number of channels can be supported. Hence, all microphone signals are present at the host, and an easy prioritization can be selected, either on the signal with the highest magnitude, or a system-controlled way to open the voice to only selected people. In this case, all terminals can be powered over the bus, without the need for a localized power on the tables.

Home automation systems are getting more and more popular and are well known for control of light, heating, cooling, and roll shutters. In addition to that, audio distribution to many rooms is also needed within these systems. In that way, you would hear your doorbell in every location in the house, even when you listen to your favorite playlist in superior audio quality in your bathroom. The ability to stream audio to any location in a home can be beneficial as well as the ability to support microphones in order to control your house with voice commands. Compared to a wireless connection over WLAN, a cabling connection will provide a better reliable link and offload wireless traffic.

A<sup>2</sup>B is a perfect fit for professional audio systems, home recording studios, and live stage installations enabling easy audio connectivity with existing cable technology like CAT5 or XLR. Learn more by watching our video, [A<sup>2</sup>B: Beyond Automotive—Studio Headphone Mixer Demo](#).

Let's have a deeper look into the A<sup>2</sup>B bus and how it is capable of handling so many audio channels in both directions. A<sup>2</sup>B bus sends superframes over the bus, in a 48 kHz heartbeat manner. The data is transferred 1024× faster, which leads to a 49.152 MHz data stream frequency on the bus. The superframe consists of two parts: one is upstream and the other is downstream, where the start is initiated by a sync control frame and a sync response frame. Inside the downstream/upstream slots, all I<sup>2</sup>S/TDM data, I<sup>2</sup>C data, GPIO, and interrupt information are inserted.

The host generates the clock, and all nodes synchronize to that clock, which makes the system all-time synchronized. A preamble in the sync control frame ensures that all nodes are in sync and can provide that clock to peripherals. That has the advantage that the whole audio chain does not need additional clocks, local oscillators, or asynchronous sample rate converters.

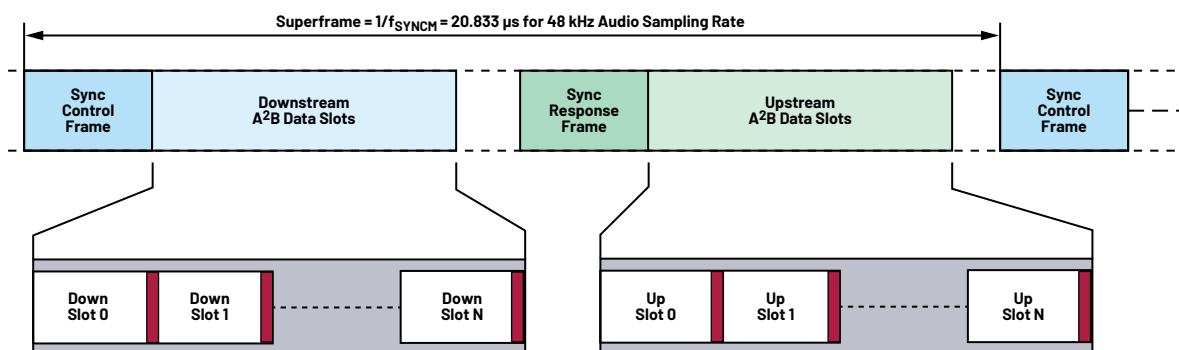


Figure 4. A²B superframe.

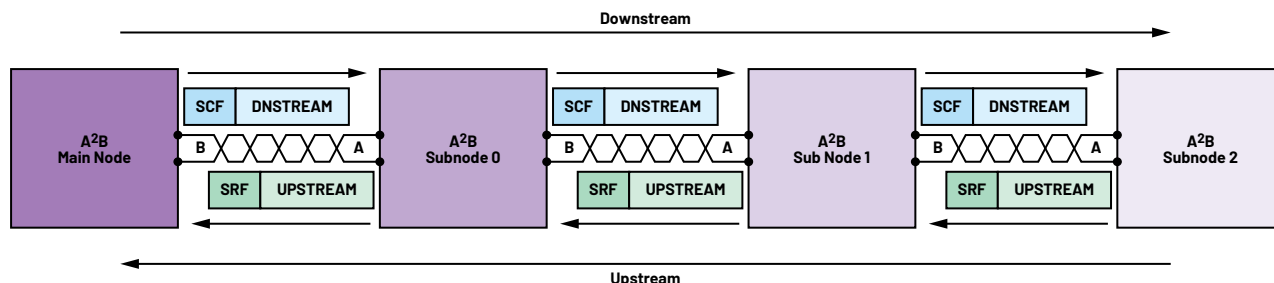


Figure 5. A²B data flow.

On startup, the host processor that is directly connected to the first A²B transceiver chip (like the AD2437) will set it as the main node. The host processor provides a stable 48 kHz signal, and the transceiver starts locking its phase-locked loop (PLL) to this signal. After the main node is set, it starts to bring up the subnodes one by one.

SigmaStudio+ software supports the entire system setup, which includes audio channel and node configuration. This is a graphical programming, diagnostic, and tuning tool that allows the designer to create a graphical user interface for an A²B network, by including peripheral devices such as audio codec, Class D amplifiers, etc. Software stacks are available for Linux along with a plug and play stack, which allows for the addition or removal of nodes from the bus during operation.

A²B is supported by main or subnode evaluation modules with XLR or RJ45 connectors, including power transfer.

## Conclusion

Further enhancements of new A²B transceivers such as longer cable range and more bus power open up a wide range of opportunities for different applications, especially when several audio channels are to be connected to control data via a simple cabling. If the devices on the bus are a mix of sometimes simple and sometimes complex nodes, the bus allows for inexpensive hardware implementations even on less complex ones. Even in nonaudio applications, such as precisely synchronized sensor networks, A²B can bring considerable simplification.

Please find more product information about the new AD2437 A²B transceiver [here](#).

Find A²B collateral and to discover more information on A²B applications by visiting [analog.com/a2b](https://analog.com/a2b).



## About the Author

Roland Prager is a staff field applications engineer at Analog Devices. He has a DI (FH) degree in association of business and engineering, from the University of Wiener Neustadt. He started his career as an electronic designer for intercom systems, and later became a field applications engineer in 2001. He joined ADI in 2018.