Inter-IC Digital Audio Interfaces

By Jerad Lewis

As audio integrated circuit (IC) designs move to finer geometries, it becomes more difficult to design, and less cost-effective to integrate high-performance analog circuits on the same piece of silicon with high-density digital circuits. Audio system architects are pushing analog portions of an audio signal chain further towards the input and output transducers and connecting everything in between digitally. As the analog circuits are pushed to the edges of the signal chain, digital interfaces between ICs in the chain become more prevalent. DSPs have always had digital connections, but now digital interfaces are being included on the transducers and amplifiers that usually have had only analog interfaces. A traditional audio signal chain may have analog signal connections between microphones, preamps, ADCs, DACs, output amplifiers, and speakers, as shown in Figure 1. IC designers are integrating the ADCs, DACs, and modulators in the transducers on opposite ends of the signal chain, which eliminates the need to route any analog audio signals on the PCB, as well as reduces the number of devices in the signal chain. Figure 2 shows an example of a completely-digital audio signal chain.

There are many different standards for transmitting digital audio data from one place to another. Some formats, such as I^2S, TDM, and PDM are typically used for inter-IC communication on the same PC board. Others, such as S/PDIF and Ethernet AVB are primarily used for data connections from one PCB to another through cabling. This article will focus on the differences, advantages, and disadvantages of the inter-IC, rather than inter-board, digital audio formats. Choosing audio components with mismatched digital interfaces needlessly complicates the system design; understanding the pros and cons of different interfaces before selecting parts helps to streamline your component selection and ensure that you have the most efficient implementation of the signal chain.

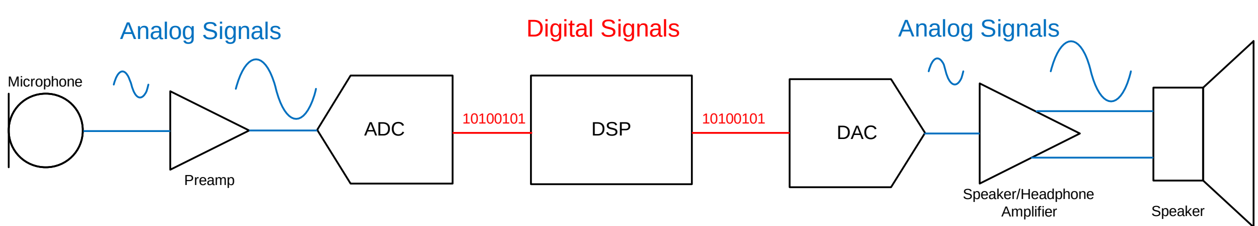

Figure 1. Traditional Audio Signal Chain

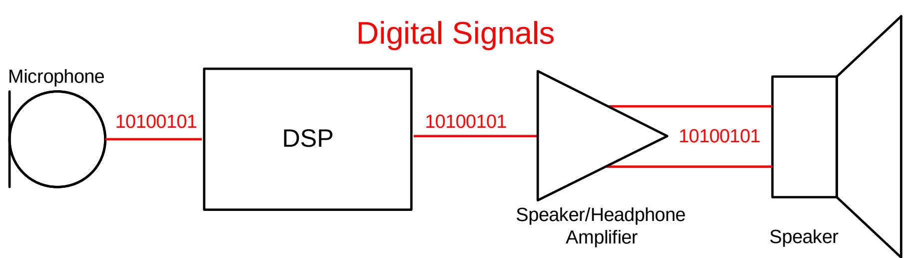

Figure 2. Fully Digital Audio Signal Chain

Inter-IC Sound (more commonly called "I-squared-S" or "I-two-S") is the most common digital audio format used for audio data transfer between ICs. The I^2S standard was introduced by Philips Semiconductors (now NXP) in 1986 and was revised in 1996. The interface was first popularly-used in CD player designs, and now can be found in almost any application where digital audio data is being transferred from one IC to another. Most audio ADCs, DACs, DSPs, sample rate converters, and some microcontrollers include I^2S interfaces.

An I^2S bus uses three signal lines for data transfer – a frame clock, a bit clock, and a data line. The two clocks can be generated by the receiving IC, the transmitting IC, or even a separate clock master IC, depending on the system architecture. An IC with an I^2S port can often be set to be in either master or slave mode. Unless a sample rate converter is being used in the signal chain, a system will usually have a single I^2S master device so that there are no issues with data synchronization.

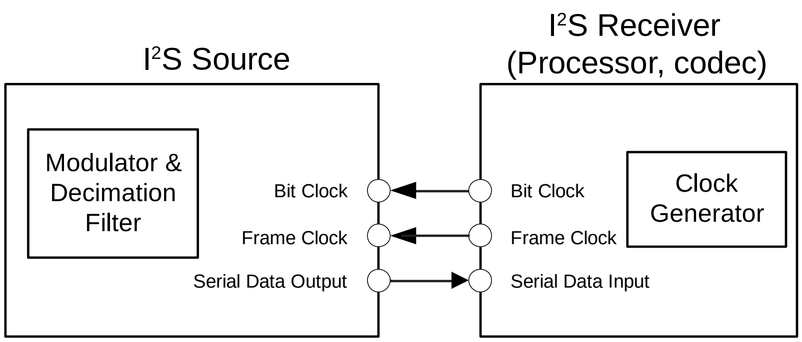

Figure 3. I^2S Connection Diagram, Receiver as Master

The Philips standard for these signals uses the names WS for word select, SCK for the clock, and SD for the data, although IC manufacturers seem to rarely use these names in their IC datasheets. Word select is also commonly called LRCLK, for "left/right clock", and SCK may be called BCLK, for "bit clock" or SCLK for "serial clock". The name of an IC's serial data pin varies most from one IC vendor to another, and even within a single vendor's different products. A quick survey of audio IC datasheet shows that the SD signal may also be called SDATA, SDIN, SDOUT, DACDAT, ADCDAT, or other variations on these, depending on whether the data pin is an input or output.

An I^2S data stream can carry one or two channels of data with a typical bit clock rate between 512 kHz, for an 8 kHz sampling rate, and 12.288 MHz, for a 192 kHz sampling rate. The data word length is often 16, 24, or 32 bits. For word lengths less than 32 bits, the frame length is often still 64 bits and the unused bits are just driven low by the transmitting IC. Although it is rare, some ICs only support I^2S interfaces with a maximum of 32 or 48 bit clocks per stereo audio frame. A system designer has to be careful when using one of these ICs to make sure that the devices on the other end of its connections can also support these bit clock rates.

While I^2S format is the most commonly used, there are other variants of this same three-wire configuration, such as left-justified, right-justified, and PCM modes. These differ from I^2S by the position of the data word in the frame, the polarity of the clocks, or the number of bit clock cycles in each frame.

Some ICs support multiple I^2S data inputs or outputs using a common clock, but this obviously increases the number of pins necessary to transfer the data. Time division multiplexed (TDM) formats are used when more than two channels of data are to be transferred on a single data line. A TDM data stream can carry as many as sixteen channels of data and has a data/clock configuration similar to that of I^2S. Each channel of data uses a slot on the data bus that is $1/N^{th}$ the width of the frame, where N is the number of channels being transferred. For practical purposes, N is usually rounded up to the nearest power-of-two (2, 4, 8, or 16) and any additional channels are left empty. A TDM frame clock is often implemented as a single bit-wide pulse, rather than I^2S's 50% duty cycle clock. Clock rates above 25 MHz are not commonly used for TDM data, since higher frequencies cause board layout issues that PCB designers would rather avoid.

TDM is commonly used for a system with multiple sources feeding one input, or one source driving multiple devices. In the former case, each TDM source shares a common data bus. The source must be configured to drive the bus only during its appropriate channel, and tri-state its driver while the other devices are driving the other channels.

There is no standard for TDM interfaces, such as the Philips standard for I^2S. This means that many ICs have their own slightly-different flavor of a TDM implementation. These differences may include clock polarities, channel configuration, and tri-stating or driving unused channels. Of course, these different ICs will usually work together, but a system designer needs to take care to ensure that outputs of one device will spit out data in the format that the inputs of another are expecting!

PDM data connections are becoming more common in portable audio applications, such as cell phones and tablet computers. This is an advantage in these size-constrained applications because PDM audio signals can be routed around noisy circuitry, such as LCD screens, without having to deal with interference issues analog audio signals might have. With PDM, up to two audio channels can be transmitted with only two signal lines. Figure 4 shows a system diagram with two PDM sources driving a common data line into a receiver. A clock generated by the system master can be used by two slave devices, which use alternate edges of the clock to output their data on a common signal line. The data is modulated at a 64x rate, resulting in a clock that is typically between 1.0 and 3.2 MHz. The bandwidth of the audio signal increases as the clock rate increases, so lower frequency clocks are used in systems where a reduced bandwidth can be traded off for lower power consumption.

A PDM-based architecture differs from I^2S and TDM in that the decimation filter is in the receiving IC, rather than the transmitting IC. The output of the source is the raw high sample rate modulated data, such as the output of a Sigma-Delta modulator, rather than a decimated data, as it is in I^2S. A PDM-based architecture reduces the complexity in the source device, and often makes use of decimation filters that are already present in a codec's ADCs. This allows system designers to use audio codecs that they may already be using, but take advantage of a digital data connection's reduced sensitivity to interference. Also, these decimation filters may be more efficiently implemented in the finer silicon geometries used for fabricating a codec or processor, rather than what is used on the microphone ICs.

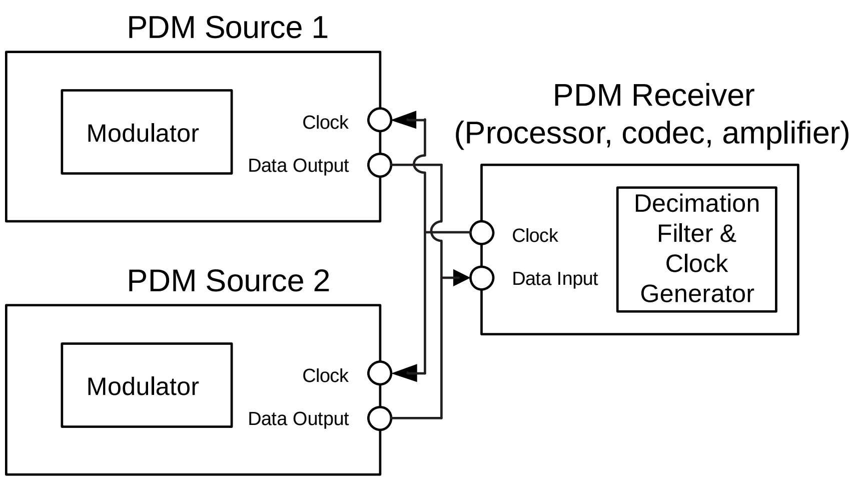

Figure 4. Dual-Source PDM Connection Diagram

Codecs, DSPs, and amplifiers have had I^2S ports for years, but until now a system's input devices, such as microphones, have had either analog or PDM outputs. As the digital interfaces are pushed further towards the ends of the signal chain, new ICs will be needed to support these new system architectures. Microphones that have an integrated I^2S interface, such as the InvenSense ADMP441 MEMS microphone, make it easier for designers to build this component into systems where PDM microphones are not easily used or where analog interfaces are not desired. Only a subset of audio codecs accepts a PDM input and very few audio processors outside of those specifically designed for mobile phones and tablets natively accept this type of data stream. In some designs, an I^2S output microphone could eliminate the need for any analog front-end circuits, since many designs may only have an ADC and PGA in order to support a microphone input to the processor. An example of a system like this is a wireless microphone with a digital transmitter. The wireless transmitter SOC may not have a built-in ADC, so using an I^2S output microphone enables the connections between the transducer and transmitter to be completely digital.

I^2S, TDM, and PDM audio interfaces each have their advantages and applications for which they are best-suited. As more audio ICs are transitioning from analog to digital interfaces, system designers and architects will need to understand which of these interfaces will be most appropriate for their particular design. With a digital signal chain from microphone to DSP to amplifier, analog signals can be pushed completely off of the PCB and exist only in the acoustic domain.

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