LM4665,LMV1012

Digital Microphones - Applications and System Partitioning

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Technology Edge

Digital Microphones - Applications and System Partitioning

By Arie van Rhijn - Amplifier Products

1. Introduction

Almost every audio application will convert the input audio signal into a digital representation to further process the information. Since real world audio signals are small analog signals, at the input of these systems, there are two distinct functions: (pre) amplification and Analog-to-Digital conversion (ADC). National Semiconductor has developed an Integrated Circuit (IC) that, by replacing the JFET in a conventional Electret Condenser Microphone (ECM), will result in a fully digital output of an ECM. This easy to manufacture, four-wire ECM has only power, ground, clock and data connections, completely eliminating the sensitive low-level analog signal present on a conventional ECM.

In this paper, we will first describe and motivate the functional partitioning chosen for this IC, followed by a description of applications which can benefit in performance, size and cost from this microphone technology.

2. Analog to Digital Conversion of Audio Signals

Most common (integrated) systems will pre-amplify the audio signal, and use an over sampled Sigma-Delta converter followed by a digital filter to process the analog audio input. With this type of ADC there are three distinct blocks: The analog pre-amplifier, the Sigma-Delta modulator and the decimation filter. The pre-amplification and Sigma-Delta conversion is usually fixed in gain and sample frequency, while the digital filtering is adaptable to enable software customization and real time adaptation.

2.1. System Partitioning

A number of factors constrain a cost-effective, manufacturable and flexible partitioning of the blocks described above. An optimal partitioning of these blocks can be defined, satisfying these constraints.

2.1.1. ECM manufacturing reliability and cost

Most common ECMs have a two wire interface: GND and Signal/Bias. Two concentric rings will form the connection pins (see Figure 1). This symmetry enables an easy alignment of the canister in its socket.

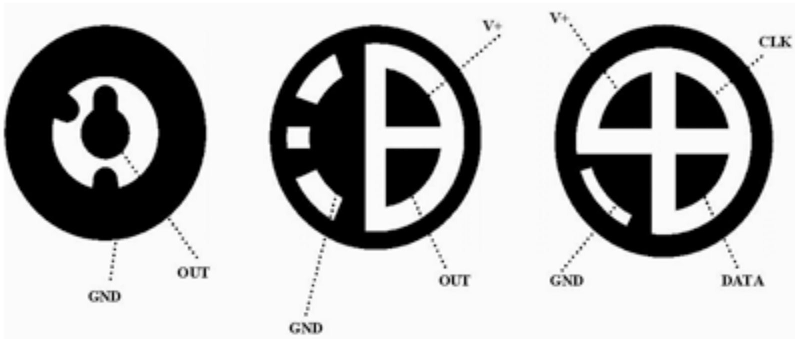

Figure 1. PCB layout diagrams showing conventional two wire, "bulls-eye" land pattern as well as two and three wire land patterns

Having more output pins changes the design of the canister and complicates the placement of the ECM. Therefore, minimizing this number to 3-4 is necessary. Currently is it very feasible to manufacture a four wire ECM. With two connections for power and GND, this leaves two connections for I/O functions.

2.1.2. User Adaptation and Customization

Most applications require some customization, either as calibration or real time adaptation and user customization of the audio signal. This often falls into the pre-amplification block or the (digital) filtering block. The latter can be very useful to shape the audio response such that the resulting digital representation is most suitable for further processing. For instance in Sigma-Delta ADCs, the decimation filter determines the signal-to-noise ratio as well as the audio bandwidth. Different filters can result in hi-fi audio quality or highly compressed voice signals. Also, unwanted signals such as echo can be suppressed through digital filtering. Therefore having user or system control over this filter is very important, making hard-wiring a specific filter inside the ECM impractical.

2.1.3. Overall Complexity and Standardization

A high number of I/O signals from ECM to a Digital Signal Processing (DSP) unit can result in a complex interface between the microphone and the digital processing system. If data transmission in complete words with associated error correction, synchronization, and other signaling required for robust communication is implemented, such a system will be complex, both in pin count as well as (digital) functionality. Finding and agreeing on a common standard will also delay market acceptance of such a microphone. Furthermore, the most cost effective process technology for the Analog signal processing (pre-amplification and A/D conversion) is seldom the best choice for Digital Signal Processing.

For the above described reasons, the following partitioning is chosen for a digital microphone. Inside the ECM a single IC will implement the pre-amplifier and the Delta Sigma modulator. The output of this block is an over-sampled single bit stream and the input a clock signal. The modulator is free-running: conversion will start as soon as a clock signal is present and stop when the clock is low. Figure 2 shows the block diagram of this system.

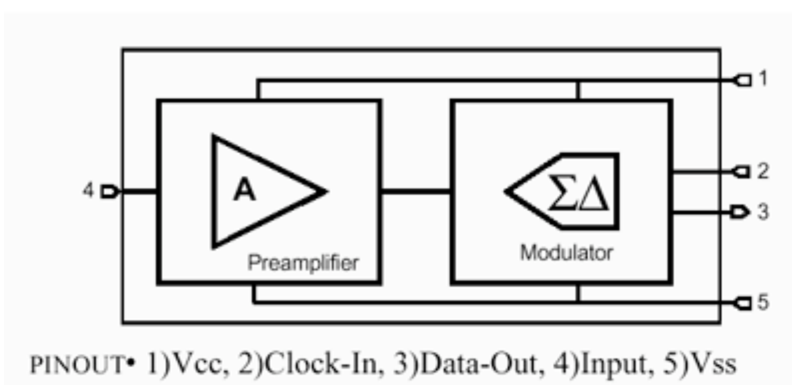

Figure 2. Block Diagram of a Digital Microphone IC inside an ECM

In any Sigma-Delta A/D converter, this over sampled bit stream exists and forms the input to the digital filter. Making this signal available to the output of the ECM, achieves the following benefits. First, the interface of the ECM has only two digital I/Os. These signals are very robust against interference. Second, the interface is simple: a clock signal is provided and a data signal is read. With little digital signal processing and memory needed, a cost effective analog technology can be used. All the necessary digital processing will take place inside the DSP or Microprocessor, often part of the system. This DSP now does not need the analog functions, enabling it to take full advantage of digital process technologies.

Finally, a four wire ECM results, which can be manufactured in high volume at low cost. Figure 3 shows a cross section of the resulting ECM.

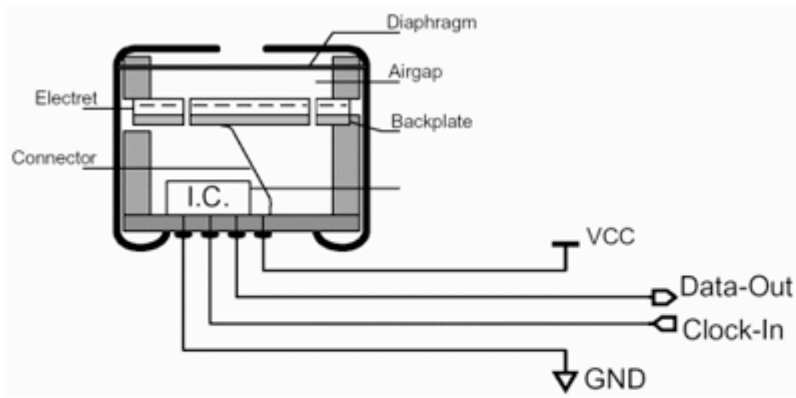

Figure 3. Cross section of a digital ECM

3. Applications

This digital microphone is applicable to many audio applications. Since the design and manufacturing of this ECM is essentially no different from a conventional JFET ECM, high volume, cost sensitive applications can be addressed. Following are two types of applications that can benefit from this technology.

3.1. Digital Mobile Communication Devices

Digital mobile communication devices can benefit from this ECM by the robustness of the interface. Today's audio input suffers from low analog voltage levels, requiring extensive design effort and external filtering components. Having a large, robust digital signal eliminates the need for filtering and eases the placement of the microphone with respect to the baseband ASIC and RF section.

The elimination of the ADC inside the baseband IC opens the way to future all-digital baseband processors which take advantage of high performance digital technologies that are seldom suitable for sensitive analog signal processing. For the audio processing to be completely digital, a digital speaker driver is needed. Today there is already a trend to use higher efficiency drivers (such as class D), which take a digitally coded signal from an all-digital baseband IC. Figure 4 shows a block diagram of such a system, using a Class D audio driver. The filter components are optional, as filter-less class D drivers are available.

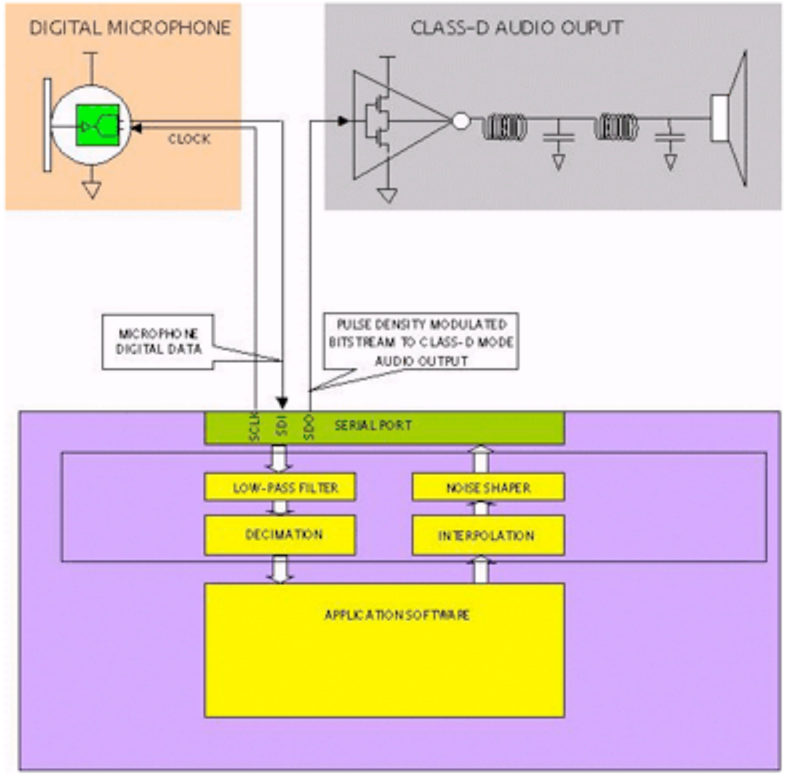

Figure 4. Block diagram of an all-digital audio system

As an illustration of how NSC microphone technology works with today's popular mobile communication DSP systems, an application has been built around one of these DSP units, the TMS320VC5510. The digital microphone communicates to the DSP through two general purpose I/O ports, one line for the Clock signal, and one for the Data signal. Figure 5 shows some test results from this application.

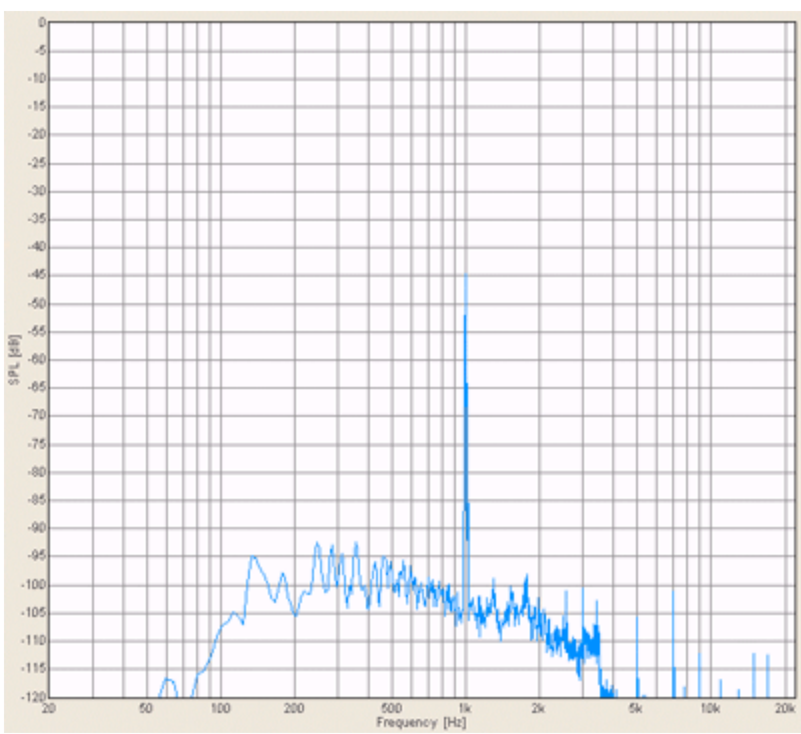

Figure 5. Output spectrum of a digital microphone with a 1kHz input tone

3.2. Digital Consumer Electronics

In low cost consumer electronic devices, a digital microphone can directly interface with a microprocessor, and form a system, using only one digital IC which reads the digital bit stream of the microphone and sends, for instance, a pulse density modulated (PDM) signal directly to the speaker (see Figure 4). Depending on the quality of audio signal desired, this can be done without any filter components. Microprocessors are available that can take a single bit stream as an input and send a PDM signal out. This results in systems that are very small using very few components.

4. Conclusion

A single chip solution, replacing the JFET in an ECM will result in a cost effective microphone that can be produced reliably in large quantities. The different blocks in a digital audio system are partitioned such that the ECM has a minimum number of pins, and optimal use can be made of process technology for integrating analog and digital functions. User flexibility is maintained and overall robustness is improved through eliminating any sensitive analog signals outside the microphone.

5. References

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Digital Speaker Drivers - LM4665

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