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Microphone with Improved Dynamic Range

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Abstract:

This publication describes systems and techniques for a microphone with an improved dynamic range. Electronic devices, including smartphones, generally include a microphone to enable users to interact with the electronic device vocally. Microphones in electronic devices, especially portable electronic devices, are often operated in a low-power mode to conserve battery power. In a low-power mode, the microphones are generally operated with reduced bandwidth and may not accurately detect communications outside that bandwidth. The disclosed systems and techniques provide a microphone that uses an adaptive sigma-delta modulator to improve its dynamic range in low-power mode.

Keywords:

Pulse-density modulation (PDM), pulse-code modulation (PCM), microphone, audio input, audio, dynamic range, sigma-delta modulator, delta-sigma modulator, sigma-delta converter, sigma-delta analog/digital (A/D) converter (ADC), gain, automatic gain control, adaptive gain control

Background:

Electronic devices, including smartphones, tablets, laptops, wearable devices, desktop computers, handheld video game consoles, video game controllers, home automation and control systems, and automobile control systems, generally include a microphone to enable users to communicate with others and the electronic device via voice. For example, the microphone can allow a user to make a telephone call or control the electronic device via voice commands. Electronic devices can also use microphones to pair with another electronic device by exchanging data tokens using ultrasonic pulses.

Portable electronic devices with rechargeable batteries as the power source often operate the microphone in a low-power mode to conserve power. The microphone generally has reduced bandwidth in low-power mode. The electronic device operates the microphone at a lower clock frequency in low-power mode, which reduces the bit rate of sigma-delta encoding in the microphone and reduces bandwidth. The reduced bandwidth can also make it difficult for electronic devices to communicate using ultrasonic communications because they occur at frequencies above the low-power bandwidth. Therefore, it is desirable to provide a technological solution to improve the dynamic range and bandwidth in low-power mode.

Description:

This publication describes systems and techniques for an adaptive sigma-delta modulator that can increase the dynamic range of the microphone while keeping the quantization noise low. In particular, the described systems and techniques apply level-adaptive encoding that scales with the input audio amplitude. Engineers or manufacturers can adapt the level-adaptive encoding for pulse-density modulated (PDM) or pulse-code modulated (PCM) systems. This document illustrates an example diagram of the described systems and techniques in Figure 1.

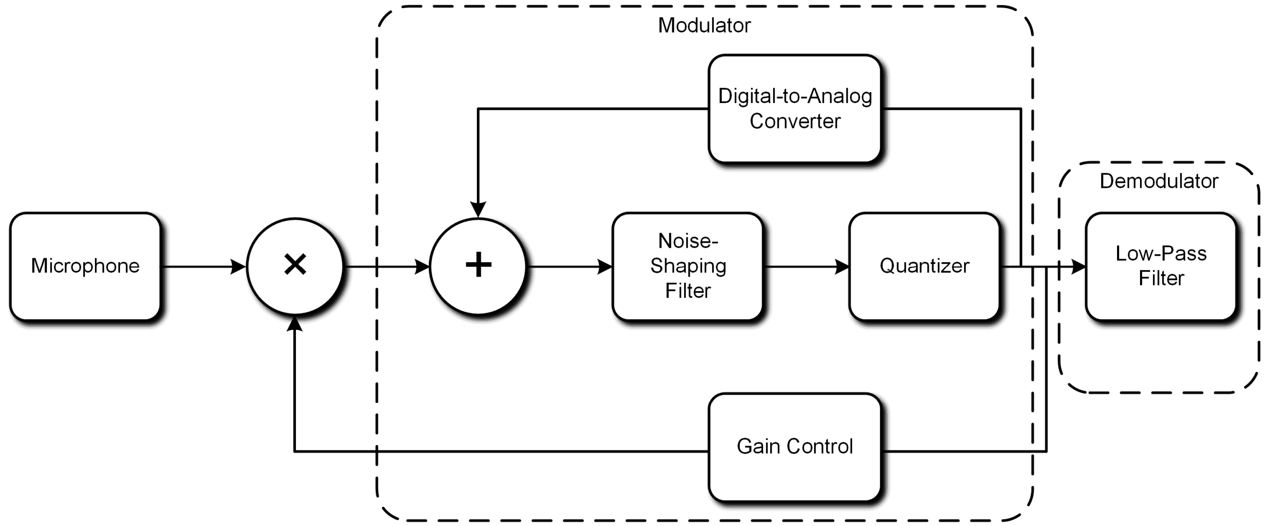

Figure 1

The described systems include a microphone to convert sounds into an analog audio signal. For example, the microphone can be a micro-electromechanical system (MEMS) microphone.

A modulator converts the analog audio signal into a sequence of digital values with a certain number of bits. The output of the modulator is a digital signal representation of the input signal. The modulator, for example, can include a noise-shaping filter and a single-bit quantizer. In general, the modulator shapes the noise power spectrum of the digital signal by moving as much noise as possible outside of the signal bandwidth to decrease the in-band noise power. As a result, the frequency spectrum of the signal of interest is generally located in the low-band portion of the digital signal. In some implementations, the modulator provides the digital signal via a digital-to-analog converter, as a feedback signal to the modulator. The feedback loop allows the modulator to track the input signal. Engineers and manufacturers can extend the modulator structure illustrated in Figure 1 to more efficient structures that include multiloop, multistage, and multibit modulators.

The described systems and techniques can apply level-adaptive encoding in several ways. As illustrated in Figure 1, the modulator can multiply the analog audio signal, via a gain control module, by a gain signal that is a slowly varying function of the modulated digital output signal to scale the processed signal. In other implementations, the modulator can scale the quantizer step-size based on the digital output signal. In contrast to the described backward gain control, the described systems and techniques can also use forward estimation. For example, the modulator can estimate the input signal strength to scale the sampled audio signal or the quantizer step size.

By computing a gain or step size from the digital output sequence, the described systems and techniques can track the time-varying gain at the demodulator. The output bitstream from the modulator is short-time linear with the time-varying gain of the input audio signal of the microphone. In other words, the output bitstream is not linearly related to the microphone input. Suppose the audio-processing system uses pulse-code modulation (PCM) rather than pulse-density modulation (PDM). In that case, the system can use the same gain-tracking techniques by applying dynamic gain control before the encoding, and reconstructing and tracking that gain at a system receiving the PCM sequence.

The demodulator then extracts audio from the digital signal. For example, the demodulator can use a low-pass filter. The described systems and techniques can also use a decimation filter to reduce the demodulated signal to a lower frequency (*e.g.*, the Nyquist frequency). In some implementations, the demodulator can ignore the varying gain and produce a digital audio signal with automatic gain control, or “compression”, applied to it. These systems can also apply proportional level-adaptive encoding in the modulator to reduce the non-linearity of the digital signal. For example, the modulator can multiply the sampled audio signal by a proportion (*e.g.*, one-third, one-half) of the signal level variation. In other implementations, the demodulator can apply an inverse of the gain to approximately re-linearize the digital audio signal. Such systems have the behavior of conventional linear systems but provide a lower noise floor when the signal level is low.

In this way, adaptive sigma-delta modulation in a microphone can increase the dynamic range while keeping ultrasonic communications and other high frequencies within the bandwidth.

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