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MEMS MICROPHONE INTERFACE

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Abstract:- Hearing loss is one of the most common human impairments. It is estimated that by year 2015 more than 700 million people will suffer mild deafness. Most can be helped by hearing aid devices depending on the severity of their hearing loss. This paper describes the implementation and characterization details of a dual channel transmitter front end (TFE) for digital hearing aid (DHA) applications that use novel micro electromechanical- systems (MEMS) audio transducers and ultra-low power-scalable analog-to-digital converters (ADCs), which enable a very-low form factor, energy-efficient implementation for next-generation DHA. The contribution of the design is the implementation of the dual channel MEMS microphones and power-scalable ADC system.

Keywords:- transmitter front end(TFE), digital hearing aids(DHA), ultra low power scalable ADCs, MEMS microphones

I. INTRODUCTION

The first generation of hearing aids usually consisted of analog variable gain amplifiers, electret microphones and speakers that compensated for hearing loss[1]. They dissipated a considerable amount of power and had flat frequency characteristics that made these devices uncomfortable for most patients since hearing loss usually varies across different frequencies. The next generation of devices adopted analog filter banks in which band-pass filters were used in parallel to amplify the acoustic signal to a specific level in each different frequency band. This design, however, resulted in bulky devices that still power consumption. A major breakthrough was achieved through the development of DHAs that exploited the power of digital signal processors (DSPs) that allowed full programmability and customization to a patient's hearing characteristic [2]. H. Neuteboom presented a digital signal processing IC including AD/DA convertors that are needed for one chip hearing instruments. These show a dynamic range range of 77 and 93 dBA. D. G. Gata, presents the analog front-end circuit including a variable gain preamplifier and a low pass filter for digital hearing system. Two-stage operational amplifier with switched-opamp technique is adopted to implement the variable gain preamplifier for power reduction [5]. P. M. Peterson described an adaptive beam forming method that functions to preserve target signals arriving from straight-ahead of a microphone array while minimizing output power from off-axis interference sources[7].

II. DESIGN

A typical Digital hearingAids system consists of a Transmitter Front-End (TFE), DSP, and a Receiver Front-End (RFE). The TFE consists of the microphone, a variable gain amplifier (VGA) and an ADC. The RFE receives the processed digital signal from the DSP and converts it to the analog domain. At the backend, a speaker delivers the acoustic sound to excite the patient's eardrums. One of the major issues with existing DHAs is the rapid degradation of performance in noisy environments in which the TFE becomes saturated due to the ambient acoustic content and background noise. Background noise interferes with the desired conversation thereby impairing intelligibility. While the use of a very high dynamic range TFE can help relieving this problem, it comes at the expense of high power consumption and complexity. Recently developed DHAs also employ microphone arrays combined with adaptive array processing that improve audio quality and perception in real-life environments through noise cancellation mechanisms.

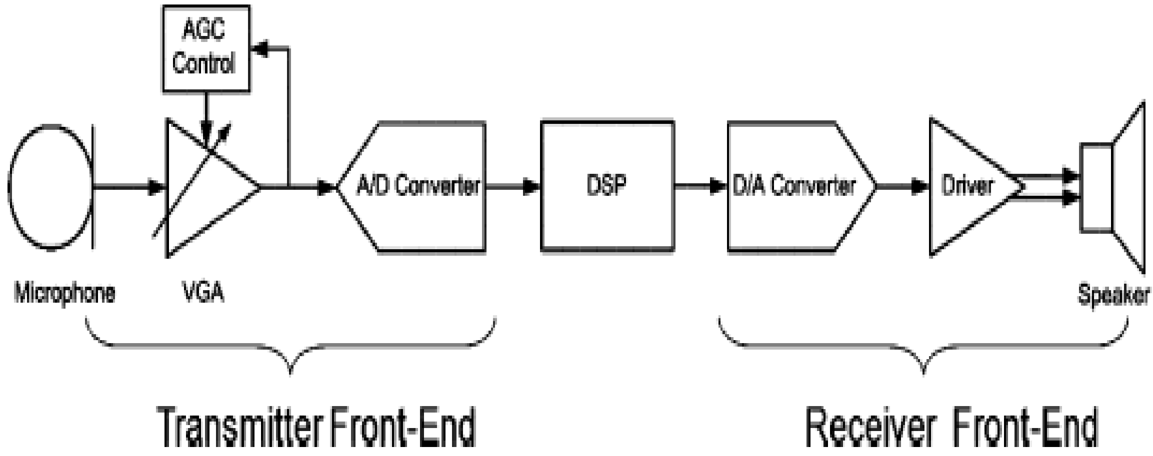

Fig.1

III. MATERIALS REQUIRED

1. MEMS microphone
2. VGA with AGC circuit
3. A/D convertor
4. DSP unit
5. D/A convertor
6. Driver
7. Speaker

IV. MEMS MICROPHONE INTERFACE

Capacitive sensing circuit architectures can be roughly divided into three broad categories: 1) the switched capacitor charge integration (SCI) usually implemented in CMOS, with its main disadvantage being the high noise floor due to noise folding and high kT/C noise; 2) the continuous-time current (CTC) sensing based on charge amplifiers, which are limited by input parasitic capacitance and noise; and 3) the continuous time voltage sensing (CTV), which can achieve superior noise performance over the other two approaches . A CTV approach based on a fully differential difference amplifier (FDDA) is presented in this paper as a reference design, which can be implemented in the same process as the rest of the analog front end The FDDA achieves single ended to differential conversion and 6dB gain. The FDDA consists of dual differential input pairs ,namely, a primary and auxiliary pair. The primary pair is connected to the MEMS microphone, while the auxiliary pair forms a feedback loop. The primary pair and the auxiliary pair implement a virtual short circuit, which provides the high input impedance required for the MEMS microphone and a low output impedance to drive the next stage. ideal FDDA amplifies the differential voltages while suppressing the common mode voltage.

V. FDDA DESIGN

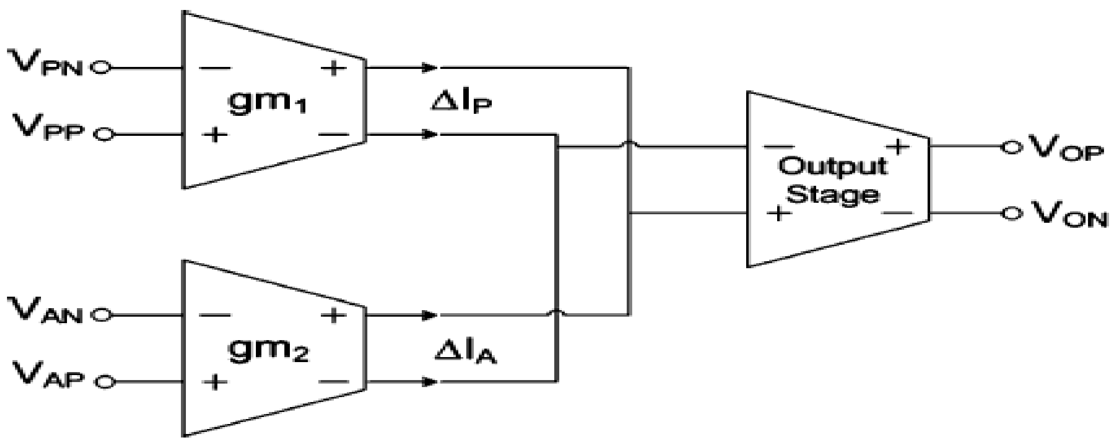

Fig. 2

$$v_{OD} = A_d \cdot [(v_D - |V_{off}|) + \frac{1}{CMMR_P} \cdot (v_{CP} - V_{CP0}) + \frac{1}{CMMR_A} \cdot (v_{CA} - V_{CA0}) + \frac{1}{CMMR_d} \cdot (v_{CD} - V_{CD0})]$$

.... (1)

The back-to-back diodes turn on as the voltage of the high impedance sense node drifts from the bias point thereby essentially clamping the voltage of the sense node to the bias point.

Block diagram of an ideal FDDA is shown in fig2. Where two differential input voltages primary (VPP,VPN), auxiliary(VAN,VAP) are converted into currents through the transconductance stages, and then amplified by an output stage. The ideal FDDA amplifies the differential voltages while suppressing the common mode voltage with respect to the FDDA behaviour is ideally defined by

$$v_{OP} - v_{ON} = A \cdot [(v_{PP} - v_{PN}) - (v_{AP} - v_{AN})]$$

……………….(2)

Since there are two differential pairs, the gain matching of the two parallel transconductance stages(i.e gm1., and gm2) is an important issue and sufficient matching to guarantee correct circuit operation is required. The signal transfer function of the FDDA can be as shown above, where' A_d 'and 'V$_{off}$ 'are the differential gain and input-referred offset, defined similar to the case of conventional opamps. However, the parameters are unique to the FDDA due to the dual input pairs. The $CMMR_p$ and $CMMR_a$ are the common mode rejection ratios of the primary and the auxiliary input pairs.

VI. MEMS CAPACITIVE READOUT LINEARITY

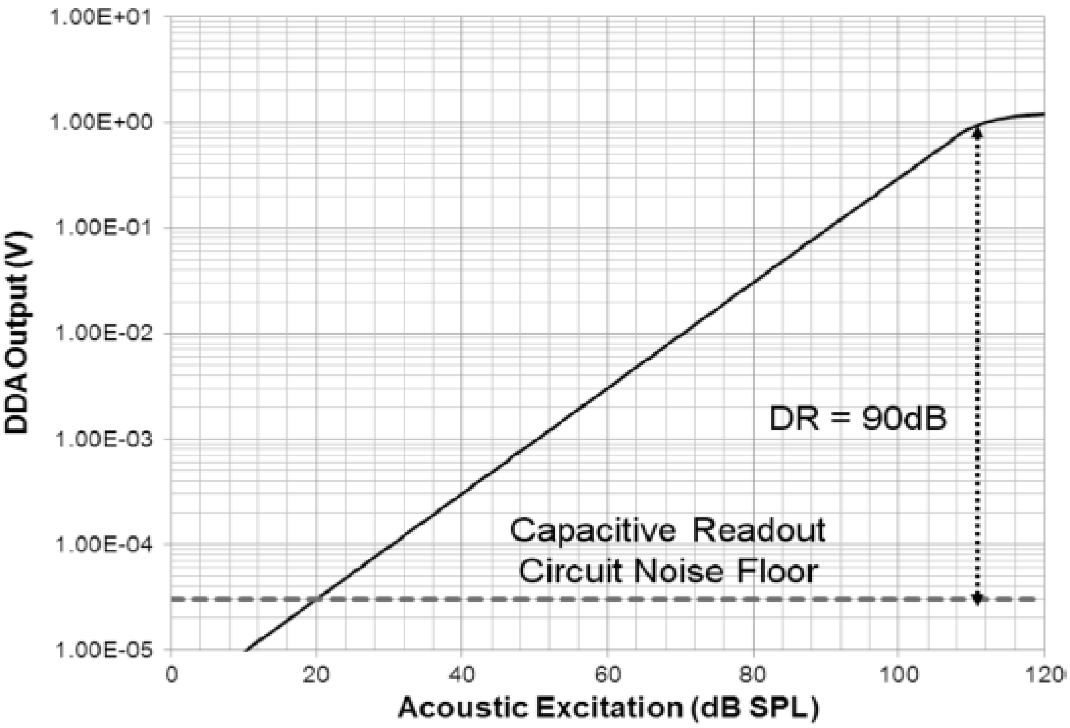

Fig.3

VII. CONCLUSION

The interface circuit is expected to achieve more than 90 db dynamic range as shown in the linearity curve of the interfacing circuit. Hence the MEMS interfacing is an efficient technique to get the required information that is to be captured, amplified and transferred. Using this technique we can built the "DIGITAL HEARING AIDS" through which the hearing loss can be treated with the digital phenomenon with the efficient dynamic range and improved signal characteristics like signal to noise ratio and directionality.

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