AN ENHANCED PRE-AMPLIFIER FOR COCHLEAR IMPLANTS

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Abstract: Analog pre-amplifier for microphone used in cochlear implants, having improved biasing technique has been described. This low-noise circuit relaxes the ADC design requirement, which is designed to replace the input stage of existing Σ/Δ modulators, and has a differential voltage output. Except for the microphone, it needs no external components and uses only two input pins. This input stage achieves a signal to noise ratio of better than 80dB in a frequency band of 100 Hz-10 kHz at a 90dB SPL input. The Total Harmonic Distortion of the circuit is around -60dB at a 90dB SPL input signal with a frequency of 1 kHz. The power consumption of the input stage is 74µA excluding the microphone DC bias current. This circuit is primarily designed for use with microphone cartridges for cochlear implants.

Keywords: *Biasing circuit, Distortion, Electret microphone, Low Power, Sigma-Delta Modulator*

I. INTRODUCTION

A Cochlear Implant (CI) is a device to restore partial hearing to profoundly deaf people. The major components of a CI include an Analog Front-End (AFE), Digital Signal Processor (DSP), electrodes, and power unit. The AFE comprises of microphone, pre-amplifier and sigma-delta (ΣΔ) based Analog-to-Digital Converter (ADC). This work is concerned with the design of low-power, high Signal to Noise Ratio (SNR) pre-amplifier. In Fig. 1, The Front-End part consists of the microphone, a pre-amplifier, and an ADC.

The Back-End part receives the processed digital signal from the processor and converts it to the analog domain. At the backend, an actuator delivers the acoustic sound to excite the patient’s eardrums.

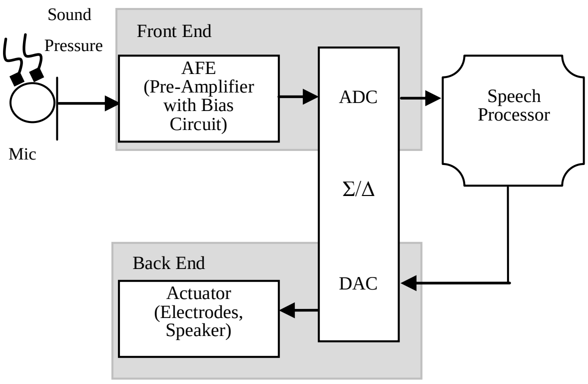

Fig. 1: Basic Block Diagram of Cochlear Implant

Recent trend in CI development aims at full implantation of sound processor, along with the receiver and the electrode array [1], [2]. In such cases, currently the access to the speech processor is over a wireless link [3], [7]. Thus, continuous improvements in the analog blocks become of interest for fully implantable speech processors.

Gain in CIs is required to meet the needs of severely hearing impaired patients. A healthy ear produces hearing sensations to over 120 dB signal pressure level (SPL) [4]. Hearing impairment translates to a reduction of the acoustical Dynamic Range (DR) by 70-80 dB, which may decrease even further as the frequency increases [7].

The microphone generally used in CIs, is Knowles FG-3329A [5] and a preamplifier circuit as described in [6], which provides 20-dB V-V amplification to the microphone signal. In this case, the input to the speech processor, which is running Continuous Interleaved Sampling (CIS) algorithm [8], [9], ranges from 50 µV to 530 mV of amplitude, which, in terms of signal pressure levels translates to 30 dBSPL to 110 dBSPL, i.e. 80 dB of input DR [7].

A pre-amplifier for direct interface with an electret microphone has been designed. The trend in Cochlear implants use an electrets microphone to capture audio or speech in a frequency range of human hearing, which in turn is converted to electrical signal. The DSP takes this electrical signal and performs calculations in order to provide stimulation through the electrodes, which are surgically inserted into the cochlea.

Electret microphones are low cost, small in size, low in weight, and have a very high sensitivity, and very flat frequency response. The low bias voltage requirements (as low as 0.8V) and low DC current drain (down to µAmperes) make them ideal for cochlear implants and other low-power applications. This type of microphone has an integrated single-stage signal amplifier in the form of a single Field Effect Transistor (FET), which requires a DC supply for operation, is configured to amplify the small changes in voltage at its gate due to capacitance

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changes, and the bypass capacitor removes the DC component of the signal due to bias voltage. We do need to provide the FET a bias-current for the microphone to operate correctly. A typical biasing-circuit can be referred to [10]. In this paper alternative circuit will be proposed which avoid the need for external components.

II. DISTORTION AND NOISE IN MICROPHONES

This FET provides additional advantage to substantially reduce the flicker noise (1/f noise) [11]. Although, FET is a good solution for the electrets interface, it provides few disadvantages as well, such as high power consumption due to noise criteria, harmonic distortion due to its non-linear behavior, limitation on SNR, low power supply rejection ratio and the need for external components [11], [12], [13], [14]. Using the ADC as a direct interface for the microphone's sensor is one way of getting rid of the FET preamplifier's problems. This SNR for most electrets is approximately 60-65dB. Thus, with respect to total system performance, the difference between an input circuit with SNR of 80dB as in Eq.1 or a SNR of 85dB as in Eq.2 is not substantial:

$$\mathbf{SNR} = 10\,.\log\left[\left(10^{-65/10}\right] + 10^{-80/10}\right) = -64.8\ \mathbf{dB} \quad (1)$$

$$\mathbf{SNR} = 10\,.\log\left[\left(10^{-65/10}\right] + 10^{-85/10}\right) = -64.9\ \mathbf{dB} \quad (2)$$

This implies that the design constraint of a SNR of 85dB can be relaxed somewhat if necessary to meet the other demands.

III. PERFORMANCE SPECIFICATIONS

The proposed circuit is specified to fulfill and improve upon the specifications found in the existing electrets microphones. The signal bandwidth (BW) is designed to be 20 kHz and this is a specification which could be relaxed as a bandwidth of about 10 kHz would suffice, even for high-end hearing aids [12]. The frequency range of hearing for a normal human is 20 Hz to 20 kHz. Human hearing is most sensitive in the range of 1 kHz to 4 kHz. Normal speech has a DR of 60dB, above 85dB the ear is at risk. Above 140dB the hearing will most certainly get damaged. Therefore, in this work, the sound level tested is around 90 dB. The fact that the SNR does not need to be as large as the DR can be utilized by designing the ADC to have a signal dependent quantization error, i.e. a large quantization error for large inputs and a small quantization error for small inputs.

Certain demands can be made with respect to signal to noise ratio, distortion levels and power consumption. As a starting-point, the goal is set to have a performance at least equal to a Σ-Δ converter with proposed circuit that must fulfill and improve upon the specifications found in the existing electrets microphones. The specifications targeted are specified as following:

- A signal to noise ratio of >80dB in a frequency band of 100Hz-10kHz
- A total harmonic distortion of better than -55dB
- Power consumption (at 1.8V supply voltage) of approximately 70μA (excluding the microphone bias current)
- The high-pass –3dB point of the transfer function at approximately 100Hz

IV. BIASING CIRCUIT

Differential circuit with current-mirroring and microphone-independent biasing of the output stage is shown in Fig 2. This amplifier circuit senses the common-mode output voltage, and adjusts these current sources accordingly. One of the advantages of this circuit is that it is fully differential. The output can be used as a differential current-output simply by removing the output resistor. Another advantage is that in Σ-Δ configuration, it would be possible to directly connect both the integrator capacitors of the first integrator and the Digital to Analog Converter (DAC) to this output.

The circuit described in this work also strictly addresses two distinct problems: stability and a microphone-dependent DC current in the output stage. To overcome the stability-problem, care has to be taken not to control the voltage of two (or more) nodes that are directly influenced by each other. If a DC component that is microphone-dependent in our output-stage is not desirable, only the AC signal from the microphone has to be allowed to flow into the output stage. These demands can be translated into the circuit shown in Fig. 2. I_1, I_2, I_3 and I_4 are equal current sources. T5 (T6) and T7 (T8) are biased so that the voltage across the microphone is approximately 0.9V (a nominal microphone biasing voltage). These cascode-transistors also ensure a low-impedance input for the microphone ($1/g_m$). The output resistors R_1 and R_2 are chosen as two separate resistors instead of one resistor so that the junction of both resistors can be used to measure the common-mode voltage at the output. This voltage is needed to control I_1 and I_2 (or I_3 and I_4) in a control loop that keeps the common-mode output voltage at ½ V_{DD}. For readability, this control loop is not shown in Fig 2.

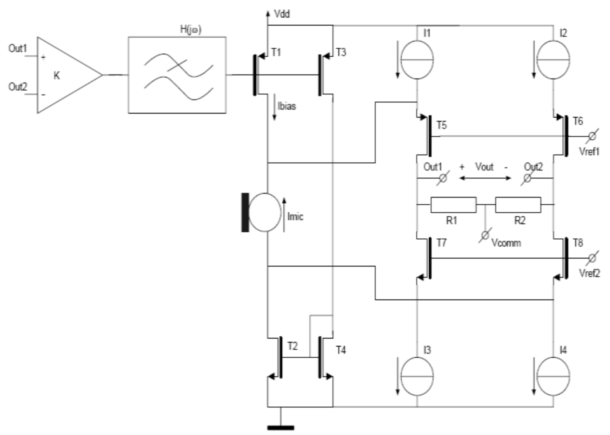

Fig. 2: Differential input stage with current mirror

1. Microphone independent biasing

If T2, T3 and T4 are assumed to be an ideal current mirror, the following equation holds:

$$\frac{Vout}{I_{mic}} = \frac{(R_1 + R_2)}{1 + g_m \cdot K \cdot H(s)(R_1 + R_2)} \tag{1}$$

where g_m is the transconductance of Tl, H(s) is the transfer function of the low-pass filter and K is the gain of the op-amp. Suppose this low-pass filter is a first order RC filter. In that case, (1) becomes

$$\frac{Vout}{I_{mic}} = \frac{(R_1 + R_2)}{1 + \frac{Kg_m(R_1 + R_2)}{1 + j\omega RC}} \tag{2}$$

If $g_m \cdot K \cdot (R_1 + R_2) >> 1$, then for low frequency microphone currents (Q) approaches 0, (2) becomes:

$$\frac{Vout}{I_{mic}} = \frac{(R_1 + R_2)}{1 + g_m \cdot K \cdot (R_1 + R_2)} \approx \frac{1}{g_m \cdot K} \tag{3}$$

Thus, if a small DC output is desired, g_m and K have to be chosen large. For (high) AC frequencies, (2) becomes:

$$\frac{Vout}{I_{mic}} = \frac{(R_1 + R_2)}{1 + \frac{Kg_m \cdot (R_1 + R_2)}{1 + j\omega RC}} \approx R_1 + R_2 \tag{4}$$

As can be seen from this equation, the entire AC microphone current flows into the output resistors R_1 and R_2 (neglecting the input-impedance of the circuit, and the output-impedance of the microphone). If a large AC output signal is required, the value of R_1+R_2 has to be increased. The transfer-curve has the form as shown in Bode Diagram in Fig. 3.

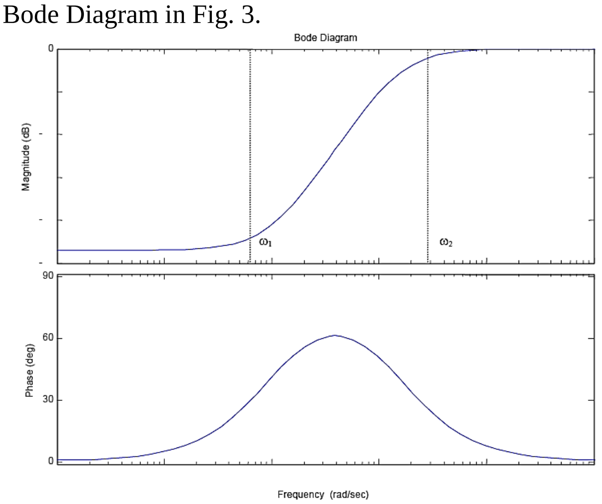

Fig. 3. Generic transfer function of equation (2)

If equation (2) is rewritten to:

$$\frac{Vout}{I_{mic}} = \frac{(R_1 + R_2)}{1 + \frac{Kg_m(R_1 + R_2)}{1 + j\omega RC}} = \frac{(R_1 + R_2) \cdot (1 + j\omega RC)}{1 + K \cdot g_m \cdot (R_1 + R_2) + j\omega RC} \tag{5}$$

the first corner frequency Q) $_1$ is found at:

$$\omega_1 = \frac{1}{RC} \tag{6}$$

and the high-pass -3dB frequency Q)$_2$ at:

$$\omega_2 = \omega_{-3dB} = \frac{1 + K \cdot g_m \cdot (R1 + R2)}{RC} \approx \frac{K \cdot g_m \cdot (R1 + R2)}{RC} \tag{7}$$

From this equation it is observed that the -3dB frequency is dependent on the output resistors R_1 and R_2. These resistors set the AC gain of the circuit. This implies that these resistors cannot be changed dynamically (e.g. as auto-gain control) without extra measures to keep the comer frequency at the same point. Also, if K and g_m are chosen large (for a small DC offset), RC has to be chosen large to obtain a suitable comer frequency (usually around 100Hz). This means a very low-frequency pole has to be created. This could be done by the means of switched-capacitor techniques.

The major advantage of this configuration is the fact that all AC current generated by the microphone flows into the output circuit, and the output voltage swing can be easily enlarged or decreased by changing the values of the output resistors. Actively controlling the voltages of the input nodes can lower the input impedance of the circuit even further (generating a virtual ground).

V. MICROPHONE MEASUREMENTS

To validate the usage of the microphone (lower bias voltages and currents), a number of measurements have been performed. These measurements are described in the following paragraphs. The measurement circuits and the results of the measurements using electrets transducer microphones can be found in Fig. 4 for DC and Fig. 5 for AC.

1. DC measurements:

The DC transfer function of the microphone was measured by connecting a voltage-source through a resistor to the microphone. The voltage across the microphone and the voltage across the resistor are measured. The microphone current can be calculated from the voltage-drop across the resistor. These measurements are performed in a place to make sure AC signals are not present.

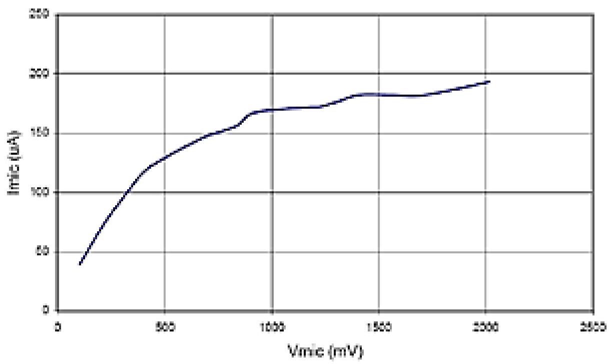

Fig. 4. DC Transfer Function

2. AC measurements:
At various DC bias points, signal transfer functions were measured. A bias point was chosen, and at different frequencies and sound pressure levels (SPL), the transfer-function was measured. Next to the microphone, a dB SPL meter was placed to be able to set the correct SPL. The microphone and dB SPL meter were placed in front of a loudspeaker with internal amplifier, which in turn was fed by a sine-generator. The amplitude of the sine wave was varied, resulting in different SPL's at the microphone membrane.

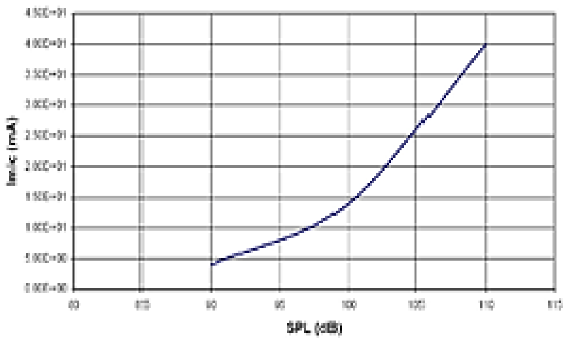

Fig. 5: AC Transfer Function(Sound-pressure vs microphone current at Vbias=1v and Fsound=1khz)

VI. SIMULATION RESULTS

1. Noise performance:
To obtain a satisfactory signal to noise ratio combined with a low current consumption of the circuit, all current sources in the output stage of the circuit have been degenerated. All transistors have been given large dimensions. It was calculated that for a simulated SPL of 90dB, an input voltage of 160mV has to be applied to the input of our microphone model. In Fig. 6, the simulated equivalent input noise voltage is 15µV in a band of 100Hz-10kHz, which translates to a simulated signal to noise ratio of 80.5dB. No dominant noise-sources are left in the circuit after scaling up the sizes of the transistors. In most of the elements thermal noise is dominant, so enlarging them further does not yield much improvement. The only way the noise contribution of these circuit elements can be reduced is by increasing the current flowing through them. The total integrated noise voltage is shown in Figure 6.

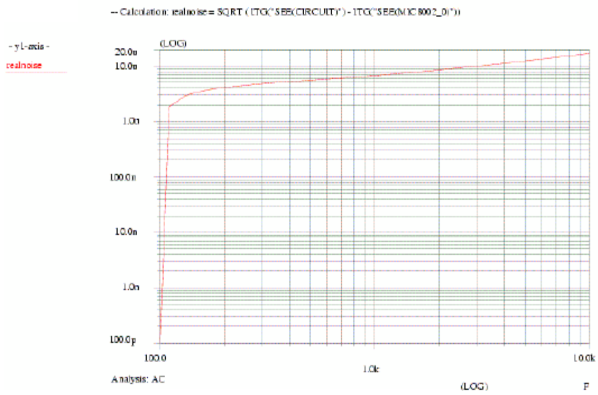

Fig. 6. Integrated Noise Voltage

From the noise analyses the reason for not regulating the microphone voltage becomes apparent. A small noise source on the gates of cascodes T1/T0 and T1/T0 results in an equally large voltage variation across the microphone. Since the microphone is essentially connected as a diode, this results in a relatively large noise current through the microphone and the output resistors. Thus, a very low-noise operational amplifier would be required. With the constraint of low current consumption, this would be very hard to realize. Therefore the use of a resistor network to generate the appropriate voltages is chosen.

2. Distortion:
For the distortion analysis of the circuit, an input signal equivalent to 90dB SPL is injected into the circuit. Taking the input impedance of the circuit and the output impedance of the microphone into account, this results in a signal of 1µA being injected. The injected sine wave has a frequency of 1kHz. The output spectrum of the circuit can be seen in Fig. 7. The total harmonic distortion is almost -60dB.

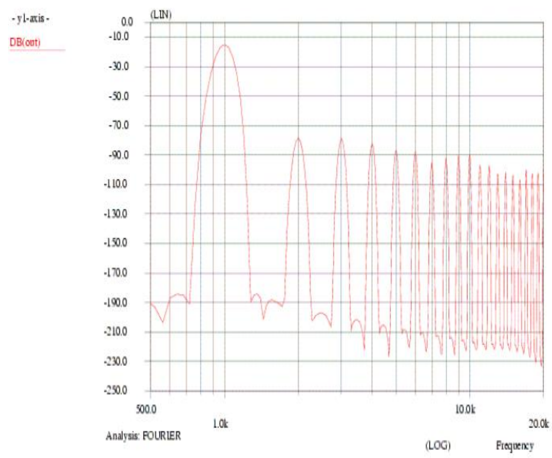

Fig. 7. Output Spectrum – Distortion Analysis

VII. CONCLUSION

A suitable circuit for the targeted applications was designed, and described. In the band of interest, the transfer characteristic of the input stage is flat. The –3dB point of this curve is around 100Hz, which is an appropriate location for speech input. The DC operating point is not sensitive to matching and/or manufacturing errors, so no special measures have to be taken. The signal to noise ratio is within the specifications for Cochlear implants, which is better than 80dB in a frequency band of 100Hz-10kHz at a 90dB SPL input and further lowering of currents and/or reducing the size of transistors would even be possible. The distortion level for a worst-case scenario is around -60dB.

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