A Low-Voltage Low-Power Fully-Integratable Front-End for Hearing Instruments

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Abstract: In this paper, the core of a universally applicable analog integrated circuit for hearing instruments is presented: a microphone preamplifier and an input-controlled automatic gain control with an adjustable knee level. The test chip demonstrates operation down to 1.05 V and a current consumption between 80 and 125 μA. The full-custom chip area in a 2.5μm BiCMOS process (using only vertical NPNs and lateral PNPs) amounts to 0.56 mm^2.

Introduction: Since 1990, our project group 'Low-Voltage Low-Power Electronics' of the Electronics Research Laboratory has been engaged in the research and development of universally applicable analog integrated circuits for hearing instruments. This project is being carried out in close cooperation with an industrial partner. The functions to be incorporated are [1]: a microphone preamplifier, a pickup-coil preamplifier, a highpass filter and a lowpass filter, both with an adjustable cutoff frequency, an input-controlled automatic gain control (AGC-I) and an output-controlled automatic gain control (AGC-O) both with an adjustable knee level, a volume control, a maximum-gain control, a gain-tolerance control, a power amplifier, a -30-dB output (used for driving external power amplifiers in so-called super-power hearing aids) and a microphone supply.
In the following sections, the design and measurement of the microphone preamplifier and the input-controlled automatic gain control (together the core of the total hearing instrument) are described.

A compression/expansion system: From the input-referred noise (< 2 μV) and the maximum input signal (> 28 mV), it follows that the dynamic range of the total hearing instrument must be at least 83 dB. This requires of all the circuits in the signal path that their dynamic range is at least more than this 83 dB. Due to the limited supply voltage, this is a problem especially in circuits that have a voltage as the information-carrying quantity somewhere inside, such as filters [1]. We thus can expect that the design of filters with more than 83 dB dynamic range is the hardest part of the total design.
In [2] a controllable second-order highpass filter has been presented that, with some minor modifications, could be applicable in this design. However, for a 83-dB dynamic range, instead of 50 dB, this would require a total capacitance of 36 nF, which is unacceptable in a fully-integrated realization. For the above reasons, a compression/expansion solution has been chosen. The key idea is that, in order to fulfill the dynamic range requirements of the total circuit, the signal must be compressed before it is passed to the 'noisy' filters. Afterwards, the signal is expanded again to restore the original input-output relation. Thus, the noise is always masked by the signal, leading to a virtually higher dynamic range. As the total circuit already contains a compressor (AGC-I) only one additional expander is needed. The operation of the total compression/expansion system is as follows. The compressor compresses the input signal when it becomes larger than a fixed reference level. This reference level corresponds with the lowest knee level of the AGC-I (60 μV). In the compressor also a control signal is generated that contains information about the magnitude of the input signal. This signal controls the gain of the controlled amplifier in the compressor but can also be used to control the gain of the expander. In the expander the control signal is compared with the knee level (60 μV – 28 mV). If the input signal is smaller than the knee level the expander will expand; the input-output relation becomes linear again. If the input signal is larger than the knee level the expander must no longer be controlled by the control signal; the input-output relation

remains compressed. Note that in this system the knee level thus is set by the expander and not by the compressor.

As discussed in [3], a compressor consists of three parts: a controlled amplifier, a comparator and an integrator. Instead of using a microphone preamplifier with a fixed transfer followed by a controlled amplifier, it is also possible to provide the microphone preamplifier with a controlled output. This reduces the complexity of the total circuit. The remaining circuitry of the compressor, viz. the comparator and the integrator, are called from now on the *envelope processor*.

During the design process, it also appeared that too much offset could be expected to originate from the microphone preamplifier. For the highpass filter this does not pose a problem as it acts as an offset filter itself. However, for the envelope processor, the offset leads to uncertainty in the value of the knee level. Therefore an additional offset filter has been placed between the microphone preamplifier and the envelope processor. The resulting block diagram is depicted in Figure 1.

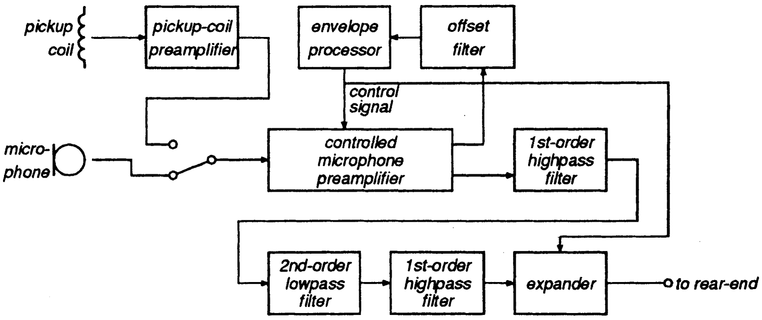

Figure 1: Block diagram of the front-end

As this is a typical low-voltage low-power analog integrated circuit, current has been chosen as the information-carrying quantity wherever possible [4]. The interface between the various circuits is thus performed by currents.

The controlled microphone preamplifier: There are three possibilities of realizing an amplifier with a 50-kΩ input impedance. We can choose a current amplifier with a 50-kΩ resistor in series with its input terminals. Although this is an easy solution, it is not applicable here, because the noise contributed by this resistor would be too large. Another solution might be a transconductance amplifier with a 50-kΩ resistor in parallel with its input terminals. However, this would turn out to be in conflict with a low current consumption. Probably the best solution is realizing the input impedance by means of a two-loop negative feedback amplifier [5,6]. We have chosen for this solution. As the input of the first highpass filter and of the offset filter is a current, we thus need an amplifier with a fixed input resistance and a current output. The basic configuration of this amplifier is depicted in Figure 2.

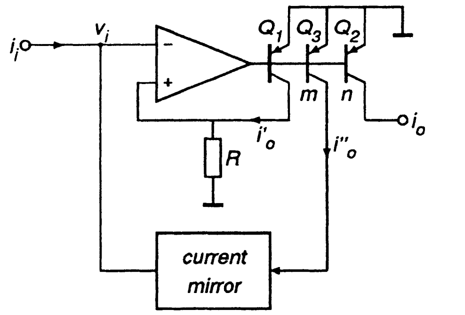

Figure 2: Basic configuration of an amplifier with a fixed input resistance and a current output

The input impedance R_i of this power-to-current converter equals: $R_i = v_i/i_i = R/m$. For the transfer function H_m of the preamplifier we can write: $H_m = i_o/v_i = n/R$. From these equations we see that R_i and H_m can be chosen independently by means of the scaling factors m and n. m can be set by a proper emitter

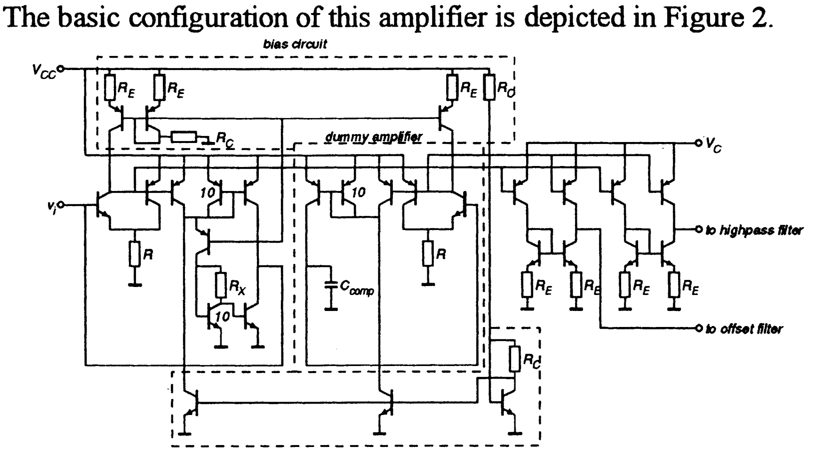

Figure 3: The controllable microphone preamplifier including its biasing scheme

area ratio of Q_1 and Q_3. The scaling factor n can be realized by means of a controllable voltage source V_C in series with the emitter of Q_2: $n = \exp(-V_C/V_T)$, V_T being the thermal voltage kT/q. The complete preamplifier, including its biasing circuitry, is depicted in Figure 3.
As the preamplifier must have two controlled outputs – one to be connected with the first highpass filter, the other with the offset filter – an extra output has been added. To reduce offset at both outputs, the amplifier is made almost symmetrical by means of a dummy amplifier. The collector bias currents are delivered either by the current mirror with two outputs (above) or by the gm-compensated current mirror (below). The emitter resistors R_E are used to improve the noise behavior of the total preamplifier. The resistor R_X reduces the current consumption at high input levels. The capacitor C_{comp} prevents the microphone preamplifier from becoming instable.

The envelope processor: Together, the controlled microphone preamplifier and the envelope processor perform the AGC-I function. In [3], it is shown that we have three possibilities of realizing an automatic gain control with a finite compression ratio. All three contain two controlled amplifiers. However, in the AGC variant with a controlled knee level – the third one – the demands that are made upon the controlled amplifier that controls the knee level can be much fewer. This reduces the circuit complexity and for this reason we have chosen this variant. Its 'current-domain' implementation is depicted in Figure 4.

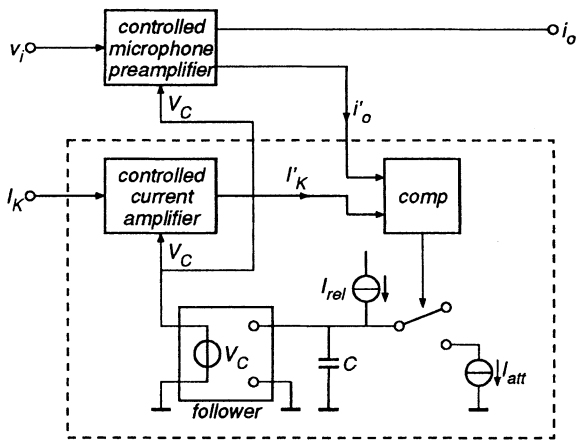

Figure 4: Current-domain implementation of the envelope processor

The compression ratio – the ratio of the variation of the input signal and the variation of the output signal, both in dBs – equals $1-m$. For a desired compression ratio of two, m thus equals -1; the divider becomes an inverter. The transfer function H_{preamp} of the controlled microphone amplifier equals $H_{preamp} = I'_K/I_K = \exp(-V_C/V_T)$. The voltage follower generates a low-impedance version of the voltage across the capacitor C, to avoid interaction. Figure 5 shows the circuit diagram of the envelope processor with ideal bias sources.

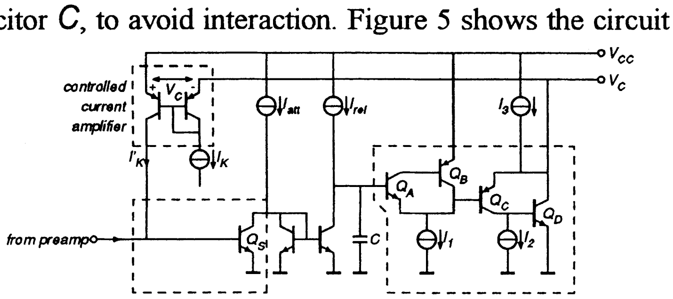

Figure 5: Circuit diagram of the envelope processor with ideal bias sources

The controlled current amplifier has been realized by means of a scaling current mirror. The output current of the controlled microphone preamplifier is subtracted from I'_K. If the remainder is negative, Q_S becomes non-conducting and the capacitor C is discharged by the current I_{att} - I_{rel}. If the remainder is positive, Q_S saturates and C is charged by I_{rel}. The voltage follower is realized by means of transistors Q_A through Q_D. I_1, I_2 and I_3 provide the collector currents of these transistors. The reference current I_K corresponds with the lowest knee level of the AGC-I (60 µV). The actual knee level is set by the expander.

The expander: In a previous section, we discussed the function of the expander, which is to expand the signal as long as the input signal of the microphone preamplifier is beneath the knee level of the AGC-I function; if the transfer function of the expander is proportional to the inverse function of the controlled microphone preamplifier, then the total input-output relation becomes linear again. However, if the input signal is above the knee level the transfer of the expander is linear and the total system functions as a compressor with a compression factor of 2. This can be accomplished by a simple circuit, which is depicted in Figure 6.

Its operation is as follows. In formula: $H_{expander}$ = exp(V_C/V_T), $V_C < V_K$ and $H_{expander}$ = exp(V_K/V_T), $V_C < V_K$. Note that both control voltages are with respect to V_{CC}. If V_C is the same control voltage that is generated in the envelope processor and that is used to control the microphone preamplifier, the input-output relation $H_{front\text{-}end}$ of the total front-end satisfies $H_{front\text{-}end} = H_{preamp}H_{expander} = 1/R$, $V_C < V_K$ and exp((V_K - V_C)/V_T)/R, $V_C > V_K$. It must be noted that the filters need not to be taken into account since they all possess a 0-dB transfer in the passband. The exponential relation between V_K and the knee level means that when V_K varies linearly, the knee level varies in dBs. If V_K is made proportional to the absolute temperature (PTAT), the knee level also becomes independent of the temperature.

Semicustom realization of the front-end: The microphone preamplifier, the envelope processor and the expander presented in the foregoing sections have been integrated in two semicustom chips fabricated at the Delft Institute of Microelectronics and Submicrontechnology (DIMES). Experiment results proved the correct operation of the circuits. The measurement results are:

- supply voltage (35 ^{0}C): 1.05 – 1.9 V
- current consumption (35 ^{0}C): 80 – 125 μA
- temperature range (1.3 V): -40 – +100 ^{0}C
- Input-output transfer: 100 μA/V
- Bandwidth: 70 Hz – 100 kHz (-3 dB)
- Distortion: < 2.1 % when input signal is between 14 and 28 mV and < 1.8 % when input signal is less than 14 mV
- attack time: 3.5 ms
- release time: 81 ms.
- knee level: 60 μV – 35 mV (linearly adjustable in dBs)
- compression ratio above knee level: 2
- Input impedance: 52 kΩ
- Input referred noise (A-weighted): 2.2 μV.
- max. input signal: 35 mV unweighted for frequencies higher than 70 Hz and 50 mV unweighted for frequencies lower than 50 Hz

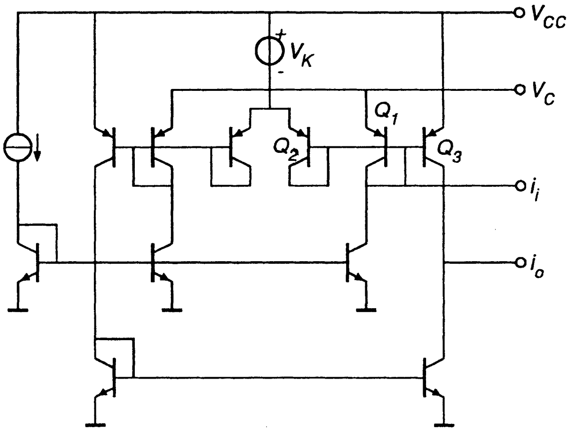

Figure 6: Circuit diagram of the expander

Fullcustom realization of the front-end: All the circuits, of the complete hearing instrument, have been realized on a single chip in a 2.5-μm BiCMOS process that contains vertical NPNs and PNPs as well as lateral PNPs. For confidential reasons no measurement results can be given here. The chip area of the microphone preamplifier, the envelope processor and the expander amounts to 0.56 mm^2. For the total hearing instrument IC, including its bond pads, the chip area is about 5.1 mm^2.

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