A Universal Telephone Audio Circuit with Loudhearing and Handsfree Operation in CMOS Technology

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A first CMOS chip, containing line interface/speech, power extraction, loudhearing, handsfree, dc/dc converter and ringer amplifier in a 28 pin package powered only from the line, is presented. The chip provides an enhanced anti-Larsen circuit to prevent acoustic howling in loudhearing mode and a revolutionary "handsfree mode" voice control system which is virtually independent of any background noise and works in a dynamic half duplex as close to full duplex as the acoustic loop gain allows. This paper describes the architectural features of the system and the design of the main building blocks of the integrated circuit.

1. Introduction

The principle difficulty with handsfree telephone design is feedback loop stability. Signals from loudspeaker travelling through air can be picked up by microphone as transmit signals. These signals are then amplified through the transmit path onto the line, where a proportion of these are fed back to the receive channel depending on the sidetone rejection. When the gain in this loop becomes greater than 0 dB there is oscillation and the telephone starts howling. Presently handsfree telephone concepts use four or five integrated circuits (i.e., dialer, ringer, speech circuit, handsfree circuit and loudspeaker amplifier) with some 100 discrete components. Here we present a solution which uses only two integrated circuits and 40 discrete components (see Fig.1). The described circuit contains all speech, power extraction and comfort phone features as well as ringer and

Fig. 1. Proposed system for telephone audio circuit.
loudspeaker driving amplifier (see Fig.2).

2 Handsfree

A conventional handsfree circuit has a channel control with three states, namely idle, transmit or receive. In the idle mode, when no signal is applied, both the transmit and receive channels are attenuated by approximately 20 dB to keep the total loop gain below 0 dB. When a signal is applied to the microphone, the circuit switches to transmit state i.e. the gain in the transmit channel is increased and the gain in the receive channel is decreased accordingly. And vice versa when an receive signal is applied. This approach requires a high degree of discipline, since the three state channel control gives a very distinct half duplex with a relative high switching time constants to avoid chopper effects. Furthermore, the system is very sensitive to the environment, noise, line conditions and acoustics (echo). Apart from keeping a distinct discipline, the user cannot do anything to minimize the effects of these constraints (thresholds, time constants, noise discrimination, etc. cannot be changed or adapted to the actual conditions by the user).

The new handsfree concept presented here has only two possible states. When no signal is applied either from the line or from the microphone, the circuit is in the only static state, which is transmit channel full open and receive channel attenuated by 30dB. The advantage of using the transmit state as the static (idle) state are that the subscriber at the far end hears an open line (the line is not dead), does not miss the initial word of a sentence when the handsfree user starts talking, and hears the level of the background noise at the handsfree user's end which will actuate her/him to speak up accordingly. When the handsfree user starts talking, the circuit remains in the static state. The signal for controlling the channel attenuation is taken after the sidetone amplifier. With volume at 0 dB (default) the threshold for entering the dynamic state is 40mV. The threshold will vary with volume setting, e.g., with volume at +10 dB the threshold is 20mV and at -10 dB at 80mV. In the dynamic state the channel attenuation is controlled by a voltage controlled amplifier. The attack time is 1ms/6dB and the decay time is 30ms/6dB. A speech compression is activated when a transmit signal with a high amplitude reaches a level corresponding to approximately 500mV on line. The speech compression prevents a large transmit signal from activating the receive channel via the sidetone.

Fig. 2. Simplified block diagram.
3 Loudhearing.

With current loudhearing telephone designs the anti-Larsen circuit decreases the gain of the loudspeaker amplifier when a microphone signal is applied. If no signal is applied either from the line or the microphone, the loudspeaker amplifier is at its full gain, the telephone starts howling (loop gain > 0), the gain will decrease until the loop gain is 0 dB and the circuit will continue howling at a low level (assuming the handset microphone is relatively near the loudspeaker). The enhanced anti-Larsen circuit described here works in the following manner: When no signal is applied either from the line or from the microphone, the gain of the loudspeaker amplifier is attenuated by 30 dB and the transmit path is full open. If a receive single is applied, the gain of the loudspeaker amplifier is increased until the total loop gain reaches just below 0 dB (sensed by the handset microphone). The circuit is, therefore, always acoustically stable, also when no signal is applied. This is implemented using similar circuitry as for “handsfree” mode.

4 Main blocks

The IC is divided into two basic blocks:

A) The telephone part which performs all standard telephone functions, namely transmit, receive, line interface and dc mask as well as power extraction circuitry. The most critical part in the telephone block is the power extraction system, since it also to larger extent defines the system performance limitations. It is quite obvious that there must exist two power supply systems. A lower priority one for the loudhearing amplifier or any additional power

![Fig. 3. Power management block.](image)

Fig. 3. Power management block.

Fig. 4. Simulation result of power management block.

demanding circuitry and a high priority system supply for the vital telephone functions.

Loudhearing supply can be generated from the line current which is normally flushed to ground by the DC mask control (see Fig.1). This means that the line current can be directed either to loudhearing supply if the supply is lower than required or drained to ground if the loud hearing supply is high enough. The current division is performed by the loudhearing supply control block which receives the information from line control amplifier, rectification comparator (if required) and loud hearing supply voltage control (see Fig.3). The voltage control can be set either to the vicinity of line DC point or lower than the line DC point reduced by maximum receive signal peak amplitude on line. Putting this into figures assuming typical PTT specifications would mean having loudhearing supply limit at approximately 4 Volts in the first and 2.5 Volts in the second case. The benefits of lower loudhearing supply is that it does not require rectification which besides lower circuit complexity also relieves the demand on line control amplifier since there are no abrupt transitions. However low loudhearing supply is possible only when using low ohmic loudspeakers and if the process technology allows designs of high power low voltage amplifiers. System supply must have priority over loudhearing supply and must be current limited not to distort the transmit signal in the operation current range. To achieve this three solutions are possible. The first and least demanding is to supply the system through AC defining resistor (i.e. 600 Ohm), if AC impedance is set by external components. This requires no active
elements, but the system supply voltage is limited to the line voltage average reduced for the voltage drop caused by the supply current on AC impedance defining resistor. The solution is acceptable in case of low supply currents and the ability of the technology for low voltage operation. Second possible solution is to introduce a third switching element into the line control loop and set the priorities so that the system supply would be dominant over loudhearing supply while the excessive current would be flushed to ground. This solution requires no current limitation network since both supplies are in the same loop but it has problems reliably setting priorities at very low currents and it would also require additional pin. The third solution is to make the system supply completely independent with its own rectification network and current limitation. This solution guarantees absolute priority and reliable operation of the system at very low currents. The penalty is paid in relative complexity of power supply system and in case all switching elements are integrated (to minimize pin count) also extremely layout demanding. Basic schematic of such a system is shown in Fig. 3 and the current division simulation in Fig. 4.

B) The handsfree part which does the input signal evaluation, decides on appropriate gain attenuation and sets the gains for the microphone and loudhearing amplifiers. The attack and decay time constants of the gain attenuation block should be chosen very carefully in order not to be too fast as to distort the nature of speech and not too slow so that the first syllable of a word is missed. The selection of variable gain system influences significantly the area consumption of the handsfree part of the system. To find the optimal solution regarding the chip area and system performance we analyzed analog, digital and combined variable gain systems. Analog solution using transistors as variable gain elements with linearization is the most optimal in terms of chip area (see Fig. 5). However having relatively low supply voltage which results in poor dynamics of the transistor gate control limits the dynamics below 20 dB if total harmonic distortion is to be kept below 2.5%. Another problem is caused by the non-linearity of the gain change which could result in unstable loop conditions in the dynamic mode of operation. Digital solution having a resistor chain with the number of taps equal to the number of gain steps is the most straightforward solution (see Fig. 6). Such a system is not very sensitive to the switch on-resistance since it does not influence the gain. The only drawback is the relatively big area consumption if the number of steps increases to 50 or even higher with the system demands. The bigger number of switches connected to the amplifier input also increase the crosstalk problem.

Two step digital solution (see Fig. 7) comprises of two separate resistor chains, one for fine steps and the other for coarse steps. It is much more area efficient than the direct digital approach, but it is sensitive to the on resistance of the switch transistor as compared to the unit resistor. This ratio should be at least 1:20. This results in either the use of large transistor switches (which is undesirable from the crosstalk point of view) or high ohmic resistors. If well resistors are used the problem of well resistor to diffusion contact resistance becomes acute since there is big dynamics in resistor ratios. This problem can be overcome but it is extremely layout dependent. Also the voltage dependency of well resistors have to be modeled accurately.

Combined analog-digital solution performs fine gain changes using analog method and coarse changes
5 System Evaluation and Comparison

The basic system concept of traditional "handsfree" circuits, as in [2]-[6], has been described in section 2. This concept has three possible channel states, namely transmit, receive and idle mode, with no intermediate states. This leads to a distinct half duplex operation. Our proposed system works in a dynamic half duplex as close to full duplex as the acoustic loop gain allows. This means that there exist several intermediate states which enable the channel to be shared by transmit and receive simultaneously (see Fig. 10). "Handsfree" circuits, e.g. [2]-[6], use BIPOLAR or BICMOS processes while in this case a standard CMOS process has been used. It should be noted that this is the first chip containing line interface/speech circuit, power extraction, loudhearing, handsfree, dc/dc converter and ringer amplifier.

Fig. 7. Two step digital gain control.

Fig. 8. Combined analog-digital gain control.

(6-10 dB) in the digital way. The system schematic is shown in Fig.8. The system performance can be described as making analog interpolation between two coarse steps using triangular shaped gate control signals. The coarse steps are chosen with the digital selector with selects the two analog transistors performing the interpolation. The gain change simulation and control signal are shown in Fig. 9.

Fig. 9. Simulation of gain attenuation block.

Fig. 10. Channel sharing between transmit and receive, (a) 50/50, (b) 20/80.
6 Conclusions

A universal IC for use in high end comfort phone "handsfree" is designed and fabricated. A revolutionary approach for "handsfree" systems has been proposed. The system components have been drastically reduced due to the high level of integration without sacrificing the flexibility of the solution. This means all the parameters that are influenced by different PTT requirements are changeable at the system level. A very simple controller/dialer interface has been defined which enables an easy telephone design for "handsfree" telephone manufacturers. The only other integrated circuit needed is a controller/dialer. With the proposed solution the cost of electronic of a handsfree telephone will be of the same order as a POT (plain ordinary telephone). This opens up whole new possibilities in mass consumer telephony market.

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References

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