General Description

The PSB 2165 ARCOFI-SP provides the subscriber with an optimized Audio, Ringing, Codec, Filter processor solution for a digital telephone. It fulfills all the necessary requirements for the completion of a low-cost digital telephone.

The ARCOFI-SP performs all coding, decoding and filtering functions according to the CCITT and AT&T standard.

Full featured applications are possible without any external elements. All the necessary hardware and software is implemented. In addition, the ARCOFI-SP offers a speakerphone function as well as a controlled monitoring. These features are fully implemented in the chip. Two independent receive channels offer a more flexible use of the tone generator, the digital gain stages, and the frequency correction.

Two transducer correction filters (one for each direction) can be programmed to correct the analog transducer frequency characteristics.

The ARCOFI-SP provides an universal DTMF, tone and ringing generator. All the generated tone sequences are switchable to each output. This flexible tone generator concept fulfills all the current specifications.

The interfacing to a handset mouth and earpiece is facilitated by a flexible analog front end. A loudspeaker output has also been integrated on the chip as well as a secondary input for a speakerphone microphone. Further auxiliary differential analog input is available. All analog inputs and outputs are gain programmable through software.

Functional Description

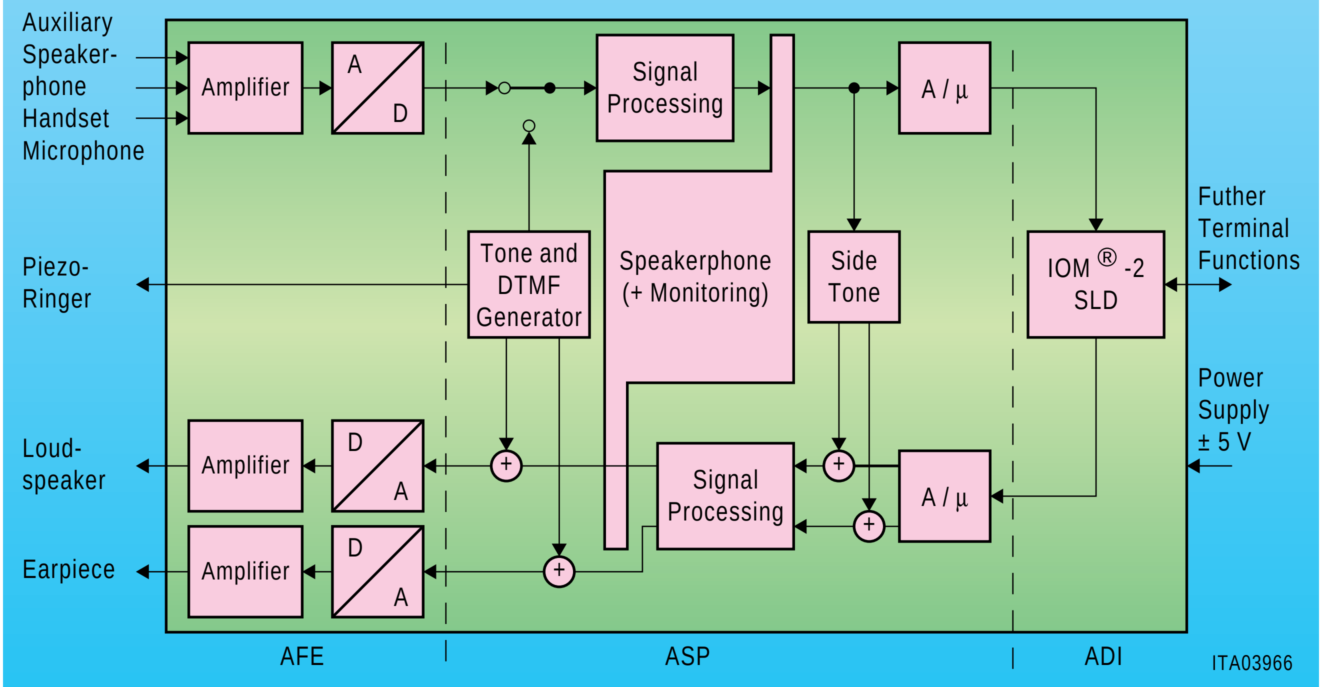
The ARCOFI-SP bridges the gap between the audio world of microphones, earphones, loudspeakers and the PCM digital world by providing a full PCM-CODEC with all the necessary transmit and receive filters as well as a speakerphone and monitoring function.

The ARCOFI-SP can be subdivided in three main blocks:
- The ARCOFI Analog Front End (AFE)
- The ARCOFI Signal Processor(ASP)
- The ARCOFI Digital Interface (ADI).

Type	Package
PSB 2165-N	P-LCC-28-1 (SMD)
PSB 2165-P	P-DIP-28-1

Features

- Applications in digital terminal equipment featuring voice functions
- Digital signal processing performs all CODEC functions
- Fully compatible to the G.714 CCITT specifications
- PCM A-Law/µ-Law and 16-bit linear data
- IOM-2 or SLD serial data interface
- Full digital speakerphone support without any external devices
- Automatic Gain Control (AGC) in transmit direction
- Controlled monitoring (loudhearing) without any external devices
- Dual analog input for the microphone in the handset and single-ended input the speakerphone plus an auxiliary differential analog input
- Two differential outputs for a handset earpiece and a loudspeaker
- 100-mW (sine wave) loudspeaker driver capability
- Independent gain programmable amplifiers for all analog inputs and outputs
- Flexible Peripheral Control Interface (PCI)
- Flexible test and maintenance loopbacks in the analog front end and the digital signal processor
- Flexible DTMF, tone and ringing generator
- Low power consumption (standby < 1 mW)
- Advanced CMOS technology, single 5-V power supply
  Hardware and software support by the Siemens ISDN PC Board SIPB 5000 and ARCOFI-SP coefficient program ARCOS SP Plus SIPO 2165



Signal Flow Diagram