APPLICATION NOTE

ABSTRACT
This application note explains the function and features of the UCB1300 audio codec and also how to select the microphone for UCB1300.

AN807
UCB1300 audio codec

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INTRODUCTION

This application note explains the function and features of the UCB1300 audio codec and also how to select the microphone for the UCB1300.

The UCB1300 audio codec consists of both input and output audio channel and an audio loopback function (see Figure 1).

The input channel provides:

- a 64X oversampling ΣΔ ADC with a digital decimation filter
- a programmable gain microphone preamplifer

The output channel consists of:

- a noise shaper with a 64X oversampling 4-bit DAC
- a BTL speaker driver (up to 16 Ω)
- a digital programmable attenuator
- a mute function.

AUDIO SIGNAL PATH

The audio signal goes through the programmable gain microphone preamplifier, if desired, to increase the signal gain. The programmable gain microphone amplifier consists of 3 stages. One of the stages features a built-in offset cancellation. The function of this built-in offset cancellation stage is to reduce the signal distortion at high gain setting. This signal distortion may be caused by the DC offset voltages of the internal amplifiers or leakage on the board.

The built-in offset cancellation can be deactivated for a signal with low gain setting to improve performance. A general rule is that a signal with a gain setting below 16 (24 dB gain), the offset cancellation will reduce the THD performance and signal bandwidth. To disable the offset cancellation feature, bit 13 on the Mode register (address 0xD) of the Control Register is set to 0 (1 to enable).

After going through the programmable gain circuit, the analog audio signal is converted to digital sample data by the 64 times oversampling sigma delta analog to digital converter. The decimation filter then reduces the data sampling rate by lowpass filtering at the final stage.

The output of the audio path consists of a digital up sample filter. This filter is a 64 times oversampling interpolation with 4-bit digital to analog converter (DAC) circuit followed by a Bridge Tied Load (BTL) speaker driver, capable of driving a 16 Ω speaker. The audio output path features a digital programmable attenuation and a mute function. The audio codec also incorporates a loopback mode, in which codec output path and the input path are connected in series.

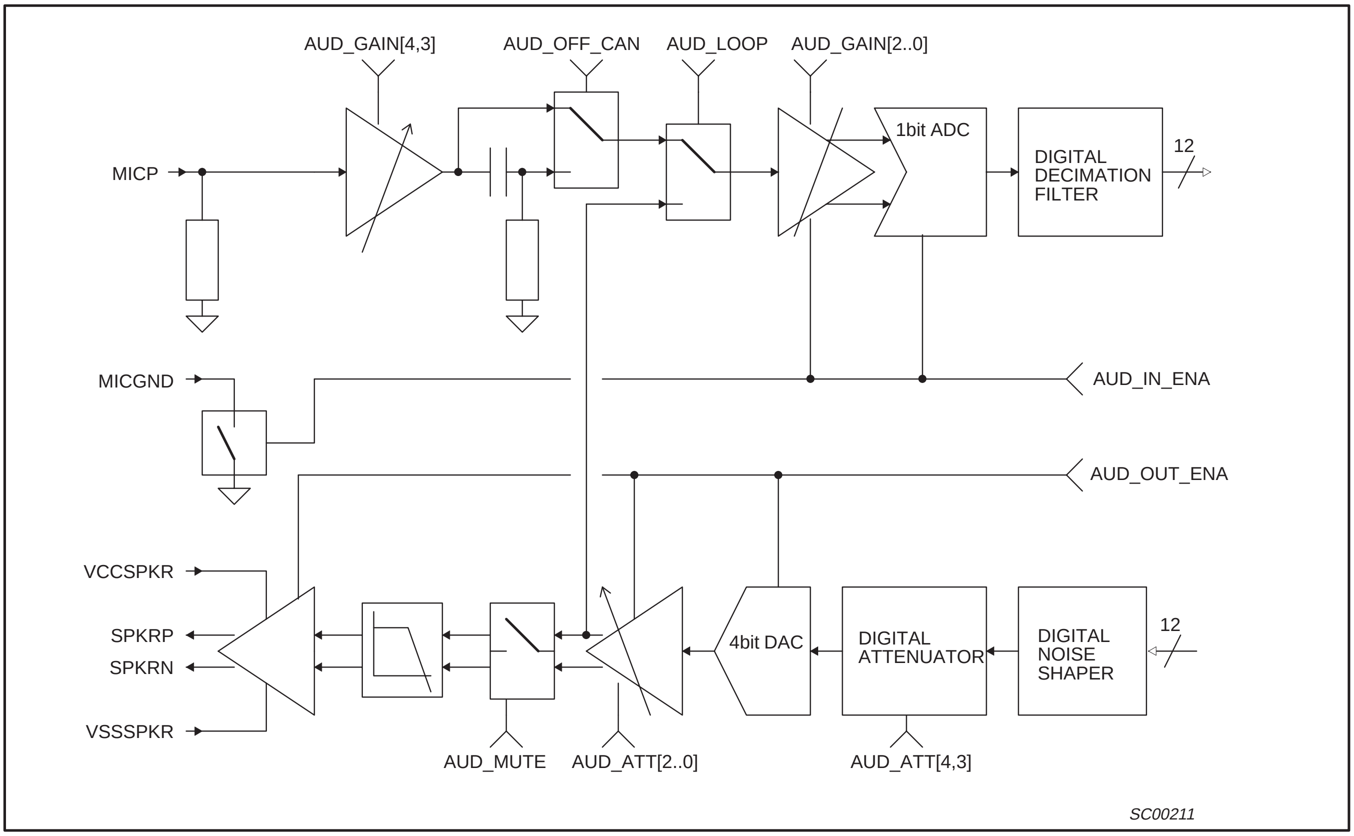

Figure 1. Block diagram of audio codec

AUDIO SAMPLE RATE

The audio sample rate (f_{sa}) is derived from the SIB interface clock pin and is programmable through SIB interface using the AUD_DIV[n]. The audio sample rate is given the the following equation:

$$f_{sa} = \frac{(2 \times f_{SIBCLK})}{(64 \times AUD\_DIV[n])}$$

where

$$(8 \leq AUD\_DIV[n] < 128)$$

For example, a serial clock of 9.216 MHz with a divisor of 12, results in an audio sample rate 24.0 kHz. Both the rising and the falling edges of SIBCLK are used in case AUD_DIV[n] is set to an odd number, which demands a 50% duty cycle of SIBCLK to obtain time equidistant sampling. As for AUD_DIV[n] set to even number, the same rising or falling edges for a cycle of the SIBCLK are used.

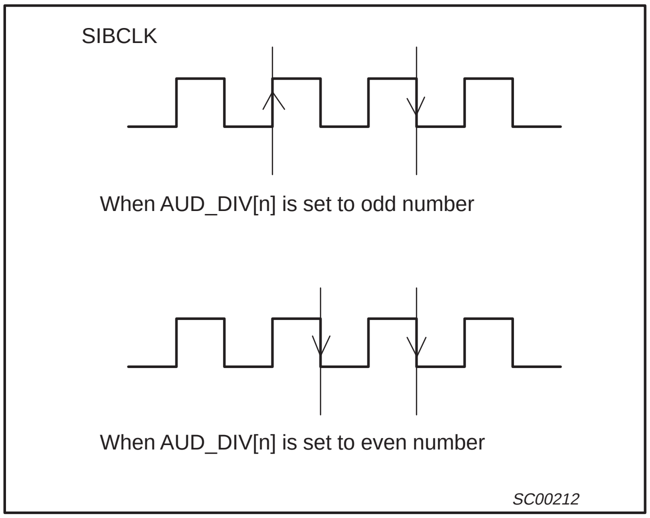

Figure 2. SIBCLK sampling

MICROPHONE CONNECTION

The UCB1300 audio codec input path can accept microphone signals directly (refer to Figure 3). A DC blocking capacitor is needed since the MICP input is biased around 1.4V. The ground side of the microphone is either connected to the analog ground (V_{SS1}) or the MICGND pin. The latter will decrease the current consumption of active microphone, since the MICGND pin will be Hi-Z when the audio codec input path is disabled (i.e., disabling bit 14 of Audio Control Register B, address 0x8).

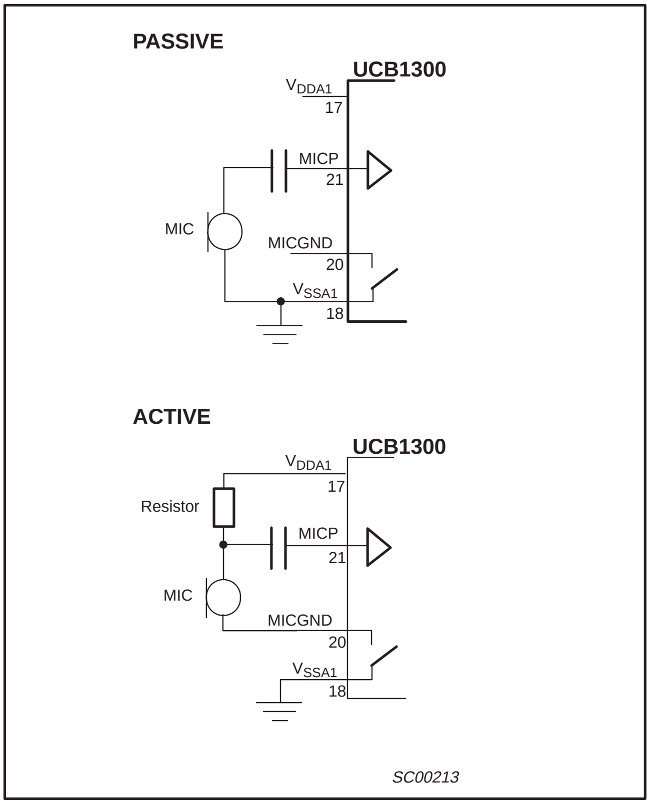

Figure 3. Possible microphone connections

ANTI-ALIASING FILTER ON THE AUDIO INPUT

The microphone input of the UCB1300 needs an analog anti-aliasing filter. This analog anti-aliasing filter prevents passage of false signal (i.e., above one-half the sample clock rate (32 f_{sa})) along with the correct signal from passing into the audio input. The false signals produce alias responses (sound roughness) which cannot be removed later. The anti-aliasing filter will smooth out the sound roughness caused by aliasing.

The analog anti-aliasing filter is not implemented inside the UCB1300. However, it is possible to implement the anti-aliasing filter outside UCB1300 with a cut-off frequency at about 100KHz, since UCB1300 uses delta-sigma architecture.

Most of the alias signal generated by the ADC is filtered by internal digital anti-aliasing filters (i.e., filtered between 0.5 f_{sa} and 32 f_{sa}). Thus, the analog filter can be greatly relaxed.

Only a simple capacitor is necessary to implement this anti aliasing filter outside UCB1300, and the value for this capacitor depends on the internal impedance of the microphone used. For most electret microphones the internal impedance is set at about 2.2k Ω, the circuit in Figure 4 can be used to implement such a filter.

Fcutoff = 1 / (6.28 * 2.2K * 1e-9) = 72.3 kHz

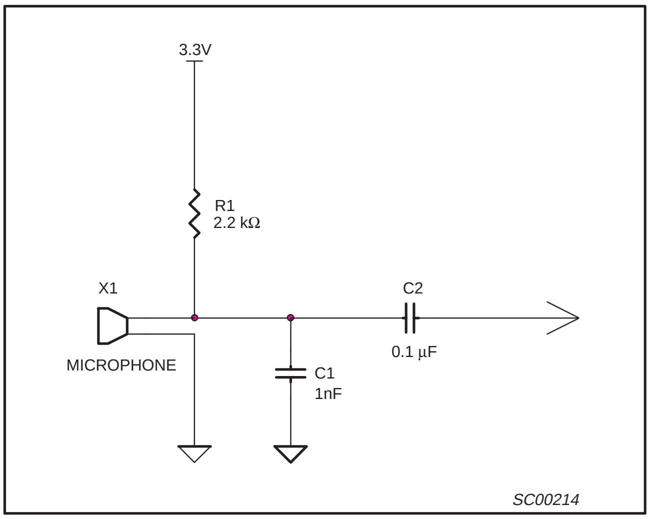

Figure 4. Anti-aliasing filter

AUDIO INPUT GAIN WITH OFFSET CANCELLATION

In order to avoid clipping of the audio ADC when a very high gain is used on the MICP path (above +30 dB), it is recommended to enable the input offset cancellation circuitry by programming the AUD_OFF_CAN bit (bit 13) of the control register 0xD.

The full scale input voltage of the audio input path is programmable in 1.5 dB steps by setting the appropriate number in AUDIO_GAIN[11:7] in the audio control register A. Using very high gains may require the use of the internal offset cancellation circuit programmable in address 0xD in the Mode Control Register to avoid clipping in the ADC.

In the UCB1300, user can set the gain AUD_GAIN[11:7] for the audio input to a maximum of 46.5 dB. Setting the AUD_GAIN[11:7] input to maximum with offset cancellation can affect the performance which is very susceptible to board layout and microphone connections.

CLIP DETECTION CIRCUIT

Whenever the input signal voltage exceeds the maximum input, the clip detection circuit will inform the user by setting the AUD_CLIP_STAT of the audio control register B. The user can clear this status bit by writing a logic ‘0’ to logic ‘1’ transition to the AUD_CLIP_CLR of the audio control register B.

When ACLIP_RIS_INT is set, an interrupt is generated on the IRQOUT pin on the rising edge of the clip detected signal. Users can also read the interrupt status of the audio clip on the ACLIP_INT_STATUS (bit 15) from address 0x4 of the control register. Likewise, when ACLIP_FAL_INT is set, an interrupt is generated on the falling edge of the clip detected signal.

AUDIO OUT ATTENUATION

The output level of the audio signal can be attenuated in 3 dB steps down to –69 dB. The first 8 attenuation steps (0 to 21 dB) are implemented in the analog domain. The digital up sample filter contains a 24 dB and a 48 dB attenuation setting. This arrangement preserves the resolution, thus the ‘audio quality’ of the audio output signal for attenuation setting till 21 dB.

MEASUREMENT OF AUDIO OUTPUT

It is strongly advised to avoid connecting any capacitor directly from any of the outputs to ground since this might cause UCB1300 outputs to oscillate. To avoid oscillation, it is recommended that a resistive load be put in series with the outputs (16 Ω or more for the audio outputs, and 600 Ω or more for the telecom outputs).

In an application where it is required to connect any of the audio outputs to ground with a capacitor for an optional low-pass filter (normally it is not required, since the UCB1300 has a built-in digital low-pass filter), it is advisable that an op-amp to be used to buffer UCB1300 audio output signal first before driving that capacitor. In Figure 5 are two of the op-amp circuits that can be used to buffer or amplify UCB1300 audio out signal.

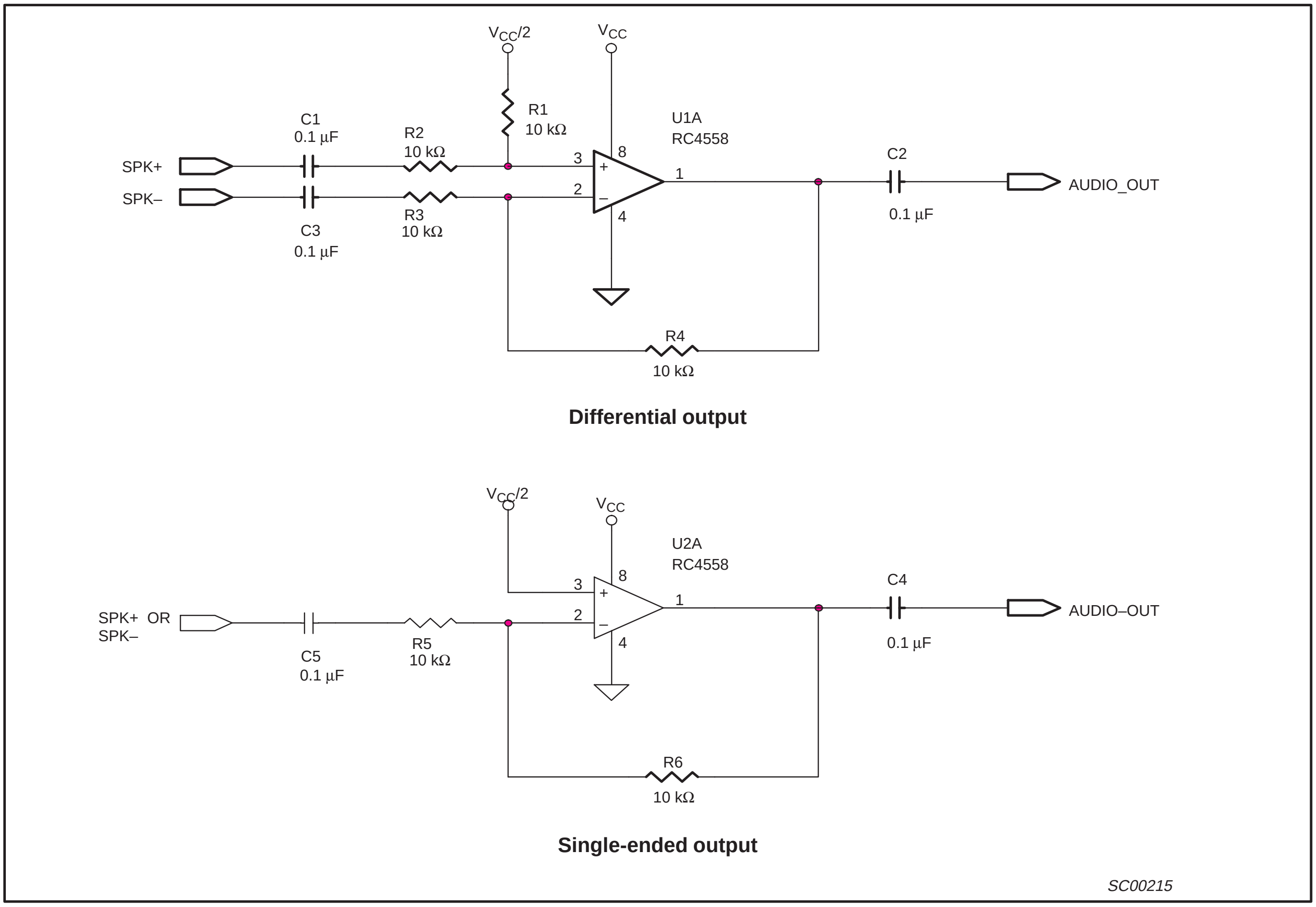

Figure 5. Operational amplifier circuits

MICROPHONE SELECTION

There are two references for Sound Pressure Level commonly used for microphone output level specifications. These are +74 dBspl and +94 dBspl. The unit dBspl represents dB sound pressure level, a standard unit in acoustical measurements. For a given SPL (Sound Pressure Level) measured in Pa (Pascal), dBspl is related to SPL by the following equation:

$$dBspl = 20\log(SPL/20\mu)$$

Thus, +94 dBspl corresponds to one Pascal (Pa). In addition, 1Pa = 10 μBar (microbars).

The microphone output signal level is given in dB reference to 1 V (dBV) or 1 mW (dBm). A typical microphone sensitivity specification might read: –74 dBm / μBar. This means that when a sound source puts out one μBar (74 dBspl) at a distance of 0.5 meter from the microphone, this microphone will produce a power 74 dB below one milliWatt. To determine the signal voltage that this power corresponds to, we need to know the load impedance.

A more useful form of sensitivity specification is:

–54 dBV / Pa.

This sensitivity means that when a sound source puts out one Pa (94 dBspl) at a distance of 50 centimeters from the microphone, this microphone will produce a voltage 54 dB below one volt. To get the exact output voltage the following equation can be used:

$$-54 = 20 \log (V_{out} / 1\ V)$$

$$V_{out} = 10 \wedge (-54 / 20) * 1V = 2\ mV$$

Microphones are linear devices, so if we double the sound pressure, the voltage output from this microphone will also increase by two. For an example, if the sound source produces 88 dBspl, using the equation which relates dBspl and SPL, we can calculate the SPL (in Pa) equivalent of 88 dBspl:

$$SPL = 10 \wedge (88 / 20) * 20\ \mu(Pa) = 0.5\ (Pa)$$

The voltage output from this microphone with 88 dBspl input is:

$$2\ mV * 0.5 = 1\ mV.$$

For HPC (handheld PC) applications where the device is held close to a speaker’s mouth (within 50 mm) during recording, we recommend a microphone with sensitivity of at least –45 dBV/Pa to produce adequate play back sound level. For the application where the device is held more than 50 mm away from the speaker’s mouth, we recommend the user to use an external amplifier to amplify the microphone output signal.

NOTES

Definitions

Short-form specification — The data in a short-form specification is extracted from a full data sheet with the same type number and title. For detailed information see the relevant data sheet or data handbook.

Limiting values definition — Limiting values given are in accordance with the Absolute Maximum Rating System (IEC 134). Stress above one or more of the limiting values may cause permanent damage to the device. These are stress ratings only and operation of the device at these or at any other conditions above those given in the Characteristics sections of the specification is not implied. Exposure to limiting values for extended periods may affect device reliability.

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