Hi3516A/Hi3516D Audio Optimization Application Notes

Issue 02

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About This Document

Purpose

This document describes the Hi3516A/Hi3516D audio optimization solutions.

NOTE

This document uses the Hi3516A as an example. Unless otherwise specified, this document applies to the Hi3516D and Hi3516A.

Related Versions

The following table lists the product versions related to this document.

Product Name	Version
Hi3516A	V100
Hi3516D	V100

Intended Audience

This document is intended for:

- Technical support engineers
- Board hardware development engineers

Symbol Conventions

The symbols that may be found in this document are defined as follows.

Symbol	Description
DANGER	Alerts you to a high risk hazard that could, if not avoided, result in serious injury or death.

Symbol	Description
WARNING	Alerts you to a medium or low risk hazard that could, if not avoided, result in moderate or minor injury.
CAUTION	Alerts you to a potentially hazardous situation that could, if not avoided, result in equipment damage, data loss, performance deterioration, or unanticipated results.
TIP	Provides a tip that may help you solve a problem or save time.
NOTE	Provides additional information to emphasize or supplement important points in the main text.

Change History

Changes between document issues are cumulative. Therefore, the latest document issue contains all changes made in previous issues.

Issue 02 (2015-10-30)

This issue is the second official release, which incorporates the following changes:

Chapter 2 Solution

The descriptions in section 2.1.4 and section 2.2 are modified.

Issue 01 (2014-12-19)

This issue is the first official release.

Contents

Figures

1 Overview

The audio optimization solutions aim to provide excellent audio quality during development based on the Hi3516A. This document describes the precautions to be taken related to audio software and hardware.

2 Solution

2.1 Circuit Solution

2.1.1 MIC Input Circuit

The audio input (AI) pins for the Hi3516A include only the line-in pins (AC_LINEL and AC_LINER) and the MICBIAS pin (AC_MICBIAS) for the microphone (MIC) input, as shown in Figure 2-1.

Figure 2-1 AI pins

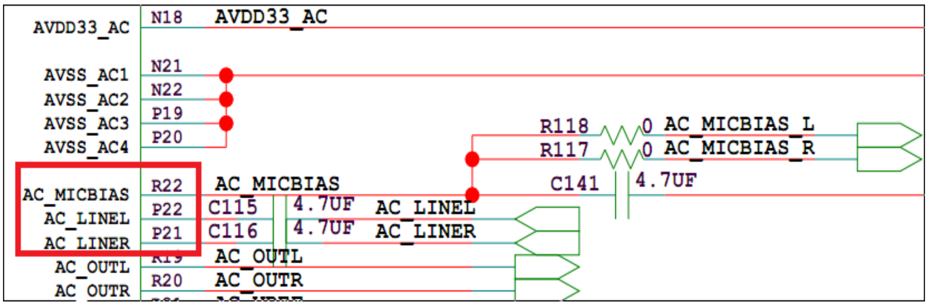
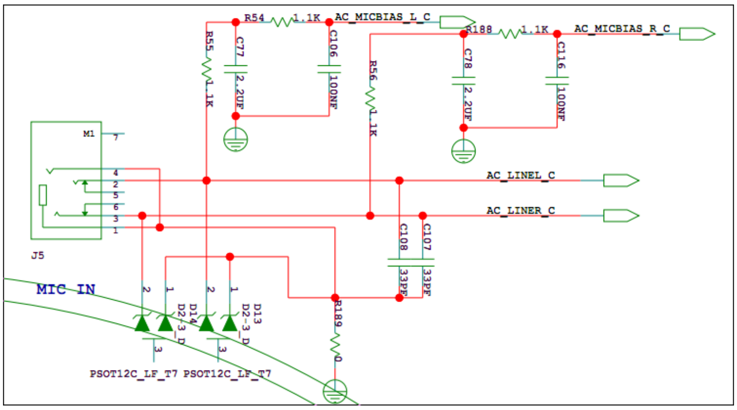


Figure 2-2 shows the recommended MIC input circuit. There are two MIC inputs. You can select only one MIC input as required.

Figure 2-2 MIC input circuit

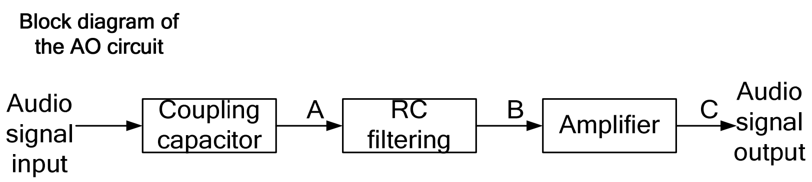


Note that the MIC input signals must be isolated before they pass the main chip. The 4.7 µF ceramic DC blocking capacitors (C115 and C116 in Figure 2-1) are recommended. On the printed circuit board (PCB), the DC blocking capacitors must be placed close to the pins of the Hi3516A.

2.1.2 Analog Audio Output Circuit

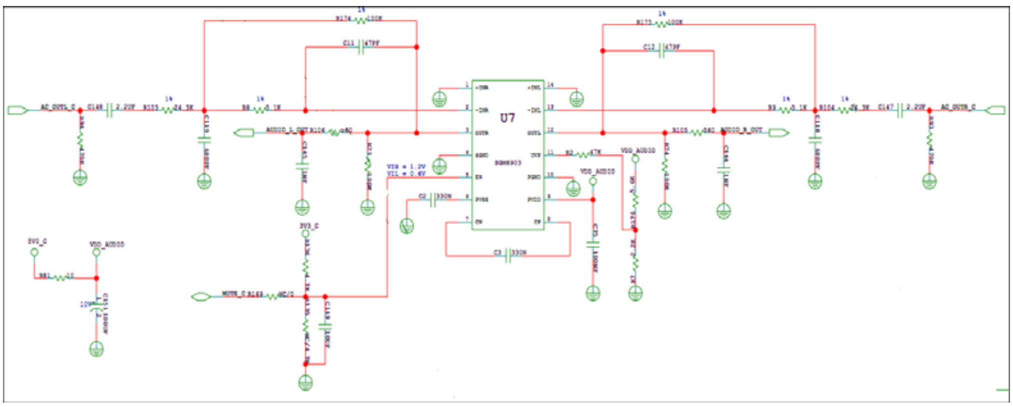
Figure 2-3 shows the analog audio output (AO) processing of the Hi3516A, which is similar to that of the IP camera (IPC).

Figure 2-3 Analog AO processing on the board



The recommended external operational amplifier for the circuit on the board is SGM8903, which can suppress crackles. Figure 2-4 shows the analog AO amplifier and the filter circuit on the board.

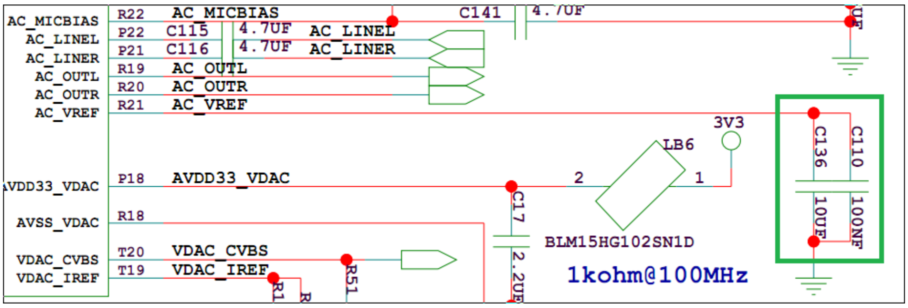
Figure 2-4 Analog AO amplifier and filter circuit on the board



2.1.3 Precaution on Pin Configuration of the Analog Audio Module

The analog audio module configuration of the Hi3516A differs slightly from that of the Hi3518A in the AC_VREF pin. For the Hi3516A, the AC_VREF pin needs to connect to only one 10 µF ceramic capacitor and one 100 nF ceramic capacitor in parallel and then to GND, as shown in Figure 2-5.

Figure 2-5 AC_VREF connection for the Hi3516A



2.1.4 Precautions During the Audio Circuit PCB Design

Note the following:

- The audio signal should be kept away from the digital signal to prevent interference.
- The return path of the audio signal (including the MICBIAS signal) must be connected to ground (GND) and must not be shared with other signals especially the digital signal. Vias must be punched on the GND of audio module and connected to the system GND directly. Note that the signals of audio GND and other analog GNDs must not be connected and vias that are connected to the system GND must not be shared.
- You are advised not to split the analog GND of audio module and the system GND and use single point grounding mode. If you want to separate the audio analog GND, use single point grounding mode and ensure a complete audio analog GND plane and sufficient analog GND vias. The signal traces must connect to the audio analog GND through GND vias.
- Parameters of the MICBIAS circuit and the Hi3516A demo board must be consistent.

2.2 Optimization of the Echo Structure for Audio Intercom

If the Hi3516A is used for the IPC that features small size and high integration, the distance between the MIC and handset may be too small. When the audio gain is large, the speaker exerts severe interference on the MIC input (the echo cancellation function must be enabled), and therefore echoes are generated during intercom. To solve this problem, note the following during product structure design:

- Keep the MIC far away from the speaker and control the angle between the MIC and the speaker to minimize the coupling of voice signals.

- Enclose the MIC chamber to prevent voices from passing through mechanical parts to the MIC. Enclose the speaker as required.
- In terms of structure, the area of the speaker sound chamber opening needs to be greater than 15% of the area of the speaker vibrating diaphragm. The opening is as large as possible or the density of the opening is as high as possible on the condition that the dust proofing and appearance are guaranteed. The distance between the speaker vibrating diaphragm and the cover cannot be too long. The optimal distance is the maximum value of the vibrating amplitude for the vibrating diaphragm plus a certain margin. If the distance is too long, a front chamber is generated and the sound effect is affected, especially when the opening density is not high enough. For the distance between the maximum vibrating amplitude for the vibrating diaphragm and the inner wall of the cover, the recommended empirical value is 1 mm and the maximum value is 1.2 mm. The distance must be set to an appropriate value to prevent the vibrating diaphragm from hitting the cover during vibration.
- Generally, the sound effect of the medium- and low-frequency parts of the voice is improved and the power is decreased when the sound chamber of the speaker is enlarged. Therefore, the sound chamber of the speaker is expected to be large. However, other factors also need to be considered for the structure design of the product. In terms of the structure, a shock pad (rubber material) that is thick enough is required for the sound chamber of the speaker. The recommended empirical value of the thickness is 1.2 mm. Note that the shock pad needs to enclose all the surroundings of the speaker as well as the front of the speaker (excluding the vibrating diaphragm of the speaker) to absorb shock for the front part of the speaker.
- Open round holes with a diameter of 0.8–1.2 mm on the MIC. Control the size of the front sound chamber of the MIC and reserve one straight hole.

  Enclose the MIC by using rubber, foam, or a voice guide sleeve (preferable). If the MIC is not enclosed, voice may leak from the speaker to the MIC, increasing the difficulty of echo cancellation. Generally, the voice guide sleeve is assembled on the product to prevent voice leakage, provide sealing effect, and avoid resonance.
- Ensure that the sound chamber mechanical part is designed with the anti-vibration function during the speaker structure design to prevent the vibration passing from the cover of the mechanical part to the RX end of the MIC. In addition, the sound chamber must be sound-proofing to avoid crosstalk caused by sound leak of the sound chamber. The recommended empirical value for the thickness of the cylinder wall of the shock pad (rubber material) for the MIC components is 1.5 mm.

2.3 Audio Register Configuration

2.3.1 Audio Input/Output Gain Control

The audio input gain includes the digital gain control and analog gain control.

The digital gain cannot be greater than 4 dB. The corresponding register is MISC_CTRL54, and the register address is 0x201200D8. Take the digital gain of the audio-left channel for example, MISC_CTRL54 bit [30:24] are used to control the digital gain input from audio-to-digital converter (ADC) of the audio-left channel. For details, see the *Hi3516A HD IP Camera SoC User Guide.*

Theoretically, the maximum analog gain of the audio input is 56 dB. The corresponding register is MISC_CTRL50, and the register address is 0x201200C8. Take the analog gain of the audio-left channel for example (similarly in the following), MISC_CTRL50 bit [14:10] are used to control the gain_mic of the audio-left channel, and the maximum gain is 30 dB.

The bit [9] is the gain_boost control bit of the audio-left channel. If bit 9 is **0**, there is no gain amplification; if bit 9 is **1**, there is 26 dB gain amplification.

The usage of gain_mic and gain_boost is as follows: When the input gain is less than or equal to 30 dB, you can set the gain value as required by configuring MISC_CTRL50 bit[14:10]. When the required input gain is greater than 30 dB, enable the gain_boost to obtain a gain of 26 dB, and then configure MISC_CTRL50 bit[14:10] to obtain the remaining required gain value. The required gain value is the sum of the 26 dB and the gain value obtained by configured MISC_CTRL50 bit[14:10].

Note that as the level of the MIC input signal is not large, gain_boost is typically used for the MIC input. The gain_boost function cannot be dynamically enabled or disabled when the audio module is working, that is, the gain_boost function can be enabled only during initialization. If the gain range specified by configuring Gain_mic meets requirements, Gain_mic is preferred. For the line-in input, the gain_boost function is not used because the audio amplitude of the linear input is large (clipping distortion occurs if the gain is too large).

To obtain an appropriate AO volume, you can set an appropriate output gain value by configuring MISC_CTRL53 bit[30:24]. Typically, when the output gain is 0 dB, the requirements on the volume of the AO on the board can be met. The maximum AO gain is 6 dB. Note that the output gain value cannot be too large; otherwise, crackles may occur.

2.4 AI/AO Interfaces and Functions

2.4.1 AI/AO Gain Interfaces in the Old Solution

The audio gain adjustment interfaces are classified by register in the old solution. The interfaces for gain adjustment are as follows:

- ACODEC_SET_GAIN_MICL/ACODEC_SET_GAIN_MICR: adjusts the analog gain of the input audio-left or audio-right channel. The ACODEC_SET_GAIN_MICL interface corresponds to bit[14:10] of MISC_CTRL50 (address of 0x201200C8), that is, the gain_micl field. The audio-left channel is used as an example. For details about the audio-right channel, see the description of MISC_CTRL50. The corresponding bits in MISC_CTRL50 are configured according to the required gain values, and then the configured register value is transferred by calling the corresponding ioctl interface.
- ACODEC_SET_ADCL_VOL/ACODEC_SET_ADCR_VOL: adjusts the digital gain of the input audio-left or audio-right channel. The ACODEC_SET_ADCL_VOL interface corresponds to bit[30:24] of MISC_CTRL54 (address of 0x201200D8), that is, the adcl_vol field. The corresponding bits in MISC_CTRL54 are configured according to the required gain values, and then the configured register value is transferred by calling the corresponding ioctl interface. Note: To implement low-noise audio effect, it is recommended that the analog gain be used to control the AI gain.
- ACODEC_SET_MIXER_MIC: selects the line_in or mic_in mode. When the mic_in mode is selected, gain_boost (bit[9] of the register with the address of 0x201200C8 for the audio-left channel) is enabled, and the analog gain increases by 26 dB. When the line_in mode is selected, the analog gain is not increased.
- ACODEC_SET_DACL_VOL/ACODEC_SET_DACR_VOL: adjusts the digital gain of the output audio-left or audio-right channel. The ACODEC_SET_DACL_VOL interface corresponds to bit[30:24] of MISC_CTRL53 (address of 0x201200D4), that is, the dacl_vol field. The corresponding bits in MISC_CTRL53 are configured according to the required gain values, and then the configured register value is transferred by calling the corresponding ioctl interface. Note: The maximum analog AO gain is 6 dB. However, 0

dB output gain can meet the requirements (the amplification multiple of the external operational amplifier circuit needs to be taken into account to avoid crackles caused by clipping). You can adjust the AO gain based on actual application scenarios.

2.4.2 AI/AO Gain Interfaces in the New Solution

In the new solution, the gains are adjusted adaptively by configuring corresponding registers to minimize background noises. Four AI/AO gain adjustment interfaces are added in the new solution, which are implemented by calling the ioctl interface of the audio CODEC. The new interfaces are described as follows:

- ACODEC_SET_INPUT_VOL: configures the AI gain. The value range of the parameter is [−87 dB, +86 dB]. Both the analog gain and digital gain are included. A larger value indicates higher volume. For example, the value 86 indicates the maximum volume of 86 dB, and the value −87 indicates the minimum volume (muted status). The volume adjustment takes effect simultaneously in the audio-left and audio-right channels. The recommended volume range is [10 dB, 56 dB]. Within this range, the noises are lowest because only the analog gain is adjusted, and the voice quality can be ensured.
- ACODEC_SET_OUTPUT_VOL: configures the AO gain. The value range of the parameter is [−121 dB, +6 dB]. A larger value indicates higher volume. For example, the value 6 indicates the maximum volume of 6 dB, and the value −121 indicates the minimum volume (muted status). The volume adjustment takes effect simultaneously in the audio-left and audio-right channels. This interface is used to adjust the AO digital gain. Typically, 0 dB gain can meet requirements. When you adjust the gain, take the amplification multiple of the external operational amplifier circuit on the board into account. The output gain value cannot be too large; otherwise, crackles may occur due to clipping.
- ACODEC_GET_INPUT_VOL: obtains the AI gain. This interface is used to obtain the gain value configured by using the ACODEC_SET_INPUT_VOL interface. The value range of the AI gain is [−87 dB, +86 dB].
- ACODEC_GET_OUTPUT_VOL: obtains the AO gain. This interface is used to obtain the gain value configured by using the ACODEC_SET_ OUTPUT _VOL interface. The value range of the AO gain is [−121 dB, +6 dB].

2.4.3 Mapping Between Interfaces in Old and New Solutions

The ACODEC_SET_INPUT_VOL interface is added, which can be used to configure all the gain values configured by all the previous interfaces for AI gain adjustment. The following describes the mapping between old and new interfaces:

- ACODEC_SET_GAIN_MICL/ACODEC_SET_GAIN_MICR: adjusts the AI analog gain. For the new interface, the value range of the AI analog gain is [10 dB, 56 dB]. You are advised to use the new interface ACODEC_SET_INPUT_VOL to configure the AI gain.
- ACODEC_SET_ADCL_VOL/ACODEC_SET_ADCR_VOR: adjusts the digital gain. For the new interface, the value range of the digital gain is [56 dB, 86 dB] or [−87 dB, +10 dB].
- ACODEC_SET_MIXER_MIC: selects the line_in or mic_in mode. When the mic_in mode is selected, gain_boost is enabled, and the analog gain increases by 26 dB. When the line_in mode is selected, gain_boost is disabled, and the analog gain is not increased. For the new interface, the gain_boost is always enabled after the AI volume is configured, and the line_in or mic_in mode does not need to be selected.

The new AO gain adjustment interface ACODEC_SET_OUTPUT_VOL corresponds to the ACODEC_SET_DACL_VOL/ACODEC_SET_DACR_VOL interface in the old solution. The transferred parameter values are the gain values for the new interface, whereas the transferred parameters values are the register values for the interface in the old solution. The corresponding bits in the register are configured according to the required gain values, and then the configured values are transferred by calling the interface.

To ensure version compatibility, the previous interfaces can remain unchanged, and the new and previous interfaces can be configured simultaneously. However, you are advised not to use them at the same time. The ACODEC_GET_INPUT_VOL interface obtains only the gain values configured by using the ACODEC_SET_INPUT_VOL interface but not the gain values configured by other previous interfaces.

2.5 Interface for the Audio 3A Algorithm

The interface for the audio 3A algorithm remains unchanged. For details, see the description of the interfaces for the audio 3A algorithm in the *HiMPP V2.0 Media Processing Software Development Reference*.

Note that there are three application modes for the audio echo cancellation (AEC) in the actual application scenarios, including the headset mode, receiver mode, and speaker mode. The details are as follows:

- In headset mode, the AEC algorithm can be disabled.
- In receiver mode, the AEC algorithm needs to be enabled. This is the default and recommended mode for the IPC.
- The speaker mode is recommended when the AO is connected to a sound box and echoes are large.

For details, see the description of AI_AEC_CONFIG_S in the *HiMPP V2.0 Media Processing Software Development Reference*.

3 Summary

The audio quality is optimal when the following conditions are met:

- The audio amplifier and filter circuit are used in the external audio circuits. For details, see the latest schematic diagrams.
- The recommended crackle suppression audio amplifiers such as SGM8903 are used.
- The echo cancellation function is enabled and the product structure meets the requirements for canceling echoes generated in audio intercom.
- The registers must be properly configured, and the interfaces and functions must be correctly called.