Rockchip Microphone Array Test Reference Document

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Preface

Overview

This document serves as a reference for microphone array testing, providing engineers with guidance on conducting microphone array tests.

Product Version

Chipset	Kernel Version
ALL	/

Intended Audience

This document (this guide) is mainly intended for:

Technical support engineers

Revision History

Version	Author	Date	Change Description
V1.0.0	Maofa Li,Binyuan Lan	2021-01-18	Initial version

Contents

1. Test Environment and Tools

1.1 Test Environment

1. Less than 40dB of local noise and a reverberation time between 200-500ms.
2. No obstructions around the device.

1.2 Test Tools

1. High-fidelity speaker --- Genelec's G series high-fidelity speakers are recommended.
2. Sound level meter --- Class 2 sound level meter or above.
3. Audio analysis software --- Audition

1.3 Test Audio

1. THD and frequency response test audio --- swp_48k_16b.wav
2. Sealing test audio --- white_noise.wav
3. 1kHz audio --- Sine1k_48k_16b.wav
4. Mute audio --- mute.wav

2. Speaker Test

The speaker test mainly tests whether the speaker meets product requirements (this item can be ignored if there is no playback requirement).

2.1 Power Test

When the speaker is at maximum power, it should meet the requirement that the data collected by the microphone does not show clipping, and the sound pressure measured at the microphone does not exceed 90 dB. Play the audio file swp_48k_16b.wav at 100% volume, save the data, and observe the amplitude through Audition to ensure that there is no clipping phenomenon for each frequency. Play the audio file white_noise.wav at 100% volume to ensure that the measured sound pressure at the microphone does not exceed 90 dB.

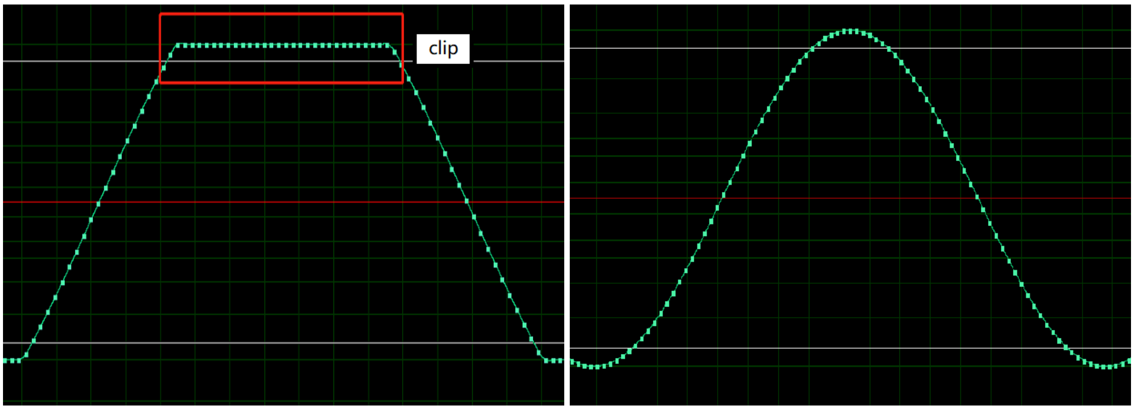

Figure 1 Schematic diagram of clipped waveform (left) and normal waveform (right).

If clipping occurs, analyze the dB value at which clipping occurs and adjust the speaker power until there is no clipping. If the measured sound pressure at the microphone exceeds 90 dB, the speaker power should be appropriately reduced.

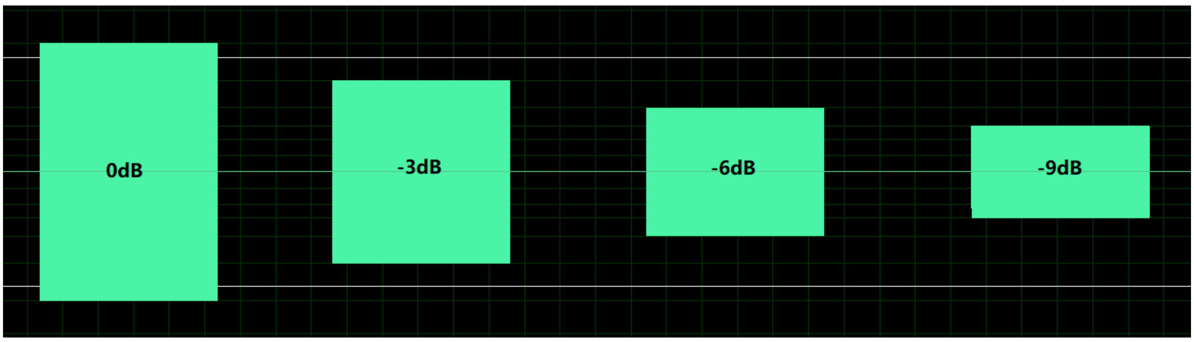

Figure 2 Schematic diagram of sweep signal

2.2 Speaker total harmonic distortion test

When the speaker outputs at maximum volume, the THD meets the following conditions: less than 10% at 100Hz, less than 6% at 200Hz, and less than 5% above 350Hz. If the low-frequency distortion exceeds 10%, it is recommended to add a filter to filter out the low-frequency part.

The total harmonic distortion of the speaker is defined as follows:

$$THD = \sqrt{\sum_{i=2}^{m}(\frac{G_i}{G_1})^2}$$

In the formula,$$THD$$is the total harmonic distortion, $$ G_i $$ is the RMS value of each harmonic,and$$m$$is the harmonic order.

The THD test is carried out in a semi-anechoic or full chamber. Play the sweep signal through the speaker and use an external microphone to obtain the speaker audio signal at the same time. By analyzing the collected signal, the fundamental frequency and the RMS value of corresponding multiple frequency points are obtained, and the THD value is calculated using the formula. If the test result does not meet the THD value, the speaker and EQ need to be modified to meet the requirements.

If there is no professional equipment, the energy of the fundamental frequency and the first few harmonic frequencies in the recording file can be calculated through Audition to roughly analyze the THD level of the speaker, as shown in Figure 4. Assuming that the fundamental wave is -30 dB, the second harmonic is -64 dB, the third harmonic is -46 dB, and the rest of the harmonics are ignored, then the THD is approximately 16%. The calculation process is as follows:

$$THD = \sqrt{({\frac{G_2}{G_1}})^2+({\frac{G_3}{G_1}})^2} = \sqrt{10^{-6.4+3}+10^{-4.6+3}} \approx 16\%$$

Calculate the THD of different frequency points. Connecting multiple points together can obtain the THD curve.

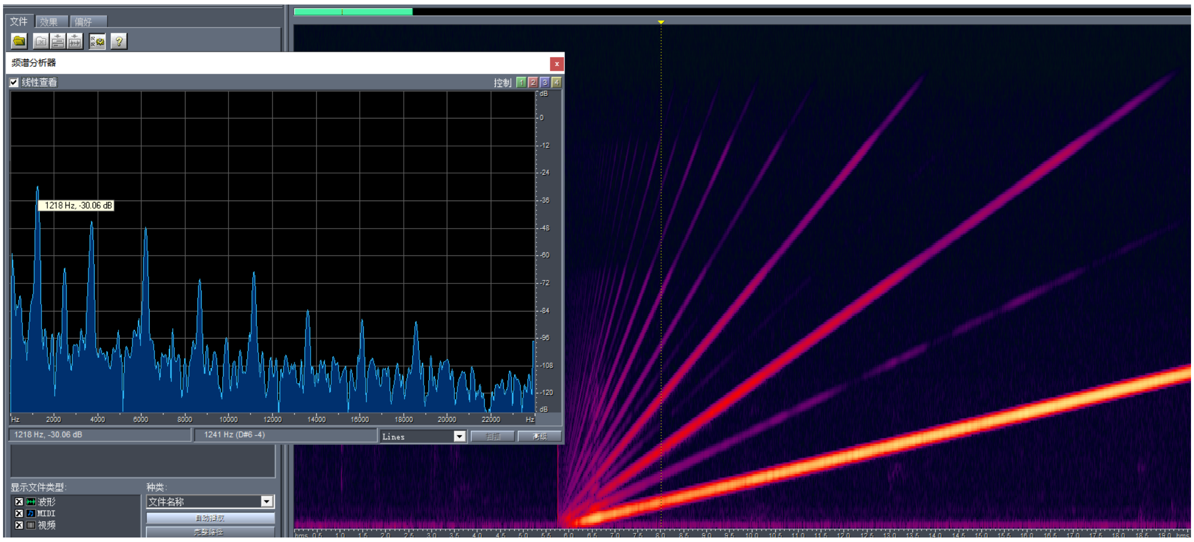

Figure 3 Schematic diagram of sweep signal

2.3 Background noise tast

When the machine plays mute.wav, record the audio and the audio amplitude of the loop at this time. The background noise is less than -65 dBFS.

When the machine is not playing audio, record the audio and the audio amplitude of the loop at this time. The background noise is less than -65 dBFS.

3. Microphone Test

3.1 Basic microphone test

3.1.1 Clipping check

1. Using an internal speaker, it should be ensured that under the maximum volume test, the sweep signal does not clip and the sound pressure from the speaker to the microphone does not exceed 90 dB (measured at the microphone).
2. Using an external speaker for testing, the speaker can be placed 1 meter away from the microphone array for measurement. When the measured sound pressure at the microphone does not exceed 90 dB, there is no clipping problem.

3.1.2 Signal-to-noise ratio test

The signal-to-noise ratio test mainly tests the far-field pickup ability of the microphone. The higher the signal-to-noise ratio, the stronger the pickup ability.

Test in a semi-anechoic or full chamber. Play white noise through a high-fidelity speaker and adjust the sound pressure level at the microphone to reach 63 dBA and record for 10 seconds; stop playing and record for another 10 seconds. Use Audition to analyze the speech energy of the blank section and the non-blank section to test the signal-to-noise ratio. The signal-to-noise ratio of the full frequency band is required to be above 20 dB.

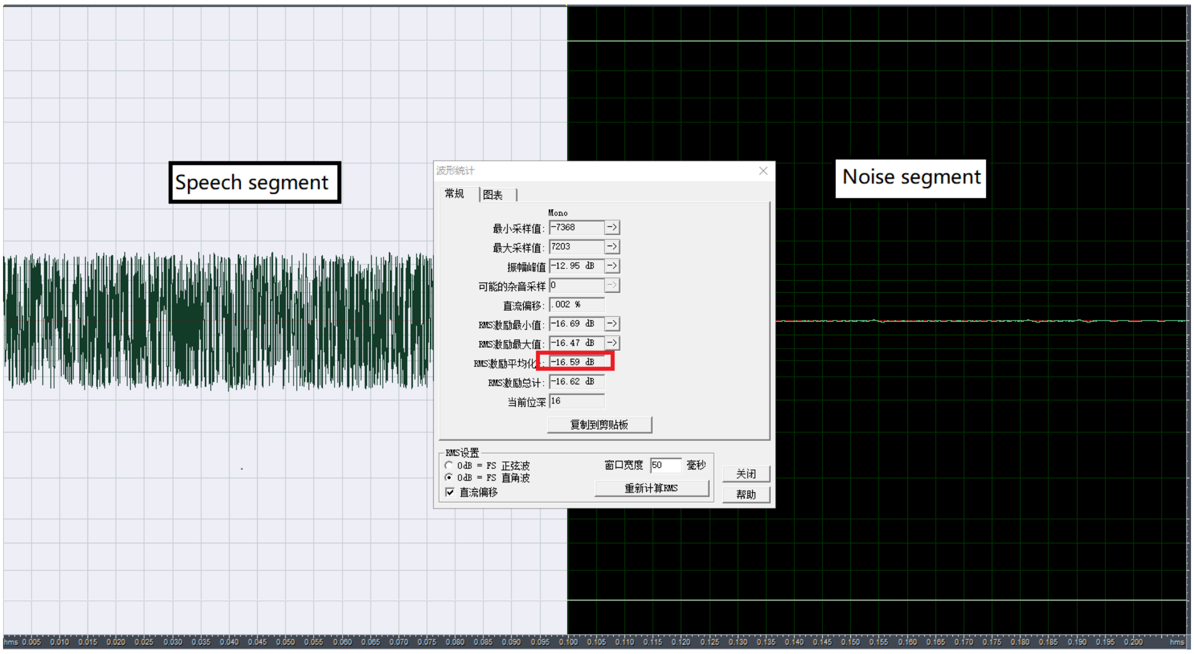

Figure 4 Schematic diagram of signal-to-noise ratio

3.1.3 Sealing test

Sealing test is mainly to test whether sound is transmitted through the air.

Place the external speaker 1 meter away from the microphone and play white noise. The measured sound pressure at the microphone is about 80 dB. Play the audio source and use black adhesive tape or rubber adhesive to block the mic hole in turn for about 10 seconds. Measure the RMS value difference of the signal received by the microphone before and after to obtain the sealing performance of the microphone.

Microphone array should have at least 15 dB or more of sound insulation performance.

Figure 5 is a schematic diagram of the sealing test. Block the mic in turn to measure the sealing performance before and after blocking.

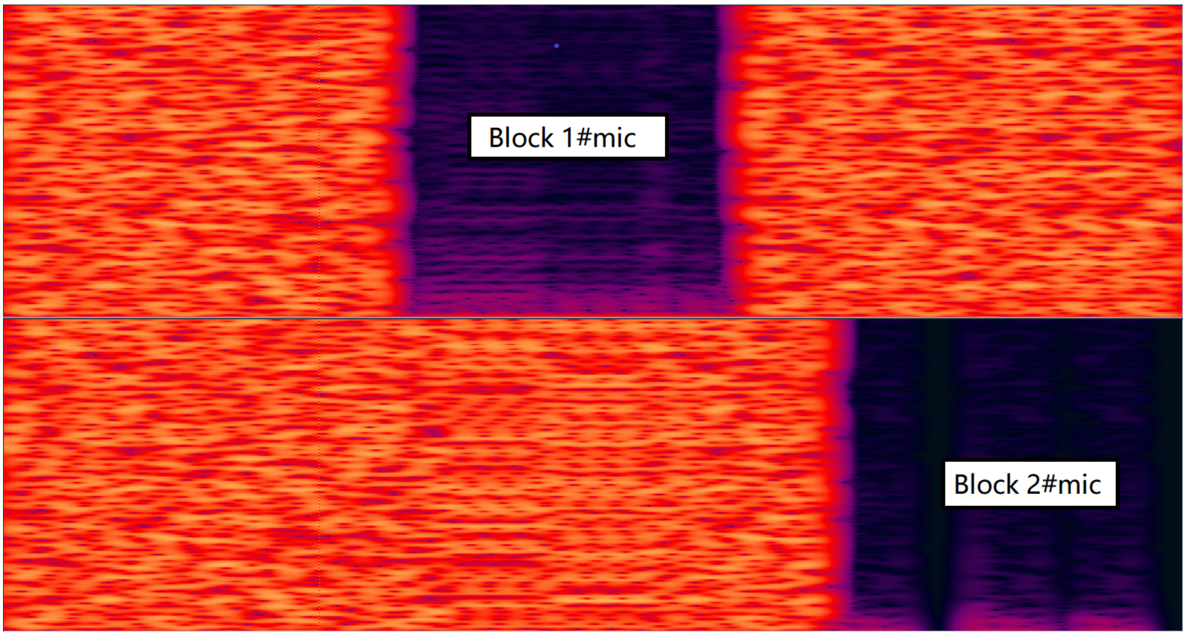

Figure 5 Schematic diagram of sealing test

3.2 Consistency test

3.2.1 Amplitude consistency

Amplitude consistency refers to the ability of microphones to maintain a basic consistent signal amplitude when receiving the same signal.

When playing the swp_48k_16b.wav at 100% volume, observe the amplitudes of multiple mics to ensure that the amplitudes of each mic are basically the same.

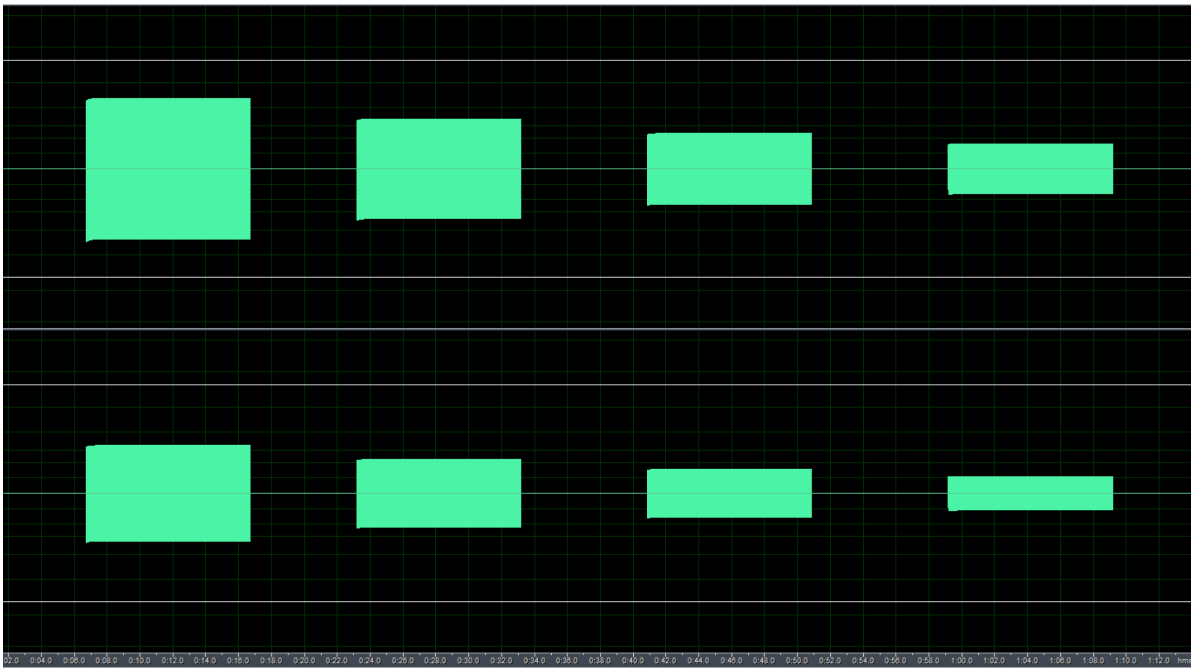

Figure 6 Schematic diagram of amplitude consistency (unqualified)

Amplitude consistency is directly related to frequency response consistency. If the amplitude difference is large in different frequency bands, the amplitude difference of each microphone at the corresponding frequency can be accurately obtained by testing the frequency response curve of each microphone to ensure that the frequency response difference of each microphone is within $\pm1dB$.

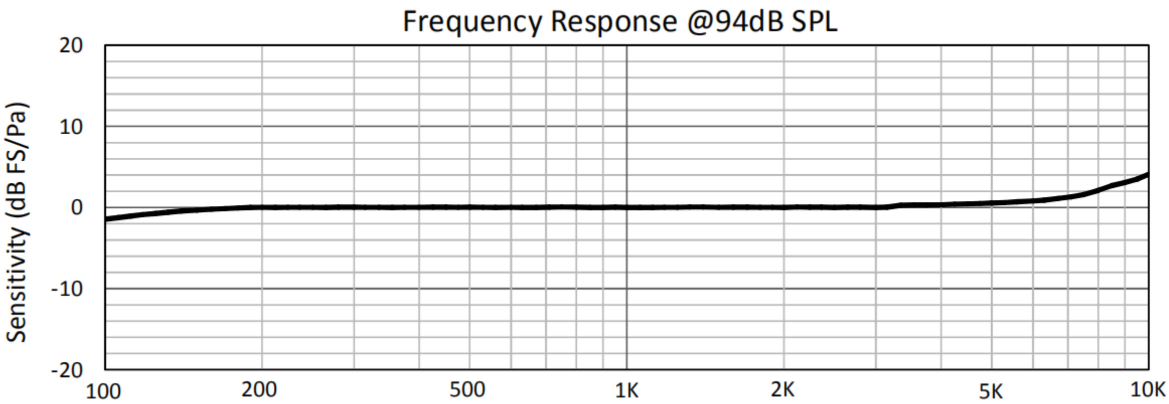

Figure 7 Schematic diagram of frequency response curve

3.2.2 Phase consistency

Phase consistency can also be understood as delay consistency. For an active audio source, different incident angles reach different microphones on the microphone array at different times, so there is a delay between each microphone and this delay remains stable.

Assuming that the microphone array is a 2-mic linear array with a distance of 3.5 cm, the time difference of 90-degree incidence is 0 ms, and the time difference of 0-degree incidence is about 0.103 ms. Assuming a 48 kHz sampling rate, the 90-degree incidence is approximately 0 sampling point delay, and the 0-degree incidence is approximately 5 sampling point delay.

For the 90-degree test, place the audio source at 1 meter and align it with the center of the microphone array. Play a 1k sine wave and calculate the signal delay between each MIC.

For the 0-degree test, place the audio source at 1 meter and align it with one side of the microphone array. Play a 1k sine wave and calculate the signal delay between each MIC.

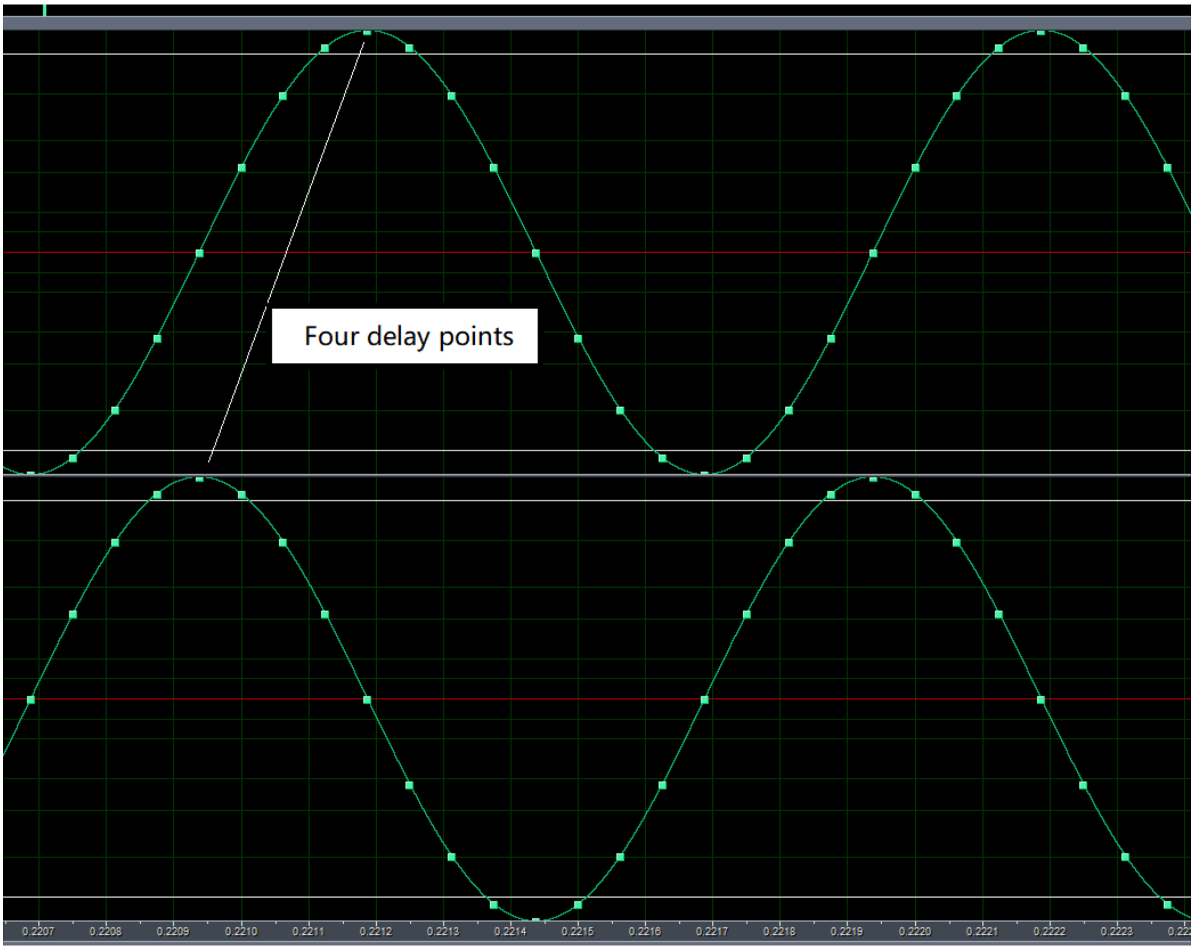

Figure 8 Schematic diagram of phase consistency

3.2.3 Channel sequence check

The microphone order should correspond to their physical arrangement. For a left-to-right linear microphone array, the PCM sequence aligns with the microphone numbers: 1#, 2#, 3#, 4# (or in reverse)mic data, followed by 5# and 6# loopback data. Alternatively, the sequence can start with 5# and 6# loopback data, then proceed with 1#, 2#, 3#, 4# (or in reverse). See Figure 9 for the channel sequence illustration.

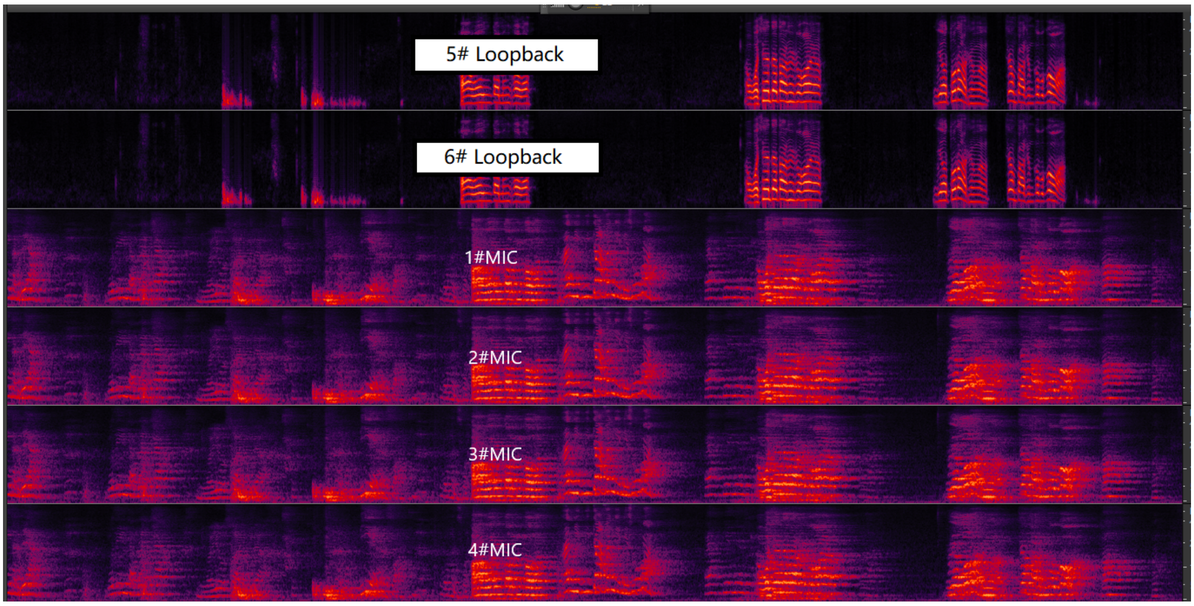

Figure 9 Schematic diagram of channel sequence

4. Appendix Test Forms

Test Item	Test Audio Source	Test Result(PASS/FAIL)
Power test	swp_48k_16b.wav white_noise.wav	
Speaker THD test	swp_48k_16b.wav	
MIC channel clipping check	swp_48k_16b.wav	
Signal-to-noise ratio test	white_noise.wav	
Sealing test	white_noise.wav	
Amplitude consistency	swp_48k_16b.wav	
Phase consistency	Sine1k_48k_16b.wav	
Microphone channel sequence		
Microphone/loopback channel DC offset		
Loopback background noise check		
Constant frequency interference check		
Overall machine vibration/resonance/abnormal sound check		

Refer to the details in the xlsx file：[Requirement-list_audio-algorithm_rk3xxx_company.xlsx]