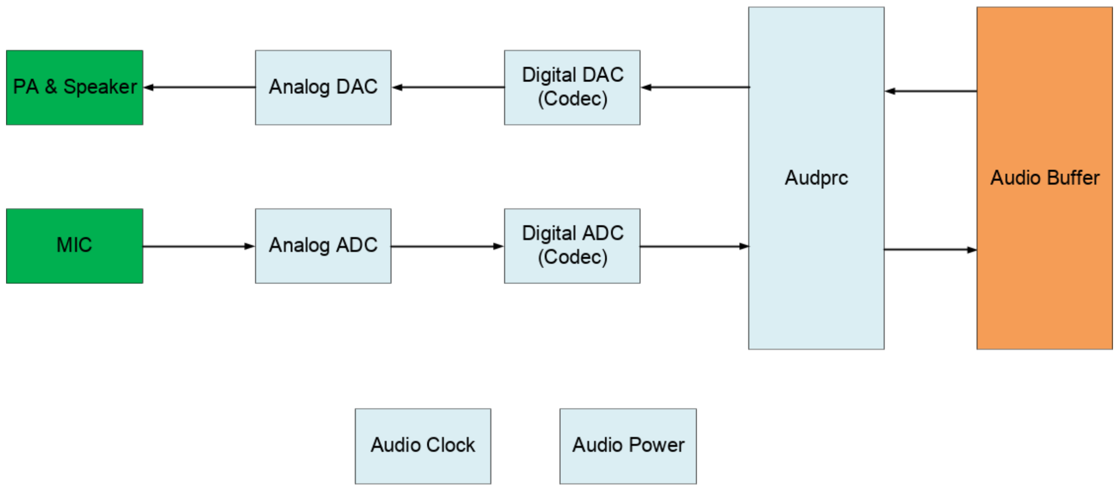
Bluetooth Audio Debugging Guide

Basic pathways and concepts



The audio path is typically composed of several modules as shown in the figure above. Among them, Audio Buffer represents the buffer for audio transmission and reception prepared by software. The green PA&Speaker represents the external power amplifier and speaker. MIC represents the external microphone. The remaining modules are all internal audio hardware modules of the chip.

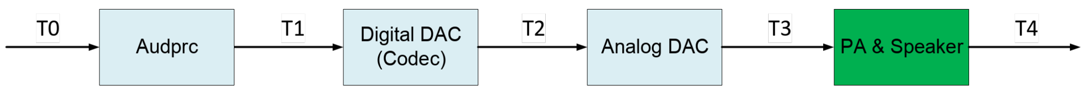
Sampling rate: Sampling rate refers to the frequency range of audio data. Common sampling rates include 8KHz/16KHz for telephone calls, and 44.1KHz/48KHz for music playback.

Frequency doubling: Frequency doubling refers to the occurrence of a single-tone signal at a certain frequency and its integer multiples in the frequency spectrum. This phenomenon is usually caused by nonlinearity in the circuit. For example, the frequency doubling of a 1KHz signal includes 2KHz, 3KHz, 4KHz, and so on. When frequency doubling is detected, priority is generally given to investigating the source of the single-tone signal at the lowest frequency point.

Saturation: Saturation refers to the clipping of the peak value of a single-tone sine wave due to limited output range.

Debugging process:

一、 DAC channel test



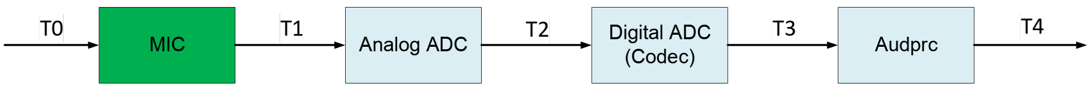
As shown in the figure, the basic DAC (Digital-to-Analog Converter) path testing process requires the preparation of original audio material by software, which is then passed through the entire audio path until it is played through the loudspeaker. Specific testing requirements are detailed in the table below:

sampling rate	audio material	Effect at the speaker
16KHz	Normal voice	The sound is clear, without any static or other noise except for background noise
16KHz	Single tone at	The signal is free from harmonics and distortion, with

	1KHz	clear sound, and there is no other noise except for background noise
16KHz	mute	There is no other sound except for the background noise
44.1KHz	Normal voice	The sound is clear, without any static or other noise except for background noise
44.1KHz	Single tone at 1KHz	The signal is free from harmonics and distortion, with clear sound, and there is no other noise except for background noise
44.1KHz	mute	There is no other sound except for the background noise

The above six sets of tests can be screened as appropriate, but at least one set at 16KHz and one set at 44.1KHz should be retained. For subsequent software debugging, refer to the DAC analog/digital module register configurations of these two tests. If conditions permit, measurements can be made on the input and output terminals of the PA using an instrument, including noise floor, 1K single-tone response, etc.

二、 ADC path test



As shown in the figure, the ADC path test requires the software to collect audio input from the MIC terminal, save it, and play it back. At this time, it is necessary to ensure that the DAC path test has been completed and there are no issues with the path test. Specific test requirements are outlined in the table below:

sampling rate	audio material	Playback effect after collection
16KHz	Normal voice	The sound is clear, without any noise or other disturbances except for the background noise
16KHz	Single tone at 1KHz	The signal is free from harmonics and distortion, resulting in clear sound with no noise other than background noise
16KHz	Quiet environment	There is no other sound except for the background noise
44.1KHz	Normal voice	The sound is clear, without any noise except for the background noise
44.1KHz	Single tone at 1KHz	The signal is free from harmonics and distortion, producing clear sound with no other noise except for background noise
44.1KHz	Quiet environment	There is no other sound except for the background noise

The above six sets of tests can be screened as appropriate, but it is recommended to retain at least one set of tests at 16KHz. For subsequent software debugging, refer to the DAC analog/digital module register configurations of these two tests. If conditions permit, measurements can be made on the output terminal of the MIC using an instrument, including noise floor, 1K single tone response, etc. At the same time, the software recording can be exported and transmitted to a PC for more detailed analysis.

三、 Bluetooth RF performance test

The watch must pass the Bluetooth radio frequency performance test, ensuring that indicators such as frequency deviation, transmission power, and reception sensitivity meet basic requirements. If there are issues with radio frequency performance, it may interfere with issues such as audio stuttering during debugging.

四、 Bluetooth music test

The mobile phone can search and connect to the watch via Bluetooth. When the mobile phone and the watch are within 1 meter of each other, the mobile phone can play music stably for more than 5 minutes without obvious issues such as pitch changes, noise, crashes, or reboots. If there are obvious abnormalities in the sound playback, attention should be paid to issues related to the clock and sampling rate. If there is noise during playback and the DAC path test is normal, interference from Bluetooth RF or the screen is usually considered. If issues such as crashes or reboots occur after prolonged playback, consider potential problems caused by deviations between the mobile phone clock and the local clock.

五、 Music EQ adjustment

Music EQ is usually adjusted according to customer requirements and in collaboration with the customer.

Music EQ is generally used to adjust the frequency response curve of speakers when playing music, in order to achieve the desired sound quality effect for customers. Speakers vary from customer to customer, and their requirements for sound quality are also different, so each product needs to be individually debugged.

六、 Bluetooth call channel test

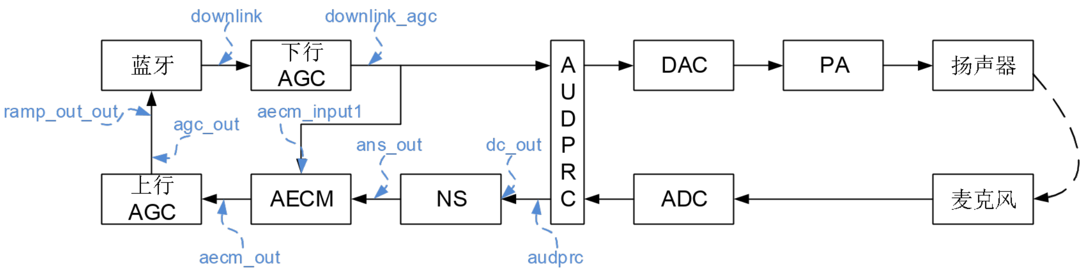
When dialing on the watch, a stable call with the remote phone can last for more than 2 minutes. During both uplink and downlink single-talk, the other end can hear voice without noticeable pitch variation. At this time, issues such as echo, noise, volume, and sound quality are not of concern. When the watch is placed 2 meters away from the nearby phone without any obstruction, and the call audio algorithm is turned off, the other end should not hear noticeable stuttering during uplink and downlink single-talk. Otherwise, it is necessary to check the radio frequency performance or phone signal issues.

七、 Bluetooth call algorithm import

1. Integrate the call algorithm, briefly listen to the general effect, and expect no obvious echoes and noise, with normal volume and sound quality, and no stuttering or word loss during single speech. Regardless of whether the call effect meets expectations at this time, subsequent confirmation of relevant indicators should be carried out.
2. After incorporating the call algorithm, it is necessary to pay attention to the CPU usage during calls, which should be reasonably stable at around 50%. If the CPU usage consistently exceeds 80%, it is abnormal and special attention should be paid to issues such as stuttering and system crashes that may be triggered when the CPU exceeds 100% momentarily. It is not recommended for customers to switch to other interfaces for operations during calls. If customers have such a

mandatory requirement, targeted special testing and optimization are needed.

3. The integration of wired data capture functionality enables stable and long-term data capture of at least one channel on the flying line machine. The debugging of call audio primarily relies on the data captured through wired means.
4. The Bluetooth call flow and wired data capture points are illustrated in the figure below.



八、 Watch speaker volume adjustment

The volume adjustment (downward gain adjustment) of the loudspeaker playback must be debugged together with the customer, and the maximum volume should be confirmed by the customer.

After incorporating the audio algorithm, the input to the DAC is typically a signal with a 16-bit amplitude value of around 16k (-6dB), which can be confirmed through wired data capture using downlink_agc. This signal undergoes gain adjustment by AUDPRC/DAC and is output to the PA, which then amplifies it and sends it to the speaker. Based on customer requirements, the downstream gain is adjusted using an EQ tool, ultimately achieving the desired volume output from the speaker. The downstream gain is primarily adjusted through AUDPRC and DAC, while the PA uses a fixed gain. After the customer adjusts the volume, an oscilloscope should be used to observe that during maximum volume calls, the waveforms of the DAC output and PA output do not show significant saturation, and there should be no noticeable distortion in the speaker sound.

When the volume played by the speaker is too high, various issues such as poor sound quality at the near end, echo at the far end, and stuttering heard at the far end during double-talk may occur. It is necessary to confirm with the customer the final volume requirements for the product, and try to choose a moderate volume, or clearly inform the customer of the negative impacts of excessive volume. Due to different speakers, sound chambers, and isolation levels chosen for each terminal product, the maximum volume that can be tolerated without compromising call quality varies. Therefore, each product requires individual confirmation of the maximum volume with the customer. Subjectively, when the watch is playing at maximum volume, it is preferable for the sound to be clear and bright at a distance of 20cm from the watch without feeling harsh.

The volume of the loudspeaker is also the primary factor affecting the power consumption of calls.

九、 Call EQ adjustment

This process is not mandatory and is typically carried out in collaboration with the customer based on their requirements. If the customer has no specific instructions, EQ can be omitted.

Call EQ is generally used to adjust the frequency response curve of the speaker during calls, in order to achieve the desired sound quality effect for customers. Speakers vary from customer to customer, and their requirements for sound quality are also different, so each product needs to be individually debugged. Call EQ and music EQ cover different frequency ranges and must be debugged separately. The call EQ also has an impact on call power consumption.

十、 Confirmation of call audio metrics

1. Recording volume (recording gain adjustment). This indicator requires wired data capture confirmation and does not require customer participation.

   During the call, mute the microphone of the remote phone, and speak at normal volume at a distance of 15cm from the watch on the near end. Collect data for more than 30 seconds through wired data capture audprc, and observe the amplitude value of the collected data. The target value is around 4~5k.

   If the amplitude does not meet expectations, it is necessary to adjust the recording gain by modifying the registers in AUDPRC to ensure that the amplitude value of the collected voice reaches 4~5k. The initial value of the software should be set to 0x24 (0dB gain), and adjustments should be made based on this.

0x5C		0x0000_0000	adc_path_cfg0	
[31:15]		17'h0	RSVD	
[14]	rw	1'h0	rx2tx_loopback	rx to tx loopback enable
[13]	rw	1'h0	data_swap	swap adc path left and right channel data
[12]	rw	1'h0	src_sel	adc path source select 1'h0: select audio codec 1'h1: select external interface
[11:6]	rw	6'h3f	vol_r	volume control from -18~13dB, step is 0.5dB 6'h0: -18dB 6'h1: -17.5dB ...... 6'h3e: 13dB 6'h3f: mute
[5:0]	rw	6'h3f	vol_l	volume control from -18~13dB, step is 0.5dB 6'h0: -18dB 6'h1: -17.5dB ...... 6'h3e: 13dB 6'h3f: mute

2. The recording has a noisy background. This indicator requires confirmation through wired data capture, and may require the customer to modify the PCB if necessary.

   During the call, if the microphone of the remote phone is muted, the near end needs to be silent in a quiet environment (with the microphone turned on). Collect data for more than 30 seconds through wired data capture (audprc), observe the amplitude values of the collected data, and ensure that the target value is less than 200 and there is no periodic disturbance.

If the noise amplitude is significantly high, it is necessary to investigate potential interference introduced on the board. This indicator can affect the processing of upstream noise and echoes. Common sources of interference may include screen refresh, backlight PWM, Bluetooth RF, etc. The interference can be roughly inferred and eliminated based on the frequency spectrum of the noise.

The noise reduction effect can be confirmed through wired data capture (ans_out). If the data amplitude value is less than 50, it can be considered to have basically met the standard.

3. Echo volume. This indicator requires confirmation through wired or wireless data capture, and may necessitate modifications to the internal structure of the watch by the customer.

   The prerequisite for measuring this indicator is that the speaker playback volume indicator has been confirmed by the customer and the recording volume adjustment has been completed.

   During the call, the near end needs to be silent in a quiet environment (with the microphone turned on) and the playback volume adjusted to maximum. The far end phone should speak at normal volume. Collect data for more than 10 seconds through audprc, and observe the amplitude value of the collected data. The target value should be below 5k.

If the echo amplitude exceeds 5k, it is necessary to consider enhancing the isolation between the microphone and the speaker. For flying wire machines, a loud echo may be caused by poor sealing due to flying wires. If necessary, holes can be drilled on the side of the machine to guide the wires out. Alternatively, the echo volume can be confirmed by wireless data capture of the entire machine.

十一、 Call audio test

Bluetooth call quality is closely related to the mobile phone platform. Different combinations of near-end and far-end phones yield inconsistent results. Therefore, after deploying the Bluetooth call algorithm, multi-platform testing must be conducted, covering at least four combinations: near-end Apple to far-end Apple, near-end Apple to far-end Android, near-end Android to far-end Apple, and near-end Android to far-end Android. It is normal for the sound quality and volume heard by watches on different platforms and far-end phones to vary slightly, but they should all meet the following standards: the sound can be easily heard in a quiet environment, there is no significant stuttering or word loss during speech, no obvious echo, and minimal background noise.

Common problems and debugging methods

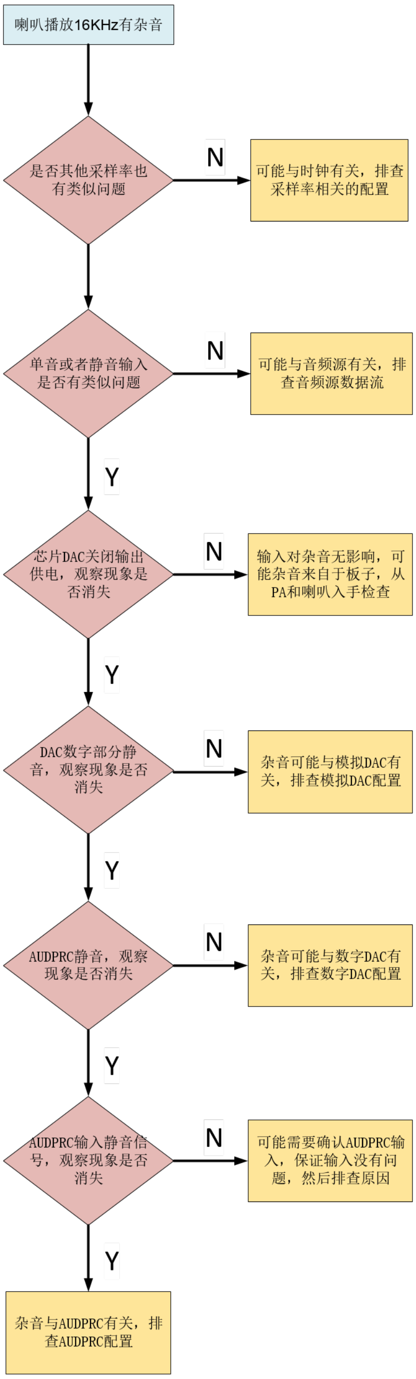
一、 DAC path issue.

Back-to-front principle: Starting from the speaker, deduce step by step forward to find the audio module that introduces the problem. The methods for forward deduction include measuring the output of the previous stage, dumping the data of the previous stage, and muting the output of the previous stage to observe the changes in the current stage.

Principle of complex input to simple input: The sequence from complex to simple input is ordinary audio, single-tone audio, silent audio, and simplified input. Simplified input can reduce the impact of input. Single-tone signals are mainly used to test audio amplitude and saturation issues, while silent signals are used to test audio noise floor.

Considering that all chips have undergone the most basic chip-level audio testing before leaving the factory, the basic audio indicators are guaranteed to some extent. The troubleshooting in testing should focus on the listening experience of the speaker. The following case is an example of troubleshooting speaker noise:

Problem: The loudspeaker produces a buzzing noise when playing 16KHz audio



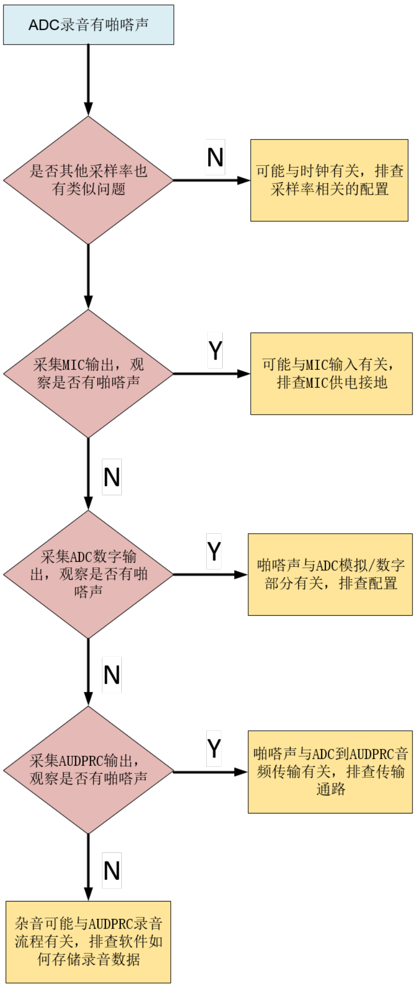
In the image above, the red box indicates the debugging method, while the yellow box represents the feedback conclusions obtained from the debugging results. These conclusions are helpful for further and more precise troubleshooting in the future.

二、 ADC path issue.

Front-to-back principle: Starting from the MIC, inspect the output of each subsequent stage level by level, identify the audio module that introduces the problem, and deduce backwards to ensure that audio output from each stage can be collected. By comparing changes in audio data, the problem can be confirmed.

Principle of Complex Input to Simple Input: The sequence from complex to simple input is ordinary audio, single-tone audio, silent audio, and simplified input. Simplified input can reduce the impact

of input. Single-tone signals are primarily used to test audio amplitude and saturation issues, while silent signals are used to test audio noise floor.
The following case is an example of troubleshooting the crackling sound in ADC recording:



The process of this case also follows the principles of ADC path problem analysis.

三、 Periodic or persistent murmurs and noises.

It is usually caused by interference on the PCB board and requires board-level testing.
Check the power supply and grounding related to audio: The power supply mainly includes the 3.3v for audio, the bias voltage for the microphone (MIC), and the power supply for the external power amplifier (PA). It is necessary to ensure stable voltage during power supply. For certain operations, such as screen refreshing and Bluetooth packet transmission and reception, there may be periodic interference on the VDD. Therefore, it is best to use an application processor (AP) for spectrum analysis during inspection, and locate the source of interference by analyzing typical signal frequencies. The grounding part is similar, and it is also necessary to check whether there is periodic interference on the grounding, starting from the spectrum analysis.

四、 Pop sound.

It refers to the briefly occurring "pop" sound, which is essentially the inconsistency of sound in the time domain.
The focus should be on checking the sequence of turning on and off each module in the audio path to

avoid transient mutations during audio acquisition and output. There are certain principles for the sequence of turning on and off audio modules. For the DAC path, it is necessary to ensure that the previous stage is stable and the volume is sufficiently low before turning on/off the next stage. For example, before switching on/off the PA, it is necessary to ensure that the DAC analog output signal is sufficiently small and stable. For the ADC path, it is necessary to ensure that the previous stage output has stabilized before turning on/off the next stage. For example, when switching on/off the ADC, it is necessary to ensure that the MIC power supply is stable and has entered a normal working state. For pop sounds, especially those heard through the loudspeaker, they can be confirmed by capturing the instantaneous PA output signal with an oscilloscope.
The pop sound may also occur during Bluetooth packet loss, and this factor needs to be eliminated. After PLC compensation, the pop sound caused by Bluetooth packet loss should not be noticeable.

五、 The sound played by the watch speaker is too loud/too soft/distorted.

Check whether the downlink gain configuration is reasonable, including AUDPRC, DAC, and PA.

六、 The sound of the watch speaker is distorted/shrill/vibrato.

Check whether the DAC output and PA output are saturated. Some speakers have poor frequency response, requiring call EQ compensation to adjust the sound quality.

七、 The watch speaker produces continuous noise.

First, mute the microphone of the remote phone. If there is still noise, it is necessary to investigate the issue with the watch's playback path. Ensure that the performance test of the audio playback path passes. If it is confirmed that the noise is coming from the remote end, it is usually due to excessive background noise or specific background noise from the remote phone, and the local volume is also set too high. There is no good solution except to reduce the volume or test with a different remote phone. You can capture downlink data through a wired connection for confirmation.

八、 The watch speaker produces a sharp whistling sound or an echo that gradually fades away.

This issue is typically caused by the close proximity between the remote phone and the nearby watch. It is recommended to ensure sound isolation between the remote and nearby devices during testing. It is advisable to conduct the test in different floors or rooms that are relatively far apart.

九、 The watch speaker produces a brief noise/hum/abnormal sound.

Usually, it is due to some action on the remote phone (such as picking up and putting down, touching the microphone hole, etc.), or sudden noise in the remote environment (such as closing the door, coughing, etc.), which is transmitted to the watch. In such cases, it is necessary to ask the remote person what happened immediately, and the cause can usually be found. Especially when the call is just connected, there are often noises caused by related actions. It is recommended that both ends of the call should not make any movements in the first few seconds before connecting, and listen to see if there are similar noises.

十、 The watch speaker frequently stutters and drops characters when it hears sound.

Check if the watch and the nearby phone are within 1 meter of each other without any obstructions. Try speaking louder at the far end to see if there is any improvement. You can also try switching to the phone's earpiece at the far end to rule out poor phone signal as a factor.

十一、 The speaker of the watch suddenly became silent.
Check whether the microphone on the remote phone is muted, whether the speaking volume is too low, and whether the speaker on the watch is muted. After eliminating the above reasons, it is necessary to record the on-site log to analyze the software status. It is best to capture downlink data through wired connection for confirmation.

十二、 The watch speaker suddenly emitted sounds like an electric drill and a Transformers robot.
Confirm whether it is a seesaw scenario. It is necessary to record the on-site log to analyze the software status, and it is best to capture downlink data through wired means for confirmation.

十三、 The sound heard by the remote phone is too low.
Focus on checking whether the recording gain is set correctly, which can be confirmed by capturing agc_out data through wired connection. Check whether the microphone through-hole of the watch is blocked, which is usually located on the left or right side of the watch. Check the volume settings of the remote phone.

十四、 The sound heard on the remote phone is stuttering and missing words.
First, it is necessary to confirm that the distance between the near-end speaker and the watch is within 20cm, and that the through-hole of the watch's microphone is not blocked. It is also important to distinguish between single-speaker and dual-speaker scenarios. In the case of single-speaker, mute the microphone of the remote phone first. If the sound is still choppy, the main suspicion is algorithm deployment issues or algorithm compatibility issues, and the software code needs to be checked. In the case of dual-speaker, choppiness is a known issue and there is currently no effective solution. Speaking closer to the watch or with a louder voice during near-end speech can alleviate the problem to some extent.

十五、 The sound quality heard by the remote phone is poor, with a sharp/dull/distant tone.
This is usually caused by individual differences in the remote phone. You can replace the phone or compare it with a competing watch. If there is a significant difference compared to the competing watch, a case-by-case analysis is required.

十六、 The remote phone heard a clear echo that lasted for a long time.
Echo testing needs to be conducted on a complete machine with good sealing, as there may be significant echoes from the flying wire machine. During testing, confirm that there are no other items covering the watch and that the audio algorithm has been correctly deployed. It is necessary to confirm whether the echo volume index meets the standard, and capture the three channels of data: ans_out, aecm_input1, and aecm_out for algorithm analysis.

十七、 Occasionally, the remote phone hears a brief echo.
It is normal for a brief echo to occur when a call is just connected, after adjusting the volume of the watch speaker, or when the watch is covered. As long as the echo does not persist, it is not a significant issue.

十八、 The remote phone detects continuous noise.

First, mute the microphone of the watch to eliminate interference from the remote phone itself. Then, the noise present when neither end is speaking is generally the louder background noise from the near end. If the noise only exists when the near end is speaking, it is also the smaller background noise from the near end, but it is filtered out when not speaking. If the noise lasts for a few seconds only after the remote end starts speaking and disappears when the remote end stops speaking, it is actually echo residual. If a customer reports a noise-related issue, it is first necessary to determine which type of noise mentioned above it is, and then analyze and solve it accordingly.

十九、 The remote phone heard a brief noise/noise/abnormal sound.

Usually, it is due to an action on the watch side (such as picking up or putting down, touching the microphone hole, etc.), or sudden noise in the near-end environment (such as closing a door, coughing, etc.) that is transmitted to the far-end. When this happens, it is necessary to ask the near-end person what happened immediately, and the cause can usually be found. Especially when the call is just connected, there are often noises caused by related actions. It is recommended that both ends of the call should not make any movements in the first few seconds before connecting, and listen to see if there are similar noises.

二十、 The remote phone suddenly lost sound.

Check whether the remote phone is muted and whether the microphone on the watch is muted. After excluding the above reasons, record the on-site log to analyze the software status. You can capture the ramp_out_out data through wired connection for confirmation.

二十一、 The remote phone heard a "buzzing" electrical current sound.

The current sound typically occurs after speaking from a distant end, indicating a problem with echo suppression. It is necessary to check the deployment of the audio algorithm. In severe cases, it is required to confirm whether the echo volume index meets the standard. Data from the three channels: ans_out, aecm_input1, and aecm_out, should be captured via wired methods for algorithm analysis.

二十二、 After a certain duration of the call, there is a sudden and noticeable audio glitch or freeze.

The main suspicion is that the audio buffer overflow is caused by the error between the Bluetooth clock and the watch clock. It is necessary to check the clock tracking algorithm against the software log.

二十三、 UI stutters during a call

The call algorithm occupies a portion of CPU resources, thus affecting the UI frame rate to some extent. Generally, it is not recommended for customers to grant permission for UI switching during calls. If the customer has such a mandatory requirement, targeted special testing and optimization (such as limiting full-screen animation playback) are needed to ensure the frame rate meets basic requirements.