Smart Audio Processor

User Manual

TS-3400MIXD

Before using this system, please read this manual carefully

Notification

WARNING

To ensure the reliability of the equipment and the safety of personnel, please observe the following when installing, using and maintaining:

- If any of the following conditions are found, please immediately turn off the power, plug out and quickly contact your nearest dealer. Do not continue using this unit, which may cause a fire or electric shock.

  - If you find smoke or have a strange taste from the machine.
  - If water or metal falls into the machine.
  - If the unit is dropped or the case is damaged.
  - If the wire is damaged (wire core exposure, broken wire, etc.).

- If the machine contains high-pressure parts, in order to avoid the fire or electric shock, absolutely don’t open the case, if any questions please inform your nearest dealer.
- Do not place cups, bowls, vases or metal and other water-filled substances on the unit. Serious spilled liquid may cause a fire or electric shock.

- Never expose the unit to rain and any moisture or water, which may cause electric shock or fire.

- Do not place metal objects or flammable materials from the vents on the machine cover, nor place coins, which may cause fire or electric shock.

- Do not place heavy objects on the unit to avoid personal injury or property damage when the unit is slipping.

- Make sure that the volume is turned on at the beginning of the boot, and the high volume of the boot may cause hearing problems.

- Make sure that the volume is turned on at the beginning of the boot, and the high volume of the boot may cause hearing problems.

- For long-term accumulation of dust to be cleaned, please inform your dealer to regularly clean the machine, so as to avoid damage to the machine or cause a fire.

- The battery must be replaced with the same type of product and the correct installation should be made in order to avoid electrical damage and explosion hazard.

- The product is a Class I device. The device must be well connected to ground. The power plug must be connected to a power outlet with a grounding device to ensure that the equipment is fully grounded.

● This product uses a power plug or appliance input socket as a disconnecting device with the power supply, and must be disconnected if necessary for safety reasons.

● 

This equipment is only suitable for safe use at altitudes under 2000 meters.

Precautions

1. The installation environment

When installing the unit, in order to ensure the normal cooling of the host, should avoid the poor ventilation of the place or high temperature environment, to avoid direct sunlight.

Recommend to install cabinet or other well-ventilated place indoor. If you use the machine in the outdoors, please pay attention to waterproof, moisture, lightning protection measures.

Avoid installing in a violent place of vibration; do not place other equipment on the machine.

2. To avoid electric shock and fire

Do not touch the hands and the source with wet hands

Do not spill liquid on the machine, so as to avoid short-circuit or fire inside the machine.

Do not place other equipment directly on the top of the unit.

Non-professional service personnel Do not disassemble the unit yourself to avoid damage and electric shock.

3. Transport and handling

The packaging of the machine is designed and tested to ensure that the host will not be accidentally damaged during transport. It is best to use the original packaging when handling the unit.

Do not move the host device between the place or cold or over hot to avoid condensation inside the machine, affecting equipment life.

4. Please follow the warning instructions on this product, the warning signs on behalf of:

	Applicable to 2000 meters above sea level and below safe use
	Safe use only in non-tropical climates

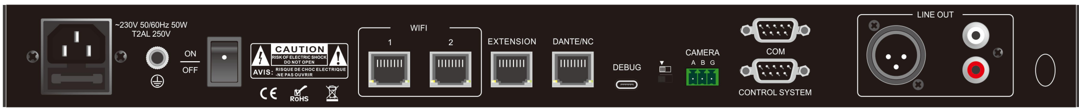
5. Agreement

Please strictly follow the instructions in this manual. The software, hardware and appearance of this product will be upgraded and updated continually. The above changes will be made without notice.

Non-professional maintenance personnel, do not remove the product, to avoid damage and electric shock.

Content

1. System Introduction



It is a professional audio processor with Dante interface that is applicable to digital conference systems. It supports intelligent mixing to solve the problem of howling when multiple microphones are turned on at the same time, and supports audio processing functions such as automatic gain and feedback suppression.

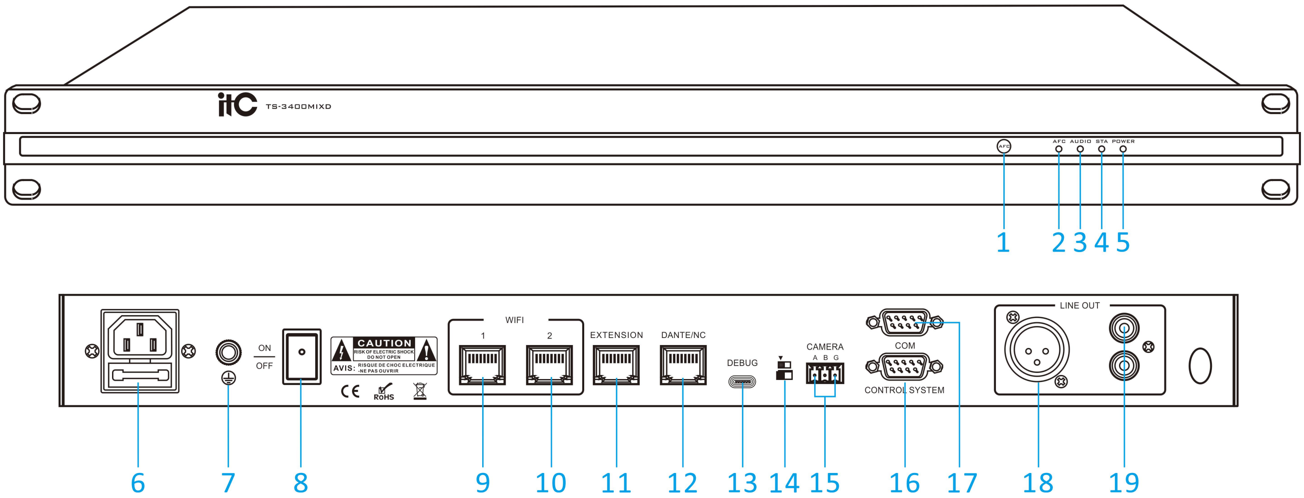
2. Product Profile

2.1 Features

1. The panel is designed with AFC touch keys and working indicators.
2. Equipped with 2 network ports for connecting wireless AP and communicating with the conference server; connect to the digital conference server through network protocol to realize audio data transmission.
3. With auto mixing functions, including gain sharing type auto mixing and gate type auto mixing.
4. Support AFC feedback suppression function, adopt dual notch + frequency shift method, automatically grab the howling point and set the notch frequency, the notch filter supports 12 fixed points + 12 dynamic points to effectively eliminate the howling function.
5. With automatic gain function, it can effectively keep the microphone volume within a certain dynamic range.
6. With the microphone voice activation function, set the tracking threshold, and the camera tracking function can be realized when the microphone speech reaches the threshold.
7. Working with the digital conference server, with the auto mixing function, it can support up to 16 wired microphones and 8 wireless microphones at the same time.
8. With 1 XLR balanced output and 1 RCA unbalanced output.
9. With 1 EXTENSION interface, used to connect the extension port of the digital conference server.
10. With 1 RS-232 communication interface (camera tracking), connected to the central controller or the camera tracking controller to realize the speech camera tracking function.

11. With 1 RS-485 communication interface, used to connect cameras to achieve camera tracking.
12. With 1 RS-232 communication interface (speech transcription), used to connect a speech transcription server to realize the function of speech transcription.
13. With Dante interface.

2.2 Front/Rear Panel Introduction



1.AFC touch key

2.AFC function working indicator

3.Audio signal light, with audio output, light flashing;

4.Working status indicator;

5.Power indicator;

6.AC power plug;

7.Cabinet grounding post;

8.Power switch;

9/10. WIFI network port, used to connect the wireless AP and the communication of the new conference server;

11. EXTENSION port, connected to the extension port of the conference server;

12. DANTE port, connected to external DANTE equipment;

13. Typ-c software recording interface;

14. Recording DIP switch, turn it in the direction of the arrow during normal use;

15. Camera 485 communication interface;

16. Camera tracking RS-232 communication interface;

17. Speech transcription RS232 communication interface;

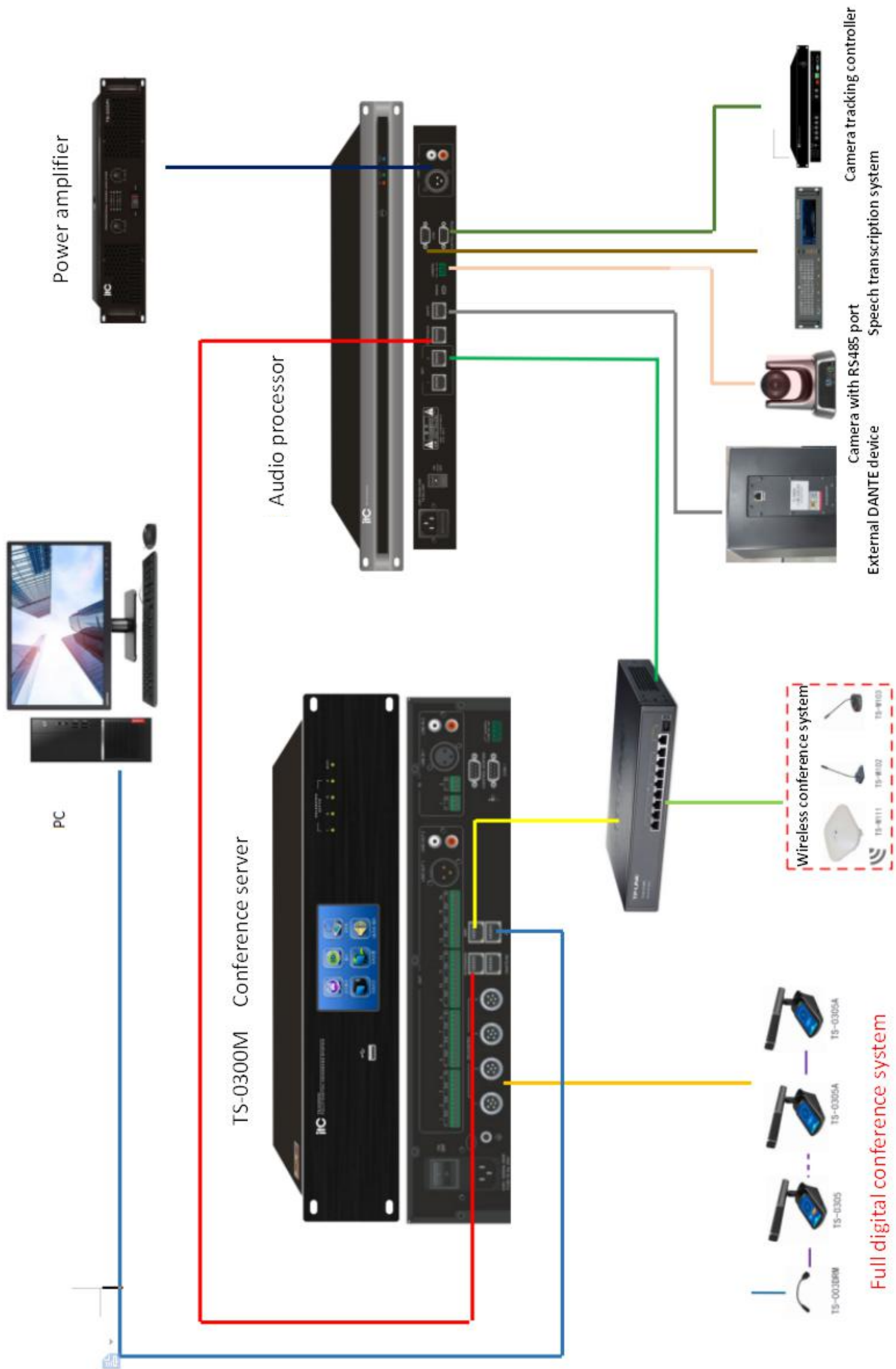
18. Line audio XLR balanced output;

19. Line audio RCA unbalanced output.

2.3 Specification

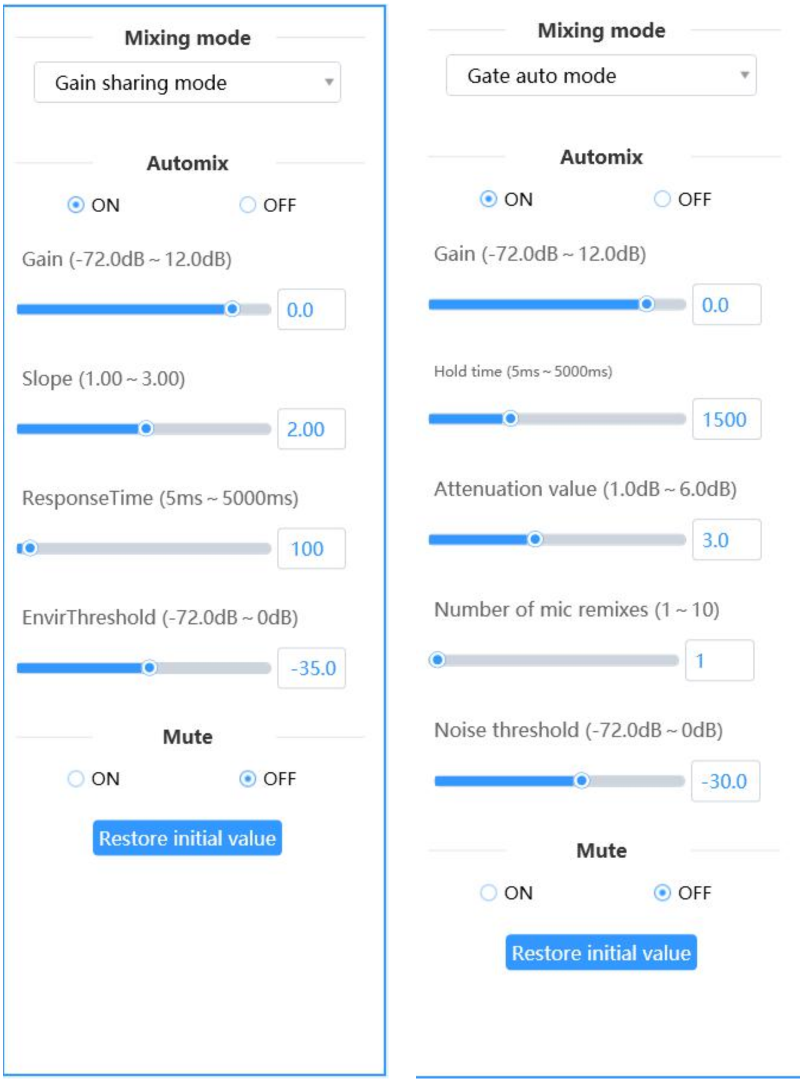
Model	**TS-3400MIXD**
Number of open microphone	16 wired mics + 8 wireless mics
Frequency response	80Hz ~ 16kHz
SNR	> 75dB(A)
Dynamic range	>75dB(A)
THD	<0.05%
Main power supply	100-240AC/50-60Hz
Audio output	LINE OUT 1: 1V XLR balanced output ; LINE OUT 2: 1V RCA unbalanced output
Output load	>1KΩ
Static power consumption	3.3W
Connection method	RJ45 network port
Indicator light	AFC function indicator, audio signal indicator, working status indicator, power indicator
Working temperature	-10℃~+60℃
Working humidity	20%～80% relative humidity, no condensation
Weight	2.51Kg
Dimension (L×W×H)	484×214×44mm
Installation method	19-inch standard cabinet

3. Connection Diagram



4. Operating Instruction

4.1 Smart Mix



4.1.1 There are two mixing modes: Gain Sharing Mode and Gate Auto Mode. This function can effectively improve the sound transmission gain when multiple microphones are turned on at the same time. The gain sharing mode automatically assigns the gain of one microphone to each microphone. When no one is speaking, each microphone is evenly distributed with the gain of one microphone. When only one person is speaking, the microphones that speak will receive all of this sound transmission gain. When multiple persons are speaking, the gain will be distributed according to how loud each person speaks, the louder the more. The gate auto mode is developed based on the noise gate, each channel has a noise gate, the channel is either open or closed.

The application scenario is as follows: turn on multiple microphones at the same time, perform comparison by connecting to the line out of the server, and the mixer can be pulled higher without causing microphone howling.

4.1.2 Parameter information:

(1) Main control parameter

A. Gain Sharing Mode

On and Off: Turn on or off the mixing function. The effect of turning off the mixing function is the same as the Line out output of the three-generation conference server, that is, the direct superimposed output.

Gain: The overall output gain of the automixer. Control the volume level of the mix output.

Slope: Similar to the extension ratio of the extender, the larger the slope value is, the more the sound transmission gain is obtained by the speaking microphone, the more the sound transmission gain is attenuated by the non-speaking microphone, and the smaller the slope value is, the less the speaking microphone obtains the sound transmission gain. Other non-speaking microphones have less attenuation, but at the same time the more the total attenuation. A common setting is 2 or approximately 2; if set to 1.0, the effect is equivalent to turning off automix for all channels; when set it to 3.0, it will cause a larger gain adjustment, possibly produce unnatural effects. The higher the value, the more channels are opened, and the more the total attenuation is.

Response Time: The time to obtain full transmission gain when a microphone is speaking or the time to decay the microphone gain of other microphones that are not speaking. The longer the time is set, the longer the time for the speaking microphone to obtain all the sound transmission gain, and the longer the time for other non-speaking microphones to attenuate the sound transmission gain. And vice versa. Therefore, if the time is set too short, it will cause the sound to be unnatural, and if it is too long, the prefix may be swallowed.

Ambient Threshold: The threshold of ambient noise received by the microphone. When multiple microphones are turned on, only one microphone is speaking, and this microphone cannot get most of the gain, because other microphones that are not speaking at this time are distributed with a certain percentage of gain due to the ambient sound, and the more microphones are turned on, the ratio bigger. The purpose of this parameter is to identify which microphones are speaking, and the speaking microphone will get most of the gain regardless of the number of microphones on.

Mute: Turn mute on or off. Control the mute function of the mix output.

B. Gate Auto Mode

On and Off: Turn on or off the mixing function. The effect of turning off the mixing function is the same as the Line out output of the three-generation conference server, that is, the direct superimposed output.

Gain: The overall output gain of the automixer. Control the volume level of the mix output.

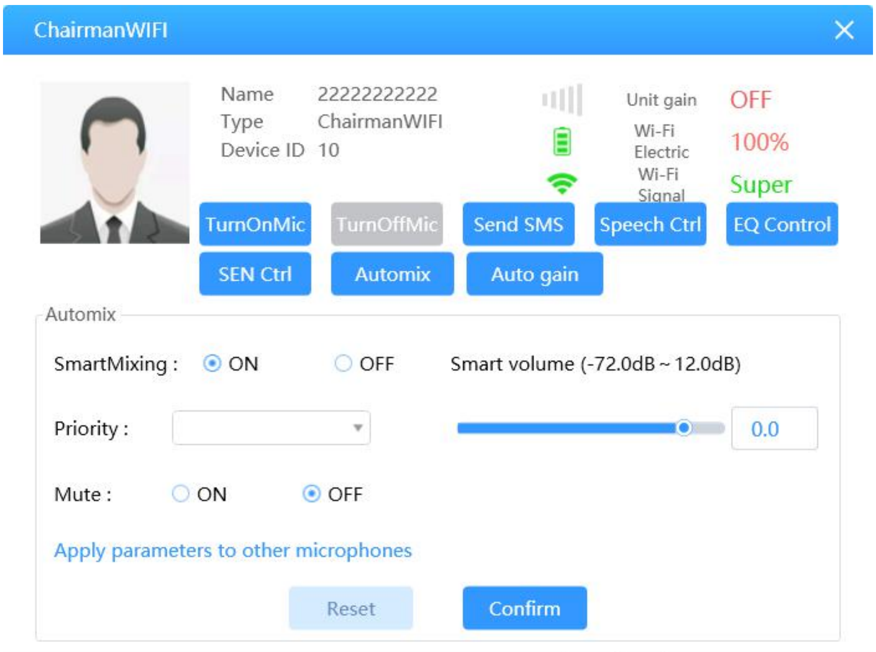
Hold Time: You can adjust the length of time the channel remains open after the microphone stops speaking. Make sure that short pauses between words and phrases when the speaker is speaking does not cause the channel to close. If this parameter is set too short, it will cause the sound to fluctuate loudly. The gate auto mixing supports preemption. For example, when the number of microphone mixing is set to 1, two microphones are speaking at the same time, and the speaker who speaks louder will preempt the microphone. This parameter, at this time, is to ensure that the newly preempted microphone does not cause the channel to close when the speech is paused.

Attenuation Amount: The Attenuation fader affects how much all channels are attenuated when more mics are turned on. According to the NOMG (Number of Open Microphone Gain) law, when NOM (NOM refers to the number of open microphones at the same time) is doubled, the total gain will increase by 3dB. If multiple microphones are turned on at the same time, it is easy to cause acoustic feedback howling or system overload. This parameter reflects the attenuation when both mics are turned on, and the additional attenuation that will be added for each doubling of the number of open mics. Generally set to 3db.

Number of Mic Mixes: The maximum number of Mic Mixes is set. That is, if it is set to 2, the sound of at most two microphones will be output. Even if the sound of more than two microphones exceeds the noise threshold and is turned on, only two of the microphones will be selected for mixed output according to the microphone priority.

Noise Threshold: When the microphone volume exceeds this value, the channel is turned on, and if it is less than this value, the channel is turned off.

（2）Channel parameter

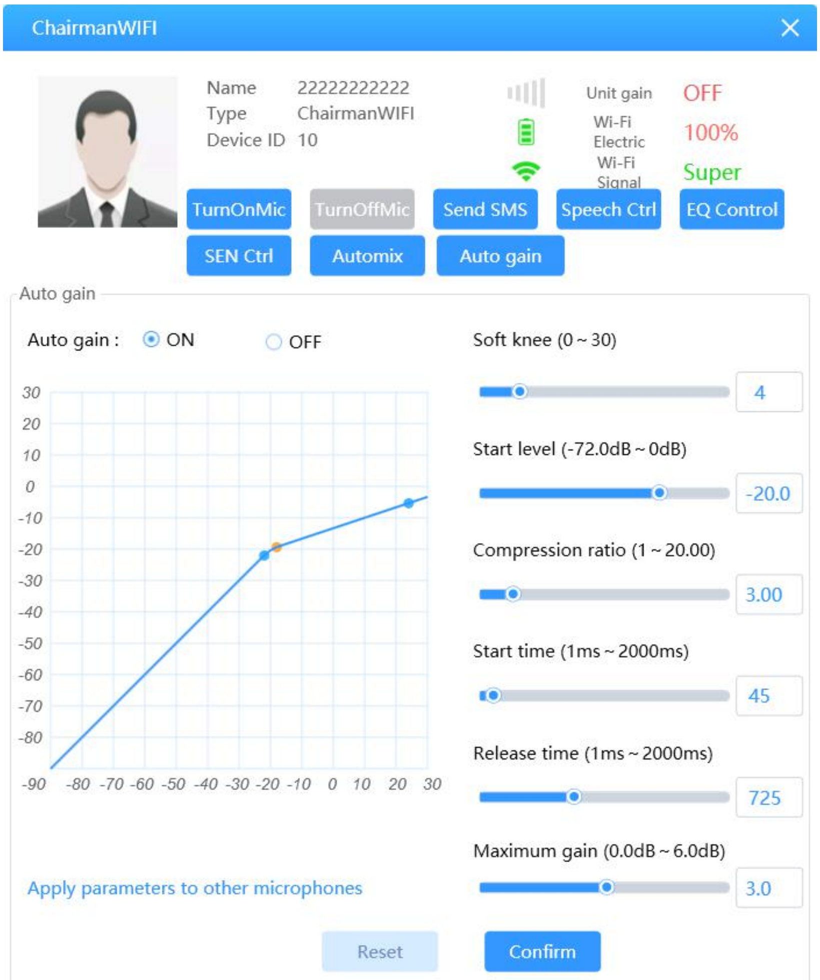


Smart Mix: Automix on/off button, the channel that needs automix should turn on this button. It can also be turned off and the channel does not need automix. Channels that do not participate in the automix will be directly superimposed with the output of the automix and then output.

Priority: The high-priority channel takes precedence over the low-priority channel, thereby affecting the automix algorithm. The parameter ranges from 1 to 10. The larger the value, the higher the priority. The effect of this parameter on gain sharing mixing is that when the sound of the two microphones is the same, the higher priority is assigned with more gain than the lower priority. The effect on the gate auto mixing is that when the sound of two microphones exceeds the threshold, the one with higher priority will be selected for mixing first.

Mute: This microphone will be muted. It will still participate in the smart mix, but the sound of the microphone will not be output.

4.2 Auto Gain



4.2.1 The purpose of the auto gain is to maintain the signal at a target level while maintaining dynamics. The application scenario is as follows: When the speaker is slightly away from the microphone, the sound picked up by the microphone will be smaller than the normal distance of the speech. At this time, the auto gain will slightly be increased. Since it is a live sound reinforcement, this gain cannot be increased infinitely, because it will cause howling or other unfavorable situations, and only a certain amount of compensation can be made. When the speaker closes to the microphone, the sound will be louder than the normal distance of speech, especially when it is very close, and the auto gain will attenuate the sound signal to a certain extent. In this way, the auto gain can achieve a certain dynamic range of the sound.

4.2.2 Parameter information

On and Off: Turn on and off this function.

Soft Knee: Soft Knee allows the compression ratio to slowly increase as the input signal level increases, so that the process of sound from not being compressed to being compressed is not too noticeable. The larger the value, the larger the influence range of the inflection point.

Start Level: Desired output signal level. If the signal is above this threshold, the controller will compress the signal proportionally. Gain compensation is performed if it is below this threshold and within the maximum gain range. For example, set the start level to -15db. If the microphone level is -10db at this time, it will be compressed according to the compression ratio. If the microphone level is -17db and the maximum gain is 6db, then the gain compensation will be 2db.

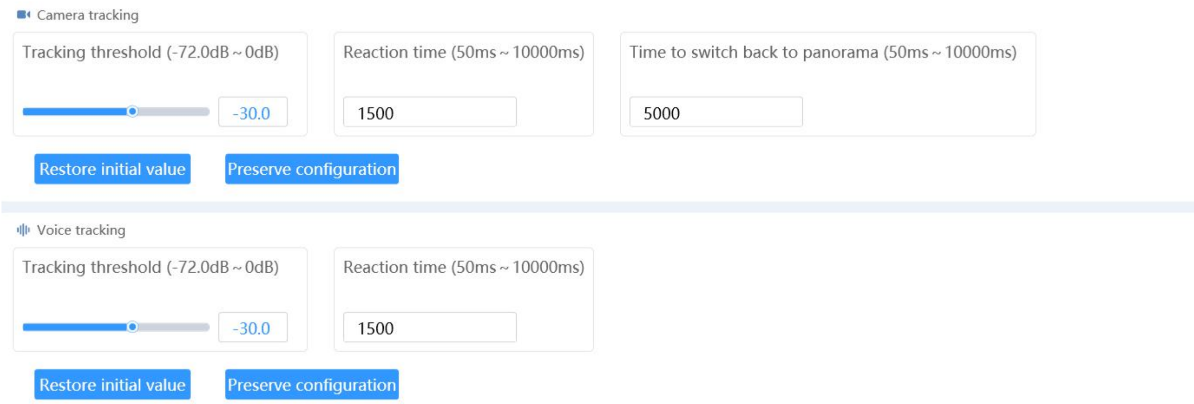
Compression Ratio: The ratio between the change of the input signal level above the threshold and the change of the output signal level.

Start Time: The response time of raising the input signal level to the start level, that is, when the input level is less than the start level, the signal needs to be boosted at this time, and this parameter is to control the speed of boosting.

Release Time: The response time of compressing the input signal level to the start level, that is, when the input level is higher than the start level, the signal needs to be compressed at this time. This parameter is to control the speed of compression.

Max Gain: When the signal level is lower than the target level, a certain gain needs to be applied to the signal. This parameter is to control the maximum gain on level.

4.3 Voice Activation



4.3.1 The voice activation is mainly to check whether someone is speaking using the microphone according to the voice amplitude, and this function can be extended to camera tracking and voice tracking.

4.3.2 Parameter information:

A.Camera tracking

Tracking Threshold: When it is detected that the microphone input signal is greater than or equal to the tracking threshold, the system will automatically enable the tracking function.

Response Time: The maximum interruptible time for a valid signal. If you use the microphone to speak, set the response time to 3 seconds, the signal is still considered to be continuously valid within 3 seconds of pause during the speech, and the signal is considered invalid if it exceeds 3 seconds.

Time to Back to Panorama: when no one is speaking, after the time to switching back to panorama, the camera will be switched back to panorama too.

B.Voice tracking

The parameters are the same as the camera tracking parameters.

4.4 Auto Feedback Control(AFC)

4.4.1 The sound signal collected by the microphone includes the sound signal amplified by the speaker. The signal is continuously superimposed and amplified in the sound feedback loop, and the positive feedback generates vibration and howling. AFC obtains the frequency of howling by analyzing the power spectrum of fixed-size audio frame by frame. After detecting the components, it uses a notch filter or a frequency shifter to change the amplitude and phase of the audio to achieve howling suppression.

4.4.2 The fixed point of the notch filter is mainly used to capture the inherent howling point of the sound field; the dynamic point is used for the howling point formed after the sound field changes. When the fixed point of the notch filter is full, the captured howling point will be set as the dynamic point. According to the actual sound field environment of the scene, you can set the corresponding fixed point/dynamic point of the notch filter.

4.4.3 If both the notch filter and the frequency shifter are set to "Enable", the frequency shift function will only be enabled when the fixed and dynamic points of the notch filter are full. The frequency shift function is always enabled when the notch filter is set to "off" and the frequency shifter is set to "on".

4.4.4 Parameter information

Notch On and Off: Turn this function on or off.

Reset Notch Point: Clear the detected fixed and dynamic points.

Fixed Point: The maximum number of fixed notch points of the notch filter. When the server is powered on again, it will resume capturing the fixed howling point before power off.

Dynamic Point: The maximum number of dynamic notch points of the notch filter. When the server is powered on again, it will capture the dynamic howling point and clear it.

Notch Amplitude: The maximum amplitude (db) of the dynamic notch points of the notch filter. If the value is set less than -9db, the notch amplitude of the notch point captured for the first time is -9db. If the notch point is captured again, the depth of the notch point will gradually increase. If the maximum notch amplitude is greater than -9db, the notch amplitude of the notch point captured for the first time is the set value.

Notch Q value: The notch filter Q value.

Frequency Shifter On and Off: Turn this function on or off. If set to "On", this function will be automatically enabled only when the fixed and dynamic notch points are full.

Frequency Shift Intensity: Overall audio movement (HZ), range: -10~10Hz.

4.5 Dante Interface

1. Support Dante channel input/output signals for a certain gain. The audio input by Dante interface will be mixed with the audio of the MIC and output from the LineOut interface; only the MIC audio will be mixed and output to the Dante channel.

2. Parameter information:

   Dante input gain: Gain compensation for the audio input from the Dante port, range: -72db ~ 12db.

   Dante output gain: Gain compensation for the audio output to the Dante port, range: -72db ~ 12db.

4.6 Others

1) STA indicator: When the Smart controller is not activated, the indicator flashes. After the Smart controller is correctly connected to the conference server and goes online, the indicator will stay on.

2) AFC switch: You can select whether the audio is AFC processed or not.

3) AFC indicator: When the notch filter or frequency shifter is turned on, the indicator is on.

Smart Audio Processor

Version 0.1