Digital Audio Processor Software

TS-YPCLR Software V1. 10

Description

The digital audio processor project aims to develop an audio processor hardware and software platform that combines software and hardware technologies to provide comprehensive audio processing capabilities and flexibility. The digital audio processor hardware has the ability to integrate with external devices to achieve more comprehensive audio processing and functional expansion. The digital audio processor software is an innovative product developed on the basis of hardware devices, with the characteristics of precise processing, real-time response, and versatility. The software integrates: phantom power supply, ducker, expander, equalizer, compressor, automatic mixing, feedback suppression, delay, crossover, limiter and other functions.

Features

1. [Phantom power] is used to power condenser microphones. Do not turn on line input or non-condenser microphones to prevent burning;
2. [Ducker] When the level of one channel exceeds the specified threshold, the level of the other channel will be attenuated.
3. [Expander]: The expander is the opposite of the compressor in principle. It can expand the dynamic range of the signal. The most basic difference between the two devices is that the compressor works on signals above the threshold, while the expander works on signals below the threshold. The expander can make a small signal larger.
4. [Parametric Equalizer] The main function of the equalizer is to correct the frequency range that is over-emphasized or missing, whether it is wide or narrow . In simple terms, the equalizer can change the timbre of the signal.
5. [Graphic equalizer]: It uses constant Q value technology, and each frequency point is equipped with a push-pull potentiometer. No matter whether a certain frequency is boosted or attenuated, the bandwidth of the filter remains unchanged. Commonly used professional graphic equalizers divide the 20Hz~20kHz signal into 10, 15, 27, and 31 segments for adjustment.
6. [Compressor] Reduces the dynamic range of signals above the user-set threshold, while the signal level below the threshold remains unchanged.
7. [Automatic Gain]Automatic gain control (AGC) is a special case of a compressor with a threshold set at a very low level , medium to slow attack time, long release time, and low ratio. Its purpose is to raise a signal of uncertain level to a target level while maintaining dynamics. Most automatic gain controls include some kind of silence detection to prevent loss of gain reduction during silence. Use automatic gain control to normalize the level of a CD player playing background music, foreground music, or music on hold to eliminate some changes in the level of a paging microphone.
8. [Automatic Mixing] In a conference room, if multiple microphones are turned on to the same gain level and only one person is speaking, the result may not be very clear. Other microphones will pick up room noise, reverberation, etc. When these signals are mixed with the normal microphone signal, the quality of the mixed audio output will be greatly reduced, and the entire sound reinforcement system is very prone to howling and cannot obtain sufficient sound transmission gain. In order to solve this problem, other microphones that are not in use temporarily need to be turned off. The automatic mixer can complete this shutdown process, and the response speed is much faster than manual operation.
9. [Feedback Suppression] The feedback suppressor is a device that automatically pulls the feedback point. When acoustic feedback occurs, it will immediately detect and calculate its frequency and attenuation, and execute the command to suppress the acoustic feedback according to the calculation results. This prevents the positive feedback of the microphone from being infinitely amplified.
10. [Echo Cancellation] Acoustic echo cancellation or AEC is a digital audio signal processing technology used in audio and video conference calls when the conversation takes place between the local conference room participants and one or more speakers at a certain distance. The AEC program increases the voice intelligibility of the remote speaker by eliminating the acoustic echo generated in the local room.

11. [Noise Removal] The noise suppression module can effectively remove non-human voices. It distinguishes human voices from non-human voices and treats non-human voices as noise. After an audio containing human voices and noise is processed by this module, theoretically, only the human voice remains.
12. [Matrix] The matrix has dual operation functions of routing and mixing. The horizontal direction represents the input channel, and the vertical direction represents the output channel. The default is one-to-one input and output. After setting up automatic mixing, echo cancellation, and noise cancellation, you still need to set up the matrix to obtain the correct signal routing relationship.
13. [Delay] The time interval from the signal input to the signal output of the processor. It is generally used to produce effects such as reverberation or echo. It can also be used to process auxiliary speakers in larger occasions.
14.[Crossover]Each output channel provides a high-pass and low-pass module, which consists of a high-pass filter and a low-pass filter.
15. [Limiter] A limiter is a circuit that can flatten the signal voltage amplitude within a limited range to ensure that the signal does not exceed the threshold level.
16.[USB Recording and Playback]Record and play audio through the USB interface, play stored audio files, and realize the transmission and processing of audio signals.
17. [GPIO Settings] The GPIO settings of the digital audio processor provide a set of general-purpose pins that can be used to connect external devices, control the functions of the audio processor, indicate the status, expand the connection to other devices, trigger specific events, and customize according to user needs, thereby achieving more flexible and customized audio processing functions.
1 8. [Serial port settings] supports docking with central control equipment and provides 4 85 serial ports and 232 serial ports to meet the communication needs between different devices .
18.[User Management] Add multiple users, modify/reset user passwords.
19.[Camera Tracking Setting] Set the camera connected to the digital audio processor hardware host.
20.[Data backup] Save the set parameters or import the previously set parameters.
2 1. [Central control command] is responsible for generating external central control commands.
22.[Firmware Upgrade] Upgrade DSP firmware.
23. [Limit Value Setting] Limit value setting allows administrators to set the maximum value of channel gain. Administrators are not restricted by limit values. On the contrary, ordinary users can only set gain within the set limit value range .
24.[Operation Log] Export the operation log generated by the system.
2 5. [Mode Switch] Supports switching between expert mode and normal mode.

Expert Mode: Selected by default, supports full-function display and control.

Normal mode: only displays input channels, output channels, USB playback, and USB recording. Channels only support gain adjustment and mute switch.