Digital Mixer TS-20DM-12

Digital mixer embedded software V3.156

Description

The product is a rack-mounted digital mixing console that uses DSP audio processing, stereo mixing, equalizer, compressor, delay timer and other technologies to amplify, mix, distribute, modify the sound quality and process the sound effects of multiple input signals, achieving high-quality audio effects and bringing users a more outstanding audio experience; it is mainly used in conference rooms, lecture halls, hotels, clubs and other scenarios.

Features

*The product is a 3U rack-mounted mixer with a 7-inch TFT capacitive multi-touch screen on the front panel with a resolution of 1024×600. The touch screen has the function of adjusting the volume of each input and output channel, equalizer, compressor, noise gate and other parameters; it can store 99 custom scene mode functions for quick call in different scenes .

*The touch screen has a system setting function, which can view the device name, DSP version, MCU version, software version, system version, CPU temperature, running time, USB connection status, customize the Bluetooth name, set the RS485 baud rate, set the software interface to display Chinese or English, select the firmware upgrade program, turn on the panel lock screen, restore factory settings and other functions.

*It has 12-channel balanced XLR microphone input channels and 2-channel line stereo input channels. The balanced XLR microphone input channels support independent control of 48V phantom function.

*The device is equipped with 1-channel USB recording and playback interface, which supports multiple audio formats, including MP3, M4A, M4R, MP2, AMR, AAC, WMA, as well as lossless music formats such as WAV and FLAC, and supports recording in lossless WAV format. Users can customize the USB playback and recording function settings through the device software interface , supporting the previous song, next song, loop playback, single loop, and random playback modes, and the playback and recording volume control uses simulated mixer fader control technology in the software interface, and the playback volume value can be set to -72dB~+12dB adjustable.

*It has 1-channel coaxial input interface and 1-channel digital optical input interface, which supports converting coaxial digital signals or digital optical signals into audio signal input.

*It has 1-channel RJ45 network interface and 1 WIFI/Bluetooth audio input interface, and supports connection with PC and Android tablet via wired network interface or wireless WiFi; users can control the device through Windows client software or Android tablet client software; the built-in Bluetooth module of the device also allows connection with mobile phones via Bluetooth to realize audio transmission.

*It has 1-channel RS485 interface control, through which users can send serial port commands for remote control.

*It has 8-channel AUX outputs (XLR interface), 1-channel Main L/R output (XLR interface) and 1-channel monitoring port, and you can independently select the output channel and monitoring port.

*It has 1-channel AES digital signal output XLR interface, which can be connected to professional digital audio signal processing equipment.

*It has 1-channel coaxial signal output, which converts the audio signal into a digital coaxial signal output.

*Each input channel has 12-band parametric equalizer, high-pass and low-pass filters, compressor, noise gate, polarity and delay.

*Each output channel has a 12-band parametric equalizer/31-band graphic equalizer, high-pass and low-pass filters, limiters, delays, and feedback suppressors.

*Built-in adaptive notch feedback suppressor and automatic mixer.

*There are 4 built-in DCA groups and 6 SUB groups. By combining multiple audio source channels together, it provides a convenient and efficient way to manage and adjust the channel volume. The panel is equipped with four DCA group knobs, so you can adjust the volume of the DCA group without switching to the DCA group interface.

*IN1-12, ST1, ST2 input channels have built-in compressor function, which compresses the dynamic range of the signal and is often used to compress the size of the output signal. It also has built-in noise gate function, which can effectively eliminate the white noise of the device.

*IN1-12 supports channel joint adjustment and channel copying functions, which can reduce repeated operations when you want to adjust multiple identical channel parameters.

*Each output channel has a delay function, providing a maximum delay adjustment of 2000ms, which is used to adjust the delay of each output signal so that each audio signal remains synchronized when reaching the listener's ears.

*Each output channel has a limiter function to limit the size of the output signal to prevent excessive signals from damaging the sound reinforcement equipment.

*Each output channel has feedback suppression function, with notch, frequency shift, dynamic/fixed (24 points) and other mode settings, which can effectively solve the problem of acoustic feedback.

*Built-in dual-engine professional effects (FX) processor, dual-channel output, with 17 effect modes such as delay, chorus, reverb, etc.: effect modes include REVERB, GATE_REVERB, DOUBLING, MONO_DELAY, STEREO_DELAY, PING_PONG_DELAY, ECHO, SLAP_DELAY, TAPE_DELAY, CHORUS, TREMOLO, WAHWAH, PHASER, SHIFTER, DISTORTION, EARLY REFLECTION, MODULATION_DELAY, etc., and can independently adjust parameters such as dry-wet ratio, delay, echo, timbre, attenuation, space, front reflection, etc.

*Supports 4 custom buttons for quick scene call/mute group control.

*Supports 6 groups of automatic mixing functions, you can select the channels to be mixed and the channels to be sent, gain sharing, threshold mixing and other mixing modes.

*It has C/S architecture client management software and Android tablet mobile software; the client software can run on Windows7/10/11, Apple macOS, Kylin desktop operating system (Zhaoxin version), Kylin desktop operating system (Feitian version), Tongxin UOS, Ubuntu and other Linux kernel operating systems ; the Android tablet mobile software can run on Android 7.0 and above systems or Hongmeng OS 4.0 and below systems.

Specification

Input channels	12-channel balanced XLR input channels, 2-channel line stereo input channels, 1-channel USB stereo input channel, 1-channel coaxial input channel, 1-channel digital optical input channel, 1-channel Bluetooth audio input channel
Output channel	8-channel AUX output (XLR interface), 1-channel Main L/R output (XLR interface), 1-channel monitor output, 1-channel AES digital signal output (XLR interface), 1-channel coaxial signal output
Phantom power	Support 12-channel independent controlled 48V phantom power supply
Communication interface	1×RJ45, 1×RS485, 1×WiFi/Bluetooth antenna interface
Signal generator	White noise, sine wave, pink noise as test signals
Maximum input level	≥+22dBu±0.2dBu
Microphone gain	75 dB (60 dB analog gain + 15 dB digital gain)
Audio sampling rate	48 kHz
Audio bit depth	32-bit
Total harmonic distortion (THD)	<0.003%,@+4 dBu
Frequency response	20 Hz–20 kHz (±0.5 dB)
Dynamic range	≥111dB
Signal-to-Noise ratio	≥110dB
WLAN	Support 2.4G, support WiFi 802.11 b/g/n
Bluetooth	Bluetooth 4.2 , support aptX & LDAC & AAC & SBC
Ethernet	Support Ethernet 100BASE-TX
Operating temperature	-10℃ ~ +45℃
Working environment humidity	20%~80% relative humidity, no condensation
Working power supply	100-240 VAC, 50/60 Hz
Maximum power consumption of the whole machine	30W
Dimensions (length x width x height)	484 × 2 12 × 132mm
Weight	5.3Kg