iTrans DX-BOX-44

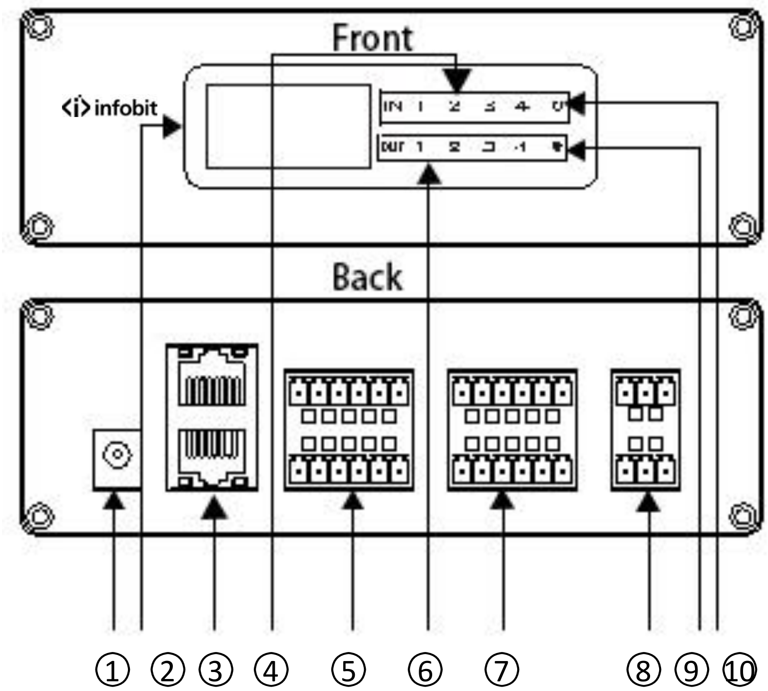
Datasheet

The iTrans DX-BOX-44 network audio interface is a product developed for Dante network audio transmission and routing. It can realize high quality and low delay of network audio transmission. It supports 4-way balanced input and 4-way balanced output analog channels, PoE+DC 12V dual power supply, and provides PC version control software for monitoring and operating the phantom power supply, mute and gain adjustment of interface machine equipment.

Features

1. This processor can realize interfacial audio signal processing.
2. This processor supports 4-way analog audio signal balanced input/ output, 4-way Dante digital audio signal input of 2 Dante devices, and the 4-way Dante digital audio signal output.
3. DSP processing functions include gain adjustment, 5 - stage parameter equalization, compressor, mixer, 15 - stage graphic equalization, limiter, setting, etc.
4. It can support network connection Windows PC control software.
5. POE mode power supply, can be used with 802.3af standard PoE switch.

Panel and Interface



① 12V Dc power interface.
② OLED Display screen, used to display IP address and other information.
③ Two-way Dante network interface, support PoE power supply, support network cascading.
④ Signal indicators, showing the status of 1-4 input signals.
⑤ Four-way balance signal input interface, if linear input, please connect "+" and "G".
⑥ Signal indicators, showing the status of 1-4 output signals.
⑦ Four-way balance signal output interface
⑧ Two-way AM100 interface, support far field pickup
⑨ The flashing light indicates that the system is in normal operation
⑩ When the indicator lights up, the system power is switched on.

Specification

Simulated maximum gain	-6~36dB
Frequency response	±0.2dB (20~20kHz)
Maximum input level	+10dBu, balanced
Maximum output level	+14dBu, balanced
Sampling rate	48K
Digitalizing bit	24bit
THD+N	< 0.05% @ 4dBu
Floor noise	-97dBu
Input impedance	20 kΩ

Output impedance	100Ω
Channel isolation	100dB @ 1kHz
Phantom power supply	48V
Input power	PoE / DC12V
Consumption	<9W

Software

Our designed PC version control software is the best tool for you to monitor and operate the digital audio processor, which can be used to edit and store scenes (such as meeting mode, artistic performance mode, concert mode, etc.) according to the acoustic characteristics of different functions. The system built-in lock screen function, effectively avoid the occurrence of wrong operation.

Technology

AUTOMIXER

1. Improve the transparency and clarity of speech;
2. The feedback, reverberation and comb filtering effects are significantly reduced.
3. Automatic adjustment, simplified settings, plug and play;
4. It can solve common problems such as insufficient gain before feedback and unclear speech.

Each input channel has a dual-band equalizer.

5. The adaptive noise threshold allows each input channel to distinguish between continuous background noise (such as air conditioning) and changing sound (such as voice), and constantly adjusts the channel activation threshold, so that the channel can only be activated when the voice volume is higher than the background noise;
6. Lock the last mic until the next mic is activated, ensuring that background ambient sounds are present (without the last mic lock, a long pause in the conversation shuts down all the microphones, as if the audio signal is missing);
7. Precisely control the priority of each microphone and lock down key speakers.

Automatic Echo Cancellation(AEC)

1. Using subband algorithm, less MIPS consumption.
2. The length of echo path can be set, the maximum echo off tail can be supported up to 512ms, suitable for all kinds of large, medium and small meeting rooms;
3. Using the stable Double Talk detection method, it is effective even in the environment of strong background noise and nonlinear distortion, and the residual echo will not increase during the simultaneous speech of both sides.
4. Strong robustness, can work in all possible applications and environments;
5. The embedded noise suppression algorithm can eliminate the additional noise in the noise environment.
6. The variable step size and post-processing algorithm greatly improve the rate of convergence and the echo rejection ratio (ERLE) of the nonlinear distortion of the terminal speaker.

Automatic feedback elimination (AFC)

1. Multi-point filtering and multi-subband frequency shifting keep the harmonic property of the original pitch period without causing sound distortion.
2. Through acoustic modeling of room feedback path, the acoustic feedback can be eliminated adaptively.
3. It can quickly track the feedback path changes and greatly enhance the ability to suppress the noise. The microphone transmission gain can be increased by 6-18db, greatly enhancing the microphone gain, suitable for various large, medium and small meeting rooms.

Automatic noise elimination (ANC)

1. It is a noise suppression technique to deal with noisy speech signals.
2. It decompositions the input signal into a series of frequency subbands, estimates the environmental noise and signal level in each subband, and then attenuates the subband signal according to the real-time SNR. The output signal is synthesized by smoothing and post-processing of these processed subband signals.
3. Because of the unique post-processing algorithm, the noise suppression algorithm can track the environmental noise changes quickly and accurately while maintaining good output sound quality. Noise suppression reaches -30db, speech is almost completely distortion free.