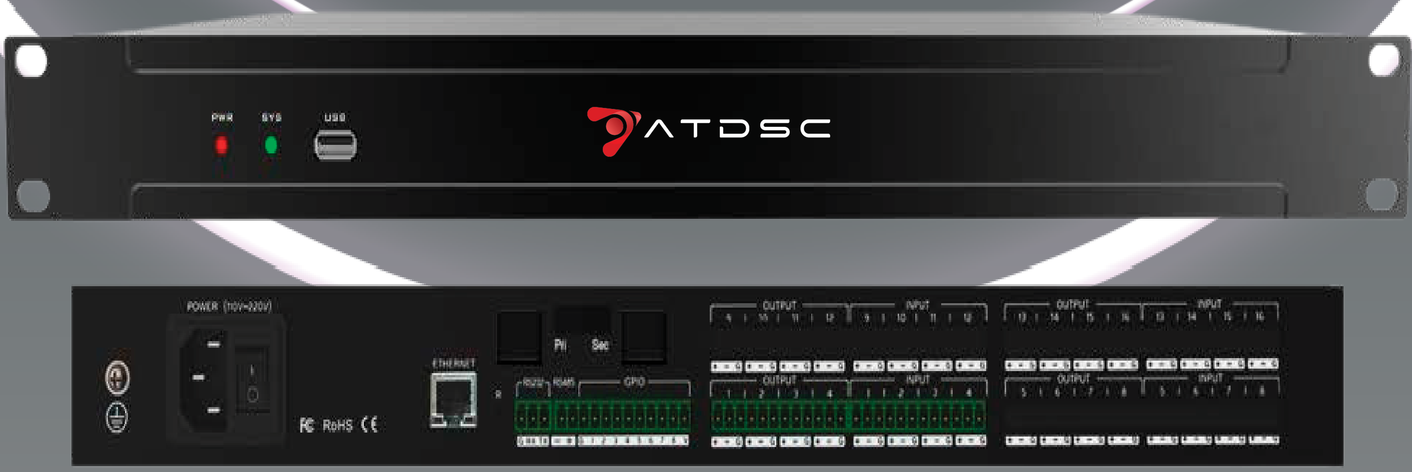
DIGITAL SIGNAL PROCESSOR



Sampling
48KHz

Phantom
Power 48V

Frequency Response
20-20K Hz, +0.2dB

Technical Specifications

ATDSC DIGITAL SIGNAL PROCESSOR

ATDSC Advanced DSP technology has new algorithms for Auto-Mixing and Feedback Cancellation to solve practical problems in targeted application scenarios. Most of the controls areoperated via the software. The appearance is made simpler, clearer and user friendly. Just click through mouse and smoothly execute the desired function. No more struggle through the field to adjust the complex large mixer to complete the function conversion. It greatly simplifies the operation.

Features

- USB Play & Recording and as sound card working as Audio in and Out when connected to PC / Laptop for Software based Video Conferencing.
- Support for mobile phone, tablet control and distributed cloud control.
- DSP audio processing, built-in Automatic Mixing Console.
- AFC Auto feedback elimination.
- AEC — Auto Echo cancellation.
- ANC: Auto Noise Cancellation module
- Inputs per channel: Preamplifier, Signal generator, Expander, Compressor, 5-band parametric equalization.
- Outputs per channel: 31-band graphic equalizer, delay, crossover, limiter. parametric equalization.
- Full function matrix mixing function.
- Scene / File presetting function.
- Password protected. Separate password for Admin and separates for users.
- IU all-aluminum chassis.

Software

- The PC version of the control software is the best tool for user to monitor and operate audio processor, allowing user to edit and store settings (e.g. Conference mode, Cultural Performance mode, Concert mode, etc.) for the acoustic characteristics required by the different functions used. Built-in lock screen function, effectively prevents misuse
- Comes with its own B/S architecture server, accessible via web browser, which enables channel control and scene selection and provides a direct link to download the PC client and platform components.

- Installed on tablets and mobile phones APP client, which is steady, simple designed, panoramic function menu, quick operation bar, makes it very convenient for all operations of the processor. Everything is designed to give a better user experience.

Core Algorithm

- Efficient and comprehensive built-in core algorithms is the soul of the processor for perfect sound quality.

AUTOMIXER

- Improves the transparency and clarity of speech.
- Significantly reduces feedback, reverberation and comb filtering effects.---
- Automatic adjustment, simplified set-up and plug-and-play.
- Solves common problems such as insufficient gain before feedback and unclear speech.
- Dual band equalizer for each input channe
- Adaptive noise threshold differentiates for each input channel between continuous background noise (e.g. air conditioning) and transformed sounds (e.g. speech)
- Precise control of the priority of each microphone, locking out important speakers.

AFC

- Multi-point filtering techniques and multi-sub-band frequency shifting which maintain the harmonic nature of the original fundamental cycle and do not cause sound distortion.
- Adaptive elimination of acoustic feedback through acoustic modelling of the room feedbac
- Quickly track feedback path changes and minimize enhanced whistle suppression.
- Microphone transfer gain enhancement can increase the transfer gain up to 6-18dB, making it suitable for use in a variety of large, medium, small conference rooms

AFC

- Sub band algorithms with less MIPS consumption.
- Configurable echo path lengths, supports a maximum echo tail-off of up to 512ms, suitable for use in a wide range of large, medium and small conference rooms

- Stable Double Talk detection method, is very effective even in environments with strong background noise and non-linear distortion, and it limits the increase in residual echoes during simultaneous speech.
- Highly robust and works well in all possible applications and environments.
- Embedded noise suppression algorithms eliminate additional noise in noisy environments.
- Variable step size and post-processing algorithms that greatly increase the speed of convergence and the echo rejection ratio (ERLE) during non-linear distortion of the end speaker.

ANC

- The noise suppression processes noisy speech signals. It decomposes the input signal into a series of frequency sub-bands, estimates the ambient noise and signal level in each sub-band, then attenuates the sub-band signal according to the real-time signal-to-noise ratio, and the output signal is synthesized from these processed sub-band signals after smoothing and post-processing. As it has unique post-processing algorithm the noise suppression algorithm quickly and accurately tracks changes in ambient noise maintaining very good output sound quality. Noise rejection of -30dB is achieved and speech is almost completely distortion-free.