Tiger D1616N

S H E N Z H E N S - T R A C K S C I E N C E & T E C H N O L O G Y C O . , L T D

TEL：0755-29983191

Service hotline：400-900-2726

Website：www.s-track.com.cn

Add:14th floor, block A, No.196 Tangtou road, Shiyan Street,Baoan district,

Shenzhen,China

Product Introduction

The TIGER D1616N audio processor is a freely designed audio processing and control system. It adopts advanced DSP processing technology, has a new automatic mixing, Feedback elimination, etc, targeted to solve various practical problems in the application scene.The built-in Dante module provides a high bandwidth, low latency, high compatibility and low cost solution for network audio transmission. Because most of the controls are handled by software, It looks much simpler. The operators only need to click the mouse gently, no longer like before, that adjust the complex large mixer in the field to complete the function conversion. The TIGER D1616N simplify the operation difficulty greatly.

Main Features

- USB Background music playback and recording function
- Support mobile phone, tablet control and distributed cloud control
- DSP audio processing, built-in automatic mixer, optional feedback elimination, echo elimination, noise elimination
- Input per channel: Front stage amplifier, signal generator, expander, compressor, 5 stage parameter equalization
- Output per channel：31 Section diagram equalizer, delay device, frequency divider, limiter
- Full function matrix remix
- Built-in Dante audio module
- Built-in automatic camera tracking function
- Support for scenario presets；
- Automatic memory protection when power off
- 1U Whole aluminum chassis

Technical Parameters	
Model No.	Tiger D1616N
DSP Processing	Ti 456MHz FLOPS DSP
Number of analog channels	16 Input +16 Output
Dante Network channel No.	16 Input +16 Output
Core Algorithm	Automatic mixing, feedback elimination, echo elimination, noise elimination
GPIO	8 (Including input and output)
RS232/RS485	1
RJ45 control interface	1
USB Port	1
RJ11 phone interface	0
DANTE network	Main network port + backup network port
DANTE network delay	<1ms
Simulated maximum gain	51dB
Digitalizing bit	24bit
Sampling Rate	48k
Frequency Response (20~20KHz)	±0.2dB
Analog-to-digital dynamic range (A-weighted)	114dB
Digital-to-analog dynamic range (A-weighted)	120dB
Input to output dynamic range	108dB
Total harmonic distortion + noise	＜0.003% @1KHz ,4dBu
Noise floor (A- weighted)	-90dBu
Delay storage	2s
Analog input to output system delay	3ms

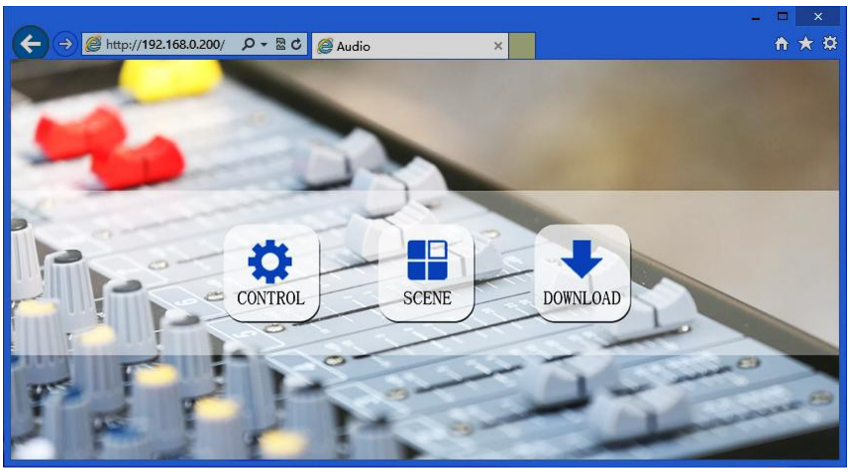
Input impedance (balance -type)	20KΩ
Output impedance (balance -type)	100Ω
Maximum Input Level	+18dBu
Maximum Output Level	+18dBu
Equivalent input noise EIN (20-20kHz , A-weighted)	≤-131dBU
Phantom power (per input)	48V
Input common mode rejection , 60Hz	70dB
Channel Isolation , 1kHz	104dB
Dimensions (W*D*H)	482*258*45(mm)
Weight	3Kg
Power Consumption	<40W
Operating Temperature	-10 to 50℃
Working Power Supply	AC110V/220V,50Hz/60Hz

The Software Experience

- 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, It can effectively avoid the occurrence of wrong operation.

Windows Platform client software

- The digital audio processor comes with B/S architecture server, which is accessed through web browser, not only realizing channel control and scene selection, but also directly providing download links of PC client and platform components.



Browser interface

- Installed on tablet and mobile phone APP client, sedate and simple design style, panoramic function menu, with fast operation bar, It can be very convenient for the processor to carry out various operations. Everything just to give you a better user experience.

IOS Platform control software

Core algorithm

Efficient and comprehensive algorithm is the basis of perfect sound quality, but also the crystallization of engineers' experience and wisdom. The built-in core algorithm is the soul of the processor.

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.
5. Each input channel has a dual-band equalizer.
6. 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;
7. 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);
8. Precisely control the priority of each microphone and lock down key speakers.

Automatic Echo Cancellation(AEC)

1. Using subband algorithm, it has 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.