INTELLIGENT MIXER IS COMING!

SK46

Adaptive Sound Field Controller

New Intelligent On-site Acoustic Test Processing Tool

Strong performance processing capacity	Sound quality similar to analog audio	Extremely simple test operation	FPGA powerful DSP processing	High frequency calculation of ARM core	Standardized network protocol connection

R&D BACKGROUND

Sound field control is a new technology developed in recent decades. The traditional method of sound field control has always been a difficult problem in solving low-frequency noise control. When a person or multiple speaker are in the moving state, it is impossible to get the relative position of a person and multiple speaker, and it is difficult to find the best dynamic balance point of sound effect. SK46 adaptive sound field controller brings hope to solve this problem.

The SK46 adaptive sound field controller is based on the world-leading audio research and development platform. This series of products is based on SineCore technology and adopts the world's first high-end real-time FIR acoustic phase calibrator using FPGA algorithm. It gets rid of the restriction of delay on digital audio algorithms. The most realistic and natural sound is reproduced through the 3D standards of sampling accuracy, sampling frequency and phase accuracy.

TECHNOLOGICAL INNOVATION

FPGA high-end real-time FIR acoustic phase calibration technology:

Significantly improve the clarity of the mixing, and accurately achieve a flat sound frequency response

Everything we do in the process of recording, mixing, mastering and sound reinforcement is based on the sound heard by the human ear. Therefore, whether it is the sound we hear in the control room or the sound captured by the microphone during live recording, it plays a decisive role for audio workers. If a studio and sound reinforcement venue cannot provide a standard acoustic space, even experienced audio engineers or creators cannot make accurate judgments in the creative process. SK46 adaptive sound field controller is your wise choice to get rid of all the acoustic problems in the control room. It can optimize your acoustic environment, calibrate the frequency response of the audio system and acoustic space, perfectly show the sound details, and create an ideal acoustic space for you, allowing you to create as much as you want in the studio.

Speakers with poor craftsmanship

Speakers with good craftsmanship

High-end FIR-calibrated Sonic Crystal speakers

TECHNOLOGICAL INNOVATION

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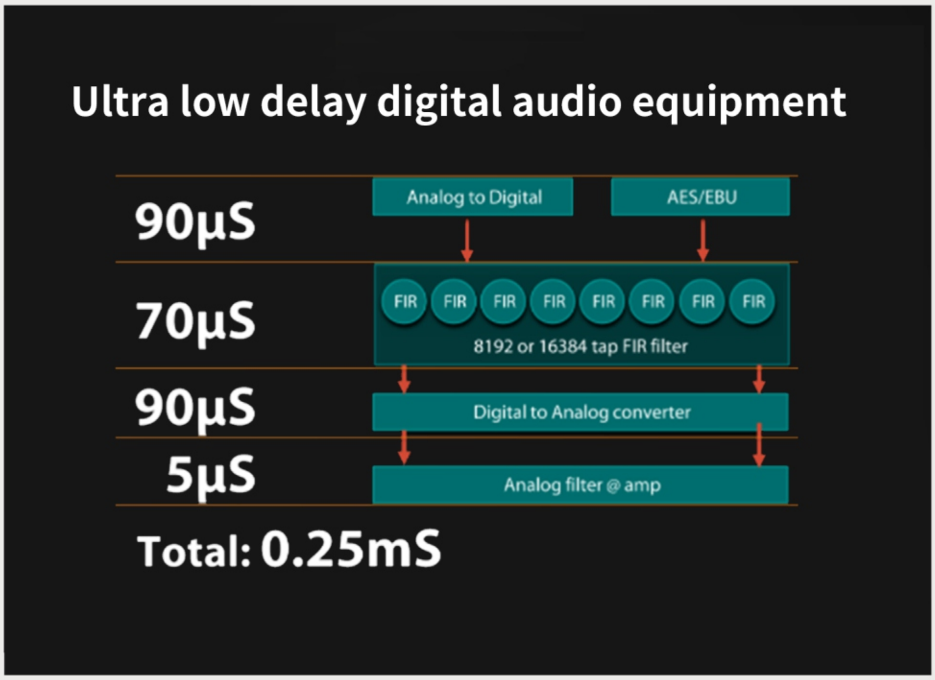
Significantly improve the clarity of the mixing, and accurately achieve a flat sound frequency response

The traditional DSP processing system will additionally produce more phase problems during the processing, and the secret of the SK46 core technology is hidden in the SineCore audio processing platform. It uses an FPGA-based high-end real-time FIR acoustic phase calibration system, which breaks the barrier between digital audio algorithm technology and delay, and perfectly reproduces the essence of sound through the 3D standards of sampling accuracy, sampling frequency and phase accuracy.

SK46 optimizes your acoustic environment by enhancing the clarity of sound images and restoring the transparency of sound. The system provides a variety of interfaces, is compact in design and simple in use, and can meet the needs of sound field measurement and calibration in small and medium auditory rooms, studios, theaters, halls and large live performances. Achieve more transparent, high-resolution high-frequency performance and more powerful bass restore.

FIR filters are characterized by linear phase, which adapts to the unique requirement of audio - that is, changing the frequency domain without phase change. However, due to the characteristics of DSP serial, FIR filters of order 10,000 or more cannot adapt to the real-time performance of audio requirements because of the long delay time. The new delay brings new phase problems, so high-order FIR and waveforms have performance problems due to DSP. Even because of the complexity of processing algorithm coefficients, some manufacturers even use the CPU of a PC to calculate coefficients and process FIR, which can cause greater system latency.



TECHNOLOGICAL INNOVATION

Traditional solutions to acoustic space problems are quite complex. Systems engineers need to use different measurement and processing tools to solve different problems. After spending a lot of time debugging, only one defective sound system can be handed over to the tuner, who can only solve the problems in the system repeatedly according to the defective system. There is no time to balance music at all.

The unique algorithm we designed only needs microphone measurements at one point in the room or at many points in a large space. Each measurement point takes only three seconds. It has a great tolerance for noise in space. Then it takes ten seconds to discover the phase problem in the system and send the processing parameters of the problem to the FPGA automatically. No human intervention is required in this series of processes. Just put the microphone on and press the measurement key. After 10 seconds, the tuner gets a standard acoustic space where there is no underbass caused by the cancellation of multiple subwoofers, no low-frequency resonance, and no feedback screaming. Because our intelligent algorithm already knows the sound field. The tuner can even design the overall style of the music audio through our frequency curve design.

TECHNOLOGICAL INNOVATION

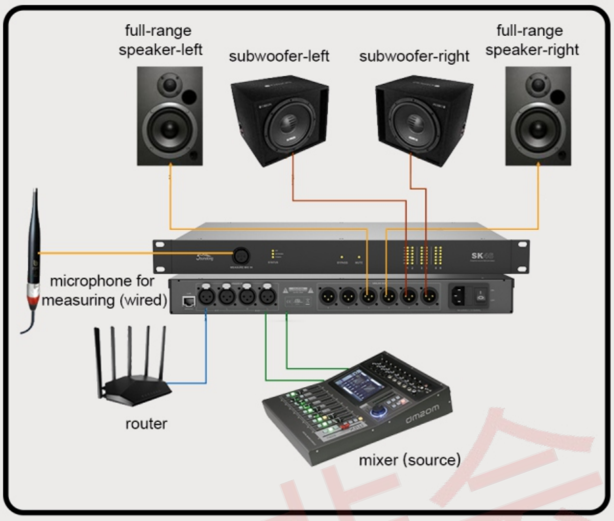
>Connect the stereo left and right speakers to Analog out3 and Analog out4 respectively

>Heavy bass, the first one connects to Analog out1, and if you need a second one, the second one connects to Analog out2

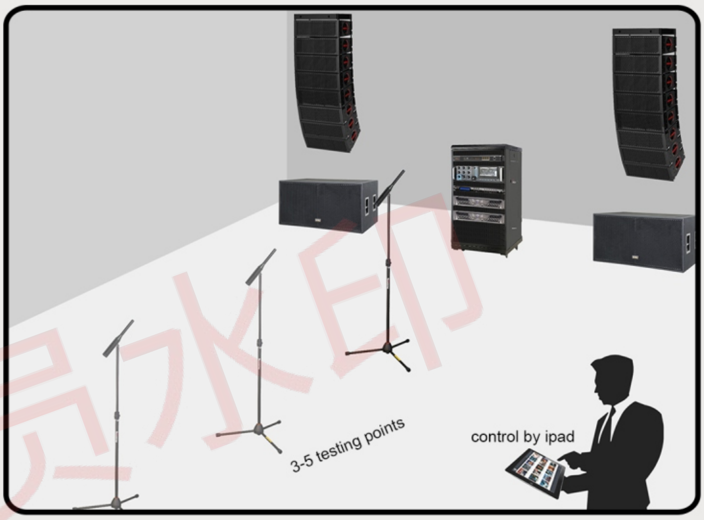
>The network cable connects to the router, and the computer program controls SK46 over the network

>Front panel XLR interface connects measuring microphone

>Source access to Analog in



Connection and calibration



Sound Field Calibration Field Simulation Diagram

Comparison of debugging times for various scenarios

Scenario	Traditional tuner	Intelligent tuner (SK46)
Theme park	1-2days	30minutes
Television studio	2-3days	30minutes
Gymnasium	2-3days	30minutes
Grand theatre	1-2days	20minutes
Live house	1days	15minutes
Large conference room	1days	15minutes
Large event site	1days	10minutes
Cinema	1days	10minutes
Multifunction room	1days	10minutes

APPLICATION

It is suitable for use in theme parks, TV studios, sports venues, grand theatre LIVE HOUSE, large conference rooms, large event rooms, cinemas, multi-purpose halls and other places.

Theme park

Large conference room

Multi-purpose hall

TV studio

LIve house

Grand theater

SPECIFICATION

SK46 Parameter

Analog input	4x XLR (L/R), +24dBu max
Analog output	6xXLR (L/R; subwoofer+bass+mid+treble), +24dBu max
Digital input	1xAES/EBU@75Ohms
Word clock input	1xInput@75Ohms 3Vpp on BNC32-192kHz
Word clock output	1xOutput@75Ohms 3Vpp on BNC32-192kHz
D/A converter	Dynamic range: 120cb THD + N: -107dB
A/D converter	Dynamic range: 120cb THD + N: -110dB
Sampling rate	44.1,48,88.2,96,176.4,192(kHz)
Electrical requirement	AC input: ~95-245V; power consumption: 20W max

MICROPHONE P30 Parameter

Type	Electret condenser microphone
Mic diameter	1/4"（7mm）
Pickup mode	Omni
Sensitivity	30mv/pa
Frequency response	20hz-20khz
Max. SPL	130db
Balanced noise level	＜30dba
Output impedance	＜600ohm
Power supply	48V phantom power