AI Adaptive Sound Pickup and Amplification System

AI adaptive sound pickup and amplification system for recording and broadcasting classrooms

CR1

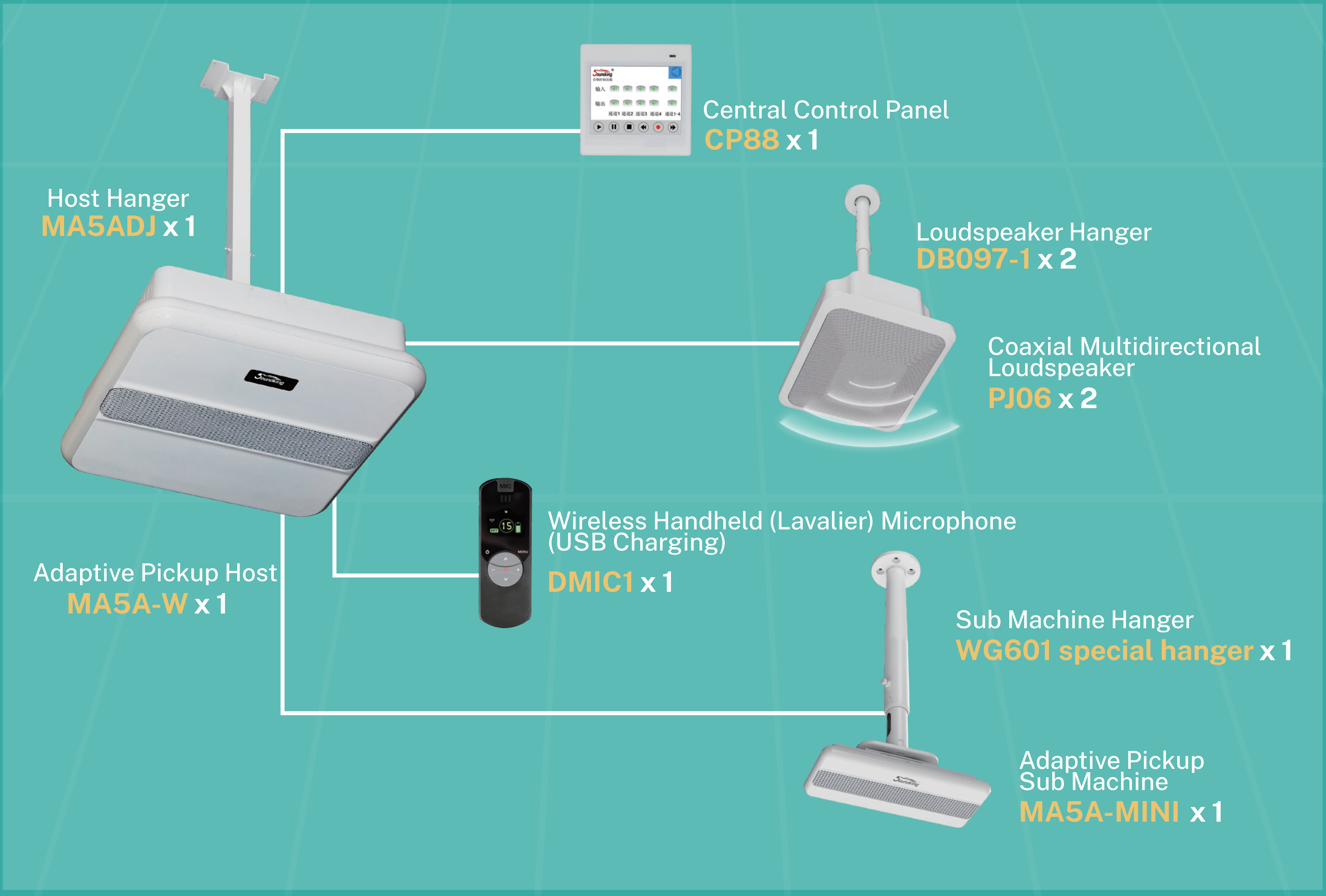
Introduction

This system is a set of high-tech, multi-functional intelligent control sound system specially developed by Soundking for schools and various training institutions. It has obtained a number of national patents, patent number: Invention 202010942665. X (an adaptive pickup communication and intelligence system); The utility model 202020562244. X (an adaptive pickup all-in-one machine) is composed of an intelligent pickup all-in-one machine, a coaxial loudspeaker, a control panel, etc. There is no need to wear a microphone in the podium area (array microphone pickup), and a wireless handheld microphone can be used for any indoor movement. When recording and broadcasting, wireless handheld microphone can be played to achieve clear and bright recording and broadcasting effect, between the two micsWith one click switching, it is convenient to use and meets the needs of any area in the classroom for sound amplification, achieving clear and natural voice amplification effects. This system is purely a protective deity for teachers' throats, created as a green and healthy project for school teachers.

Innovation

- Mixing console, audio processor, digital amplifier, feedback suppressor, switch and array microphone are all in one, highly integrated.
- The built-in processor can equalize and filter the microphone signal. There are signal processing functions such as algorithm AFC, AEC and ANC in sound feedback suppression.
- The pickup microphone has high sensitivity and can be adjusted by software. The pickup range can be adjusted within 2-6m as required, with high definition.
- It can be controlled through the central control panel, and can be switched and adjusted for easy use.
- Dual microphone integration, which can pick up sound through array microphones or enable wireless handheld microphone pickup with one click, meeting the needs of various places and people.
- The wireless handheld microphone adopts dedicated RF technology and digital encoding methods, making it have low noise, no crosstalk, stable communication, and long transmission distance.
- Equipped with on-site sound reinforcement and recording functions.

Topology Map



MA5A-W

Adaptive Pickup Host

Introduction

MA5A-W adaptive pickup host is an intelligent sound amplification system; It is a highly integrated intelligent all-in-one machine integrating mixing console, digital amplifier, audio processor, feedback suppressor, switch and array microphone. It is equipped with handheld microphone, which can be switched to handheld microphone with one button, meeting the different needs of smart classrooms, various conferences and other places for high-quality sound systems.

Features

- The machine provides 4 analog inputs (band independent phantom power supply, including 2 built-in and 2 external), and the interface adopts 12 core Phoenix plug.
- Built in 2 x 150W (4Ω) digital amplifier.
- The input end has built-in preamplifier, signal generator, expander, compressor, 5-segment parametric equalization, automatic gain, and automatic mixing.
- 31 segment equalizer, delay device, crossover, high-low pass filter and limiter are built in the output end.
- Support AFC adaptive feedback elimination.
- Communicate with PC through network port, debug and monitor instrument.
- It supports RS232 communication and intelligence system interface and can be connection with other central control instrument.
- Supports USB recording and playback function.
- Supports dual microphones, one click switch to handheld microphone.

Parameters

Gain Difference Between Channels	Within ± 0.5dB (20Hz-20kHz)	Output Interface	2-way amplifier output
Amplitude Frequency Response	Within ±1dB (20 Hz-20kHz)	Number of Storable Scenes	99 of them
Signal to Noise Ratio (A-weighted)	≥85dB (1000Hz)	Sampling Rate	24bit/48kHz
Crosstalk	≥80dB (1000Hz)	Net Weight	5.5Kg
Total Harmonic Distortion Plus Noise (weighted by A)	≤0.35% (20Hz-20kHz)	Input Interface	2x12 pin Phoenix plugs, 1x3-core Phoenix plug One network port, one USB 2.0 interface
Phase Difference Between Channels	≤0.5° (1000Hz)	Machine Dimension	420x420x115mm (W x H x D)
Dynamic Range (A-weighted)	≥85dB (1000Hz)	Input Voltage	AC 176-264V 50/60Hz

MA5A-MINI

Adaptive Pickup Sub Machine

Introduction

MA5A-MINI adaptive microphone is a high sensitivity polar microphone, which can be used with the MA5A series adaptive polar all-in-one machine. Meet the different needs of high-quality sound systems in smart classrooms and various conference venues.

Parameters

Power Supply	48V
Pick Up Sound	3-5 meters long distance
Interface	3-core Phoenix plug
Product Dimension (W x H x D)	190 x 28 x 80mm
Package Dimension (W x H x D)	520 x 330 x 270mm
Gross Weight	5.9kg
Net Weight	4.4kg
Number of Items Per Box	20PCS

PJ06

Coaxial Multidirectional Loudspeaker

Introduction

PJ06 is a coaxial multidirectional loudspeaker. The product is completely independent and innovative, and Obtained national patent, patent number: ZL202121067811.5. The sound is clear, bright, thick and plump. It can be installed in flying mode on the exposed top, or in fixed mode with hidden locks on the semi concealed top. It can be perfectly used in various indoor sound reinforcement occasions.

The PJ06 loudspeaker is composed of five 1-inch neodymium magnetic dome tweeter and one 6.5 inch neodymium magnetic MF/LF driver, forming a group with great headroom. The HF coaxial arc array is arranged above the MF/LF, with accurate positioning of sound image and uniform coverage of the sound field.

Application: classrooms, conference rooms, public broadcasting, multi-function halls, banquet halls, and other indoor sound reinforcement occasions.

Parameters

Frequency Response	60Hz~20kHz
Coverage	100° x 70°
Chassis	1 x 6.5"
Tweeter	5 x 1"
Sensitivity (1W@1m)	89dB
Rated Impedance	8Ω
Rated Power	120W
Max SPL (@ 1m)	116dB
Dimension of Cabinet (W x H x D)	330 x 391 x 220mm
Weight	5kg

SC10

Wireless Handheld (Lavalier) Microphone (USB Charging)

Features

- SC10 is a multifunctional wireless handheld (lavalier) microphone that can be used in multimedia classrooms, speech conferences, and more.
- The product works in the UHF band, with a sampling rate of 24K, meeting the requirements of high-quality voice transmission.
- Specialized RF technology and digital encoding methods enable it to have advantages such as low noise, no crosstalk, stable communication, and long transmission distance.
- The OLED screen is used to display all kinds of working message, including signal strength, power, volume, working channel and frequency matching mode.
- This product has four frequency matching modes: UHF, U_L, IR and PPT, which can avoid the problem of incorrect connection between different spaces.
- Type-C charging method.

Parameters

Sampling Accuracy	16bit
Sampling Rate	24kHz
Frequency Response	100Hz-10kHz@±3dB
Distortion Degree	0.4%>@94dBSPL,1kHz
Signal to Noise Ratio	83dB
System Delay	15ms
Full Range Input	220mVrms
Full Range Output	882mVrms
Input/Output Gain Ratio	12.2dB

CP88

Central Control Panel

Introduction

CP88 is a panel type communication and intelligence system processor, which cooperates with automation instrument to realize power switch, volume adjustment and other functions.

Meet the different needs of high-quality sound systems in smart classrooms, various conferences, and other places.

Features

- 3.5-inch (resistive touch) 5-26V voltage.
- Built in Cortex-M3+high-speed FPGA, dual core processor.
- Equipped with RS485 communication interface, it can connection with other central control instrument.
- Volume adjustment can be realized with the adaptive all-in-one machine.

Parameters

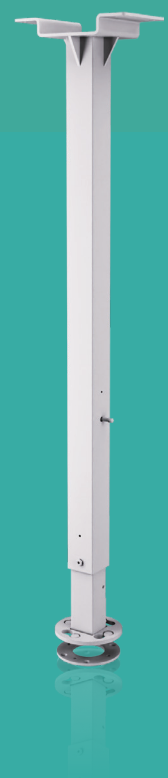
Dimension	3.5 inches
Resolving Power	320 x 240
Power Waste	Brightest backlight: 200mA@5V ; Turn off backlight: 96mA@5V
Communication Interface	RS485
Voltage	5-26V (error+0.2V)
Product Dimension (W x H x D)	86 x 27.5 x 86mm
Package Dimension (W x H x D)	97 x 33 x 89mm
Net Weight	0.22kg
Number of Items Per Box	1PCS

MA5ADJ

Host Machine Hanger

Parameters

Adjust the Height Range	888~1388mm
Adjust the Gear	5 levels
Adjust the Distance for Each Gear	100mm
Load Bearing Range	≤50kg
Texture of Material	iron
Surface Treatment	Spray painted white matte
Net Weight	4kg
Dimension of Cabinet (W x H x D)	895x105x185mm



DB097-1

Loudspeaker Hanger

Parameters

Adjust the Height Range	200~300mm
Adjust the Gear	3 levels
Adjust the Distance for Each Gear	100mm
Load Bearing Range	≤25kg
Supervisor Material	Φ 41x1 aluminum profile
Expansion Pipe Material	Φ 35x1 aluminum profile
Decorative Cover Material	PP
Guiding Sleeve Material	PP
Surface Treatment	Spray painted white matte
Net Weight	0.39kg
Dimension of Cabinet (W x H x D)	385 x 460 x 260mm (12PCS)

AI adaptive sound pickup and amplification system for standard classrooms

CR2

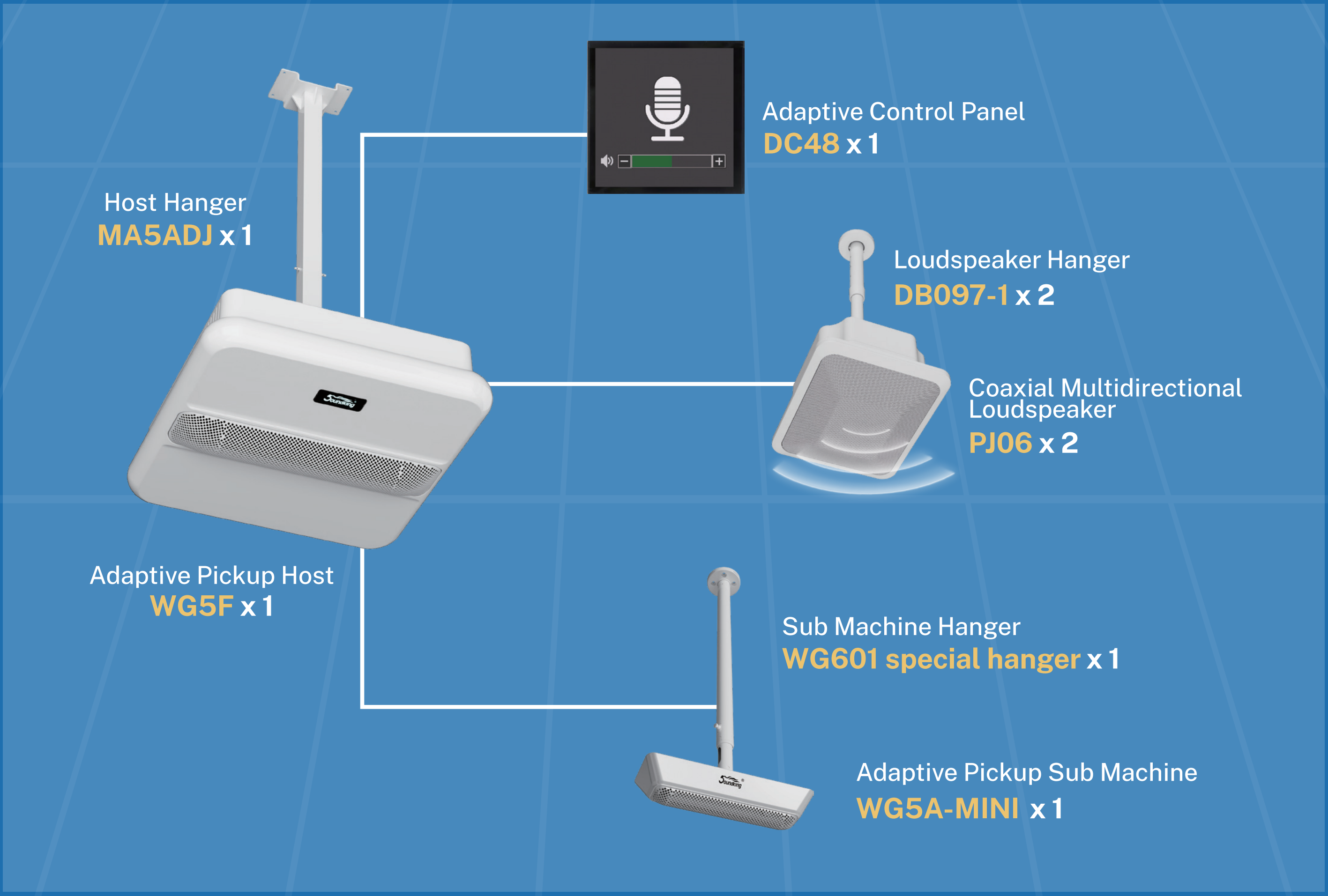
Introduction

This system is a set of high-tech, multi-functional intelligent control sound system specially developed by Soundking for schools and various training institutions. The system consists of intelligent pickup machine, coaxial loudspeaker, control panel, etc. Teachers do not need to take a microphone when giving lectures. They can walk around the room freely to achieve a clear and natural sound reinforcement effect. This system is purely a protective deity for teachers' throats, created as a green and healthy project for school teachers.

Innovation

- 3-5 meters ultra long pickup distance, freeing hands and capturing more details of human voice.
- Built in advanced DSP algorithm, high speech clarity, high voice restoration degree.
- Intelligent noise reduction technology, original ultra-low latency AI noise reduction model, effectively suppresses various transient noises such as typing and keyboard.
- One click debugging, automatic debugging can be completed in 3 minutes, freeing up debugging personnel.
- Four-channel balanced analog input (with independent phantom power, where channels 1 and 2 are connected to the main and sub-microphone inputs, and channels 3 and 4 are connected to external line inputs), featuring a 12-pin phoenix connector.
- It can be controlled through the central control panel, and can be switched and adjusted for easy use.

Topology Map



WG5F

Adaptive Pickup Host

Introduction

WG5F adaptive pickup host is an intelligent sound amplification system; It is an intelligent all-in-one machine integrating mixing console, digital amplifier, audio processor, feedback suppressor and array microphone. It is highly integrated, meeting the different needs of intelligent classrooms, conferences and other places for high-quality sound systems.

Features

- This device provides 4 analog inputs (with independent phantom power, where channels 1 and 2 are connected to the main unit and sub-unit microphone inputs, and channels 3 and 4 are connected to external line inputs). The interface uses a 12-pin Phoenix plug.
- Built-in 2 x 300W (8Ω) digital power amplifier.
- Intelligent noise reduction technology, featuring an original ultra-low latency AI noise reduction model, effectively suppresses various transient noises such as tapping and keyboard clacking.
- Support AFC (Adaptive Feedback Cancellation).
- With a super long pickup distance of 3-5 meters, it frees up both hands and captures more details of human voices.
- Supports 485 interface, can be connected to central control equipment for control.
- One-click debugging, automatic debugging completed in 3 minutes, freeing up debugging personnel.
- With ducking function, it suppresses sounds from other channels.

Parameters

Sampling Rate	32kHz	Audio Processing Startup Threshold	-45dB
Frequency Response	25Hz-15kHz	Digital Processing Delay	8ms
Gain Adjustment Range	0~40dB d-time	Amplifier Power	2 x 300W
Adaptive Noise Reduction	0-12dB	Input Voltage	AC 220-240V 50/60Hz
Overall Signal-to-noise Ratio	>105dB	Size	420 x 420 x 140mm

WG5A-MINI

Adaptive Pickup Sub Machine

Introduction

WG5A-MINI adaptive microphone is a high sensitivity polar microphone, which can be used with the WG5F series adaptive polar all-in-one machine. It boasts a long-range pickup capability of 3-5 meters, ensuring a wider range of sound source collection. Through the intelligent adjustment function of the adaptive pickup integrated main unit, it delivers ultra-high speech clarity.

Parameters

Microphone type	13-element array microphone
Directional	Corridor Coverage
Maximum withstandable sound pressure level	112dB
Output impedance	200Ω
Power supply	48V phantom power supply
Interface	3-pin Phoenix plug
Net weight	0.68kg
Product dimensions （W x H x D)	305 x 97 x 71 mm

PJ06

Coaxial Multidirectional Loudspeaker

Introduction

PJ06 is a coaxial multidirectional loudspeaker. The product is completely independent and innovative, and Obtained national patent, patent number: ZL202121067811.5. The sound is clear, bright, thick and plump. It can be installed in flying mode on the exposed top, or in fixed mode with hidden locks on the semi concealed top. It can be perfectly used in various indoor sound reinforcement occasions.

The PJ06 loudspeaker is composed of five 1-inch neodymium magnetic dome tweeter and one 6.5 inch neodymium magnetic MF/LF driver, forming a group with great headroom. The HF coaxial arc array is arranged above the MF/LF, with accurate positioning of sound image and uniform coverage of the sound field.

Application: classrooms, conference rooms, public broadcasting, multi-function halls, banquet halls, and other indoor sound reinforcement occasions.

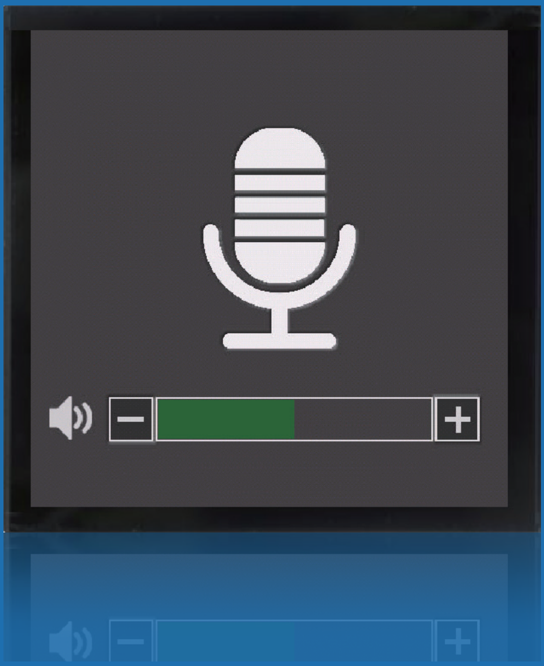
5 x 1" neodymium magnetic dome tweeter
1 x 6.5" neodymium magnetic MF/LF driver

Parameters

Frequency Response	60Hz~20kHz
Coverage	100° x 70°
Chassis	1 x 6.5"
Tweeter	5 x 1"
Sensitivity (1W@1m)	89dB
Rated Impedance	8Ω
Rated Power	120W
Max SPL (@ 1m)	116dB
Dimension of Cabinet (W x H x D)	330 x 391 x 220mm
Weight	5kg

DC48

Adaptive Control Panel



Introduction

The DC48 adaptive control panel is capacitive touch type and equipped with an RS485 communication interface. It can be connected to the device through this interface to adjust device parameters and volume.

Parameters

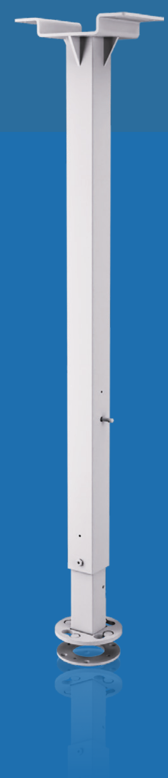
Screen size	4.0"
Resolution	480 x 480
Power consumption	Backlight at its dimmest with no speaker: 0.5W; Backlight at its brightest with no speaker: 1.3W
Communication interface	RS485
Display type	IPS LCD screen
Touc-hscreen type	Capacitive Touch Screen
Working power supply	4.5-30V
Dimensions（W x H x D）	86 x 86 x 27.5mm

MA5ADJ

Host Machine Hanger

Parameters

Adjust the Height Range	888~1388mm
Adjust the Gear	5 levels
Adjust the Distance for Each Gear	100mm
Load Bearing Range	≤50kg
Texture of Material	iron
Surface Treatment	Spray painted white matte
Net Weight	4kg
Dimension of Cabinet (W x H x D)	895x105x185mm



DB097-1

Loudspeaker Hanger

Parameters

Adjust the Height Range	200~300mm
Adjust the Gear	3 levels
Adjust the Distance for Each Gear	100mm
Load Bearing Range	≤25kg
Supervisor Material	Φ 41x1 aluminum profile
Expansion Pipe Material	Φ 35x1 aluminum profile
Decorative Cover Material	PP
Guiding Sleeve Material	PP
Surface Treatment	Spray painted white matte
Net Weight	0.39kg
Dimension of Cabinet (W x H x D)	385 x 460 x 260mm (12PCS)

AI adaptive sound pickup and amplification system (Economical)

CR3

Introduction

This system is a set of high-tech, multi-functional intelligent control sound system specially developed by Soundking for the school, and has won many national patents. The system is composed of intelligent pickup machine, loudspeaker, etc. Teachers do not need to take a microphone when they are lecturing. They can walk around the podium area freely to achieve clear and natural sound reinforcement effect. The sound reinforcement effect is excellent and the voice is clear. This system is purely the protector of teachers' throat and a green health project for school teachers.

Innovation

- The host adopts flying mode, with adjustable height, and faster installation and commissioning.
- 5A algorithm: AFC, AEC, ARC, ANS, AGC beamforming, sound source localization and blind source separation technology, which meets the local sound amplification requirements for consistent sound and image.
- The diameter is 5m, the pickup angle is adjustable, and it covers the platform area for teachers to pick up.
- Free your hands, protect your teacher's voice, and create a clear and healthy listening environment.

Topology Map

WG601

Adaptive Pickup Host

Features

- The end-to-end processing delay is 10 ms, which can ensure no echo in the whole field.
- The energy difference between the sound inside and outside the pickup beam is greater than 30dB, which can significantly suppress reverberation and interference.
- The dynamic width of sound amplification is greater than 40dB, and the pickup energy range is greater than 40dB.
- The sound pickup is smooth and there is no sudden drop of words.
- Single beam support adjustable from 20 ° to 180 °, multi beam full field amplification (360 °)
- Noise suppression greater than 30dB can filter out most noise sources.
- Under the condition of unchanged sound quality, the sound transmission gain can stably exceed 15dB.
- The sound is clean and thorough, and the speech transmission intelligibility (ST) is high.

Parameters

Microphone Type	Digital silicon microphone 6+1, 360° omnidirectional pickup
Hardware Interface	1 x 3.5mm audio output, Type-C interface DC 5V power supply
Dimension (W x H x D)	200 x 110 x 24.2mm
Pick Up Distance	Clear sound pickup within 5 meters
Sensitivity	-26dBFS
Signal to Noise Ratio	68dB(A)
Sampling Rate	32k sampling high-definition broadband
Sound Reinforcement Algorithm	BF,ANS,AFS,ARC

P50A/P50

Active/Passive Speaker

P50A
Active Speaker

P50
Passive Speaker

Parameters

Driver Composition	6.5" LF x 1, 1.5" HF x 1
Output Power	P50A:50Wx2 P50:50W
Input Impedance	8Ω
Sensitivity（1W@1M）	90±3dB
Frequency Response	65Hz-19kHz
Dimension of Cabinet (W x H x D)	290 x 403 x 220mm

AI adaptive sound pickup and amplification system for lecture halls

CR1

Ultra Far Sound Pickup

Using 32 microphones and 6 meters of ultra long range pickup, capturing more details of human voice, making amplification and interaction more realistic and dynamic.

AI Noise Reduction

Artificial intelligence noise reduction technology, a unique ultra-low latency AI noise reduction model, can effectively suppress various transient noises such as typing and keyboard.

Beam Steering

It contains 12 pickup beams, which can accurately control the pickup area and reduce the impact of reverberation, noise and other environmental interference on speech clarity and intelligibility.

Sound Source Localization

720 blocks are divided at 360 degree horizontally and 90 degree vertically in an area, and highly accurate coordinate positioning is provided through TCP/IP or RS485 to realize camera linkage function.

Remote Interaction

It can be used as the main sound pickup device for remote meetings without switching modes, while providing seamless sound amplification.

Multiple Cascaded Units

It supports 1-4 instrument cascading (simulation, dante cascading) function, and the instrument can be switched smoothly directly, adapting to different scenarios.

Non Sensory Amplification

It supports the 5A algorithm of local sound Amplification, with high restoration degree, and realizes the industry-leading 5-8m sound reinforcement experience without feeling.

POE/Dante Protocol (Corresponding board is selected)

Supports POE power supply and Dante digital audio transmission protocol, making transmission diversified and able to integrate multiple different systems for flexible matching.

Topology Map

Central Control Panel
CP88 x 1

Beamforming Array Microphone
MA60 x 1

Active Injection Moulded Coaxial Customized Loudspeaker
KK61A x 2

100-300m² Large Classroom	60-100m² Medium Classroom
2-6 Sets	2-4 Sets

MA60

Beamforming Array Microphone

Introduction

MA60 beam forming array microphone is a 32 Maiduo ring array, high sensitivity digital microphone. The single microphone has a signal to noise ratio of up to 70db, 360 degree omnidirectional pickup without dead angle, and the normal pickup distance can be up to 8m. The voice is clear and natural, perfectly restoring the original sound of human beings.

MA60 can simultaneously provide local sound amplification and remote interactive pickup. Local sound amplification supports multiple array cascading, making it an ideal solution for large conference rooms, lecture halls, or collaborative spaces.

Features

- Integrated product design, without audio processing host, can flexibly adapt to all kinds of audio, while meeting the needs of local sound reinforcement and remote interaction.
- Support important extension functions such as multi array cascading, POE, DANTE, and camera linkage.
- Support instrument software upgrade, support the setting and adjustment of stable noise, transient noise, reverberation suppression, sound feedback suppression, echo suppression, and automatic gain level; Support input, output and microphone volume adjustment.

Parameters

Basic Features / Working Features			
Output Audio	16kHz, 16bit	Low Volume Pickup	Mini mum 30dB
Frequency Response	100Hz-8KHz	Maximum Gain	30dB
Pick Up Distance	Maximum up to 20m	Radius of Noise Reduction	≤30dB
Number of Microphones	32	Reverberation Suppression	<18dB
Support Algorithm	Support gain communication and intelligence system, noise suppression and reverberation suppression	Sound Quality Rating	MOS >3.6
		Supply Voltage	DC12V
		Working Current	460mA
Support Business	Small voice pickup and speech recognition front-end	Working Temperature	-10°C~50°C
		Storage Temperature	-20°C~70°C
Sensor Features			
Frequency Response	40Hz~20kHZ		
Sensitivity	-32dB	Distortion Degree	0.6%
Directionality	Omnidirectional type	Signal to Noise Ratio	70dB(A)

KK61A

Active Injection Moulded Coaxial Customized Loudspeaker

Introduction

KK61A is a single 6.5-inch two crossover active injection moulded coaxial loudspeaker, with built-in high-performance 6.5-inch coaxial speakers, which presents clear and bright voice, and has thick and full sound quality and low distortion characteristics.

Built in dual channel Class-D amplifier, providing a total power of up to 150W, ensuring excellent sound quality performance. The bottom is equipped with a 25mm double hole tray, which can be tilted at different angles for easy use. At the same time, the rear is equipped with a wall hanging part that can rotate up, down, left, and right for easy installation and layout. The loudspeaker is molded with reinforced polypropylene engineering plastic, which has a beautiful appearance and is durable and portable.

Product is available in black and white.

Features

- The loudspeaker adopts the two-way active coaxial design, with a built-in 6.5 inch coaxial speaker. The high and LF speakers adopt the coaxial magnetic circuit neodymium design, which is highly sensitive, low distortion, high power and high efficiency.
- Built in dual channel Class-D amplifier, amplifier power 100W+50W, 100-240V AC wide voltage, 24bit 48kHz DSP processor is used for the front input of amplifier, XLR line input and output socket and bluetooth module are provided, facilitating connection of sound source.
- The rear is equipped with wall hangers that can rotate up, down, left, and right for easy installation and use.
- The bottom is equipped with a 25mm double hole tray, which can be tilted at different angles of 0 ° and 7.5 ° to meet different usage needs.
- The shell adopts reinforced polypropylene engineering plastic injection molding, which is both aesthetic and durable.

Parameters

Frequency Response	65Hz~20kHz
Coverage Plane Angle (H x V) (-6dB)	80° x 80°
Middle Chassis	1 x 6.5" LF/50mm vc/neodymium magnet
Tweeter	1 x 1.35" HF/34mm vc/neodymium magnet
Amplifier Power	150W + 50W
Max SPL (@ 1m)	117dB
Power Supply	100V - 240V
Dimension (W x H xD)	220 x 350 x 200mm
Material of Cabinet	PP

CP88

Central Control Panel

Introduction

CP88 is a panel type communication and intelligence system processor, which cooperates with automation instrument to realize power switch, volume adjustment and other functions.

Meet the different needs of high-quality sound systems in smart classrooms, various conferences, and other places.

Features

- 3.5-inch (resistive touch) 5-26V voltage.
- Built in Cortex-M3+high-speed FPGA, dual core processor.
- Equipped with RS485 communication interface, it can connection with other central control instrument.
- Volume adjustment can be realized with the adaptive all-in-one machine.

Parameters

Dimension	3.5 inches
Resolving Power	320 x 240
Power Waste	Brightest backlight: 200mA@5V ; Turn off backlight: 96mA@5V
Communication Interface	RS485
Voltage	5-26V (error+0.2V)
Product Dimension (W x H x D)	86 x 27.5 x 86mm
Package Dimension (W x H x D)	97 x 33 x 89mm
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