AD204

(48K-2 In 4 Out Audio Processor)

PRODUCT INTRODUCTION

AD204 is a high-performance 2 input 4 output digital audio processor with 48KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 11 band EQ for each input and output channel, there are also with HPF and LPF, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene, preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 48KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,11-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes:11-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has input and output level indicator, function buttons, the rear panel has RS232 control port and the 100M network port connection. One-button

connection makes the user's operation easier and faster.

- ✓ It supports 32 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate：48KHz

Analog Input：2 x XLR Balanced

Analog Output：4 x XLR Balanced

Input Impedance：10KΩ Balanced / 20KΩ Unbalanced

Output Impedance：50Ω Balanced / 100Ω Unbalanced

Frequency Response：20Hz – 20KHz±0.1dB

AD & DA Converter：32bit ADC& 24bit DAC

Default Output Level：0dBu

Maximum Input /Output Level：18dBu

THD+N Distortion：≤0.002% @ 4dBu 20Hz-20KHz

System Delay：≤1.7ms

Noise Floor：-92dBu

Input To Output Dynamic Range：≥110dB

S/N Ratio：≥110dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 48KHz sampling rate.

Parametric equalizer： input and output channels with 11-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

AD306

(48K-3 In 6 Out Audio Processor)

PRODUCT INTRODUCTION

AD306 is a high-performance 3 input 6 output digital audio processor with 48KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 11 band EQ for each input and output channel, there are also with HPF and LPF, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 48KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,11-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes:11-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has input and output level indicator, function buttons, the rear panel has RS232 control port and the 100M network port connection. One-button

connection makes the user's operation easier and faster.

- ✓ It supports 32 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate：48KHz

Analog Input：3 x XLR Balanced

Analog Output：6 x XLR Balanced

Input Impedance：10KΩ Balanced / 20KΩ Unbalanced

Output Impedance：50Ω Balanced / 100Ω Unbalanced

Frequency Response：20Hz – 20KHz±0.1dB

AD & DA Converter：32bit ADC& 24bit DAC

Default Output Level：0dBu

Maximum Input /Output Level：18dBu

THD+N Distortion：≤0.002% @ 4dBu 20Hz-20KHz

System Delay：≤1.7ms

Noise Floor：-92dBu

Input To Output Dynamic Range：≥110dB

S/N Ratio：≥110dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 48KHz sampling rate.

Parametric equalizer： input and output channels with 11-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

AD408

(48K-4 In 8 Out Audio Processor)

PRODUCT INTRODUCTION

AD408 is a high-performance 4 input 8 output digital audio processor with 48KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 11 band EQ for each input and output channel, there are also with HPF and LPF, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 48KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,11-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes:11-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has input and output level indicator, function buttons, the rear panel has RS232 control port and the 100M network port connection. One-button

connection makes the user's operation easier and faster.

- ✓ It supports 32 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate：48KHz

Analog Input：4 x XLR Balanced

Analog Output：8 x XLR Balanced

Input Impedance： 10KΩ Balanced / 20KΩ Unbalanced

Output Impedance：50Ω Balanced / 100Ω Unbalanced

Frequency Response：20Hz – 20KHz±0.1dB

AD & DA Converter：32bit ADC& 24bit DAC

Default Output Level：0dBu

Maximum Input /Output Level：18dBu

THD+N Distortion：≤0.002% @ 4dBu 20Hz-20KHz

System Delay：≤1.7ms

Noise Floor：-92dBu

Input To Output Dynamic Range：≥110dB

S/N Ratio：≥110dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 48KHz sampling rate.

Parametric equalizer： input and output channels with 11-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct
Bessel slope: 6/12/18/24/30/36/42/48dB/Oct
Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS
Start control time: 1ms ~ 1000ms;
Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS
Ratio: 1 ~ 20
Start-up time: 1ms ~ 1000ms
Recovery time: 1ms ~ 10000ms
Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS
Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

LM824T

(96K-2 In 4 Out Audio Processor)

PRODUCT INTRODUCTION

LM824T is a high-performance 2 input 4 output digital audio processor with 96KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 13 band EQ for each input and output channel, there are also with HPF and LPF, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,13-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes:13-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 64 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate： 96KHz

Analog Input： 2 x XLR Balanced

Analog Output： 4 x XLR Balanced

Input Impedance： 10KΩ Balanced / 20KΩ Unbalanced

Output Impedance： 50Ω Balanced / 100Ω Unbalanced

Frequency Response： 20Hz – 20KHz±0.1dB

AD & DA Converter： 32bit ADC and 24bit DAC

Default Output Level： 0dBu

Maximum Input/Output Level： 22dBu

THD+N Distortion： ≤0.002% @ 4dBu 20Hz-20KHz

System Delay： ≤3.15ms

Noise Floor： -90dBu

Input To Output Dynamic Range： ≥112dB

S/N Ratio： ≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer： input and output channels with 13-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

LM836T

(96K-3 In 6 Out Audio Processor)

PRODUCT INTRODUCTION

LM836T is a high-performance 3 input 6 output digital audio processor with 96KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 13 band EQ for each input and output channel, there are also with HPF and LPF, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,13-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes:13-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 64 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate: 96KHz

Analog Input: 3 x XLR Balanced

Analog Output: 6 x XLR Balanced

Input Impedance: 10KΩ Balanced / 20KΩ Unbalanced

Output Impedance: 50Ω Balanced / 100Ω Unbalanced

Frequency Response: 20Hz – 20KHz±0.1dB

AD & DA Converter: 32bit ADC and 24bit DAC

Default Output Level: 0dBu

Maximum Input/Output Level: 22dBu

THD+N Distortion: ≤0.002% @ 4dBu 20Hz-20KHz

System Delay: ≤3.15ms

Noise Floor: -90dBu

Input To Output Dynamic Range: ≥112dB

S/N Ratio: ≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer： input and output channels with 13-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct
Bessel slope: 6/12/18/24/30/36/42/48dB/Oct
Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS
Start control time: 1ms ~ 1000ms;
Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS
Ratio: 1 ~ 20
Start-up time: 1ms ~ 1000ms
Recovery time: 1ms ~ 10000ms
Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS
Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

LM848T

(96K-4 In 8 Out Audio Processor)

PRODUCT INTRODUCTION

LM848T is a high-performance 4 input 8 output digital audio processor with 96KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 13 band EQ for each input and output channel, there are also with HPF and LPF, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,13-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes:13-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 64 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate： 96KHz

Analog Input： 4 x XLR Balanced

Analog Output： 8 x XLR Balanced

Input Impedance： 10KΩ Balanced / 20KΩ Unbalanced

Output Impedance： 50Ω Balanced / 100Ω Unbalanced

Frequency Response： 20Hz – 20KHz±0.1dB

AD & DA Converter： 32bit ADC and 24bit DAC

Default Output Level： 0dBu

Maximum Input/Output Level： 22dBu

THD+N Distortion： ≤0.002% @ 4dBu 20Hz-20KHz

System Delay： ≤3.15ms

Noise Floor： -90dBu

Input To Output Dynamic Range： ≥112dB

S/N Ratio： ≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer: input and output channels with 13-band EQ adjust

6 filter types: peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range: -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range: -72dB to +12dB, step accuracy: 0.1dB

Center frequency: adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth: Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate: Threshold range: -100dBFS ~ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor: Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter: Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay: Each input and output channel has up to 2000ms delay adjustable

LM824RT

(96K+FIR: 2 In 4 Out Audio Processor)

PRODUCT INTRODUCTION

LM824RT is a high-performance 2 input 4 output digital audio processor with 96KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 16 band EQ for each input and output channel, there are also with HPF and LPF, FIR filter, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,16-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes: FIR filter, 16-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 128 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate：96KHz

Analog Input：2 x XLR Balanced

Analog Output：4 x XLR Balanced

Input Impedance：10KΩ Balanced / 20KΩ Unbalanced

Output Impedance：50Ω Balanced / 100Ω Unbalanced

Frequency Response：20Hz – 20KHz±0.1dB

AD & DA Converter： 32bit ADC and 24bit DAC

Default Output Level：0dBu

Maximum Input/Output Level：22dBu

THD+N Distortion：≤0.002% @ 4dBu 20Hz-20KHz

System Delay：≤3.15ms

Noise Floor：-90dBu

Input To Output Dynamic Range：≥112dB

S/N Ratio：≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer： input and output channels with 16-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

LM836RT

(96K+FIR: 3 In 6 Out Audio Processor)

PRODUCT INTRODUCTION

LM836RT is a high-performance 3 input 6 output digital audio processor with 96KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 16 band EQ for each input and output channel, there are also with HPF and LPF, FIR filter, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,16-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes: FIR filter, 16-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 128 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate：96KHz

Analog Input：3 x XLR Balanced

Analog Output：6 x XLR Balanced

Input Impedance：10KΩ Balanced / 20KΩ Unbalanced

Output Impedance：50Ω Balanced / 100Ω Unbalanced

Frequency Response：20Hz – 20KHz±0.1dB

AD & DA Converter：32bit ADC and 24bit DAC

Default Output Level：0dBu

Maximum Input/Output Level：22dBu

THD+N Distortion：≤0.002% @ 4dBu 20Hz-20KHz

System Delay：≤3.15ms

Noise Floor：-90dBu

Input To Output Dynamic Range：≥112dB

S/N Ratio：≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer： input and output channels with 16-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

LM848RT

(96K+FIR: 4 In 8 Out Audio Processor)

PRODUCT INTRODUCTION

LM848RT is a high-performance 4 input 8 output digital audio processor with 96KHz sampling rate. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 16 band EQ for each input and output channel, there are also with HPF and LPF, FIR filter, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, noise gate, gain control, mute, phase,16-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes: FIR filter, 16-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 128 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate：96KHz

Analog Input：4 x XLR Balanced

Analog Output：8 x XLR Balanced

Input Impedance：10KΩ Balanced / 20KΩ Unbalanced

Output Impedance：50Ω Balanced / 100Ω Unbalanced

Frequency Response：20Hz – 20KHz±0.1dB

AD & DA Converter：32bit ADC and 24bit DAC

Default Output Level：0dBu

Maximum Input/Output Level：22dBu

THD+N Distortion：≤0.002% @ 4dBu 20Hz-20KHz

System Delay：≤3.15ms

Noise Floor：-90dBu

Input To Output Dynamic Range：≥112dB

S/N Ratio：≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer： input and output channels with 16-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

LM824RTS

(96K+FIR+AES: 2 Input 4 Output ）

PRODUCT INTRODUCTION

LM824RTS is a high-performance 2 input 4 output digital audio processor with 96KHz sampling rate. Supports 2 channel analog/1 channel AES/EBU input and 4 channel analog/2 channel AES/EBU output. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 16 band EQ for each input and output channel, there are also with HPF and LPF, FIR filter, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, AES/EBU input, noise gate, gain control, mute, phase,16-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes: FIR filter, 16-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, AES/EBU output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1^{st}, all pass 2^{nd}, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 128 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate：96KHz

Analog Input：2 x XLR Balanced

Analog Output：4 x XLR Balanced

Input Impedance：10KΩ Balanced / 20KΩ Unbalanced

Output Impedance：50Ω Balanced / 100Ω Unbalanced

Frequency Response：20Hz – 20KHz±0.1dB

AD & DA Converter： 32bit ADC and 24bit DAC

Default Output Level：0dBu

Maximum Input/Output Level：22dBu

THD+N Distortion：≤0.002% @ 4dBu 20Hz-20KHz

System Delay：≤3.15ms

Noise Floor：-90dBu

Input To Output Dynamic Range：≥112dB

S/N Ratio：≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer： input and output channels with 16-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct
Bessel slope: 6/12/18/24/30/36/42/48dB/Oct
Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS
Start control time: 1ms ~ 1000ms;
Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS
Ratio: 1 ~ 20
Start-up time: 1ms ~ 1000ms
Recovery time: 1ms ~ 10000ms
Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS
Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

LM836RTS

(96K+FIR+AES: 3 Input 6 Output ）

PRODUCT INTRODUCTION

LM836RTS is a high-performance 3 input 6 output digital audio processor with 96KHz sampling rate. Supports 3 channel analog/2 channel AES/EBU input and 6 channel analog/3 channel AES/EBU output. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 16 band EQ for each input and output channel, there are also with HPF and LPF, FIR filter, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, AES/EBU input, noise gate, gain control, mute, phase,16-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes: FIR filter, 16-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, AES/EBU output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 128 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate：96KHz

Analog Input：3 x XLR Balanced

Analog Output：6 x XLR Balanced

Input Impedance：10KΩ Balanced / 20KΩ Unbalanced

Output Impedance：50Ω Balanced / 100Ω Unbalanced

Frequency Response：20Hz – 20KHz±0.1dB

AD & DA Converter：32bit ADC and 24bit DAC

Default Output Level：0dBu

Maximum Input/Output Level：22dBu

THD+N Distortion：≤0.002% @ 4dBu 20Hz-20KHz

System Delay：≤3.15ms

Noise Floor：-90dBu

Input To Output Dynamic Range：≥112dB

S/N Ratio：≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer：input and output channels with 16-band EQ adjust

6 filter types：peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range：-30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range：-72dB to +12dB, step accuracy: 0.1dB

Center frequency：adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth：Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate：Threshold range: -100dBFS ~ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor：Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter：Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay：Each input and output channel has up to 2000ms delay adjustable

LM848RTS

(96K+FIR+AES: 4 In 8 Out)

PRODUCT INTRODUCTION

LM848RTS is a high-performance 4 input 8 output digital audio processor with 96KHz sampling rate. Supports 4 analog/2 AES/EBU input and 8 analog/4 AES/EBU output. Audio processing is performed by 32bit ADC and 24bit DAC and high-speed DSP processing. In addition to 16 band EQ for each input and output channel, there are also with HPF and LPF, FIR filter, all-pass filter, port input, noise gate, gain control, input compressor, output limiter, scene preset, and delay for each channel. Powerful functions such as adjustable delay up to 2000ms, equipped with network port, USB port and RS-232 can realize quick configuration and debugging. This equipment performs a series of optimizations and adjustments for high-quality sound reinforcement systems.

PRODUCT FEATURES

- ✓ 32-bit DSP processing, 96KHz sampling rate, 32bit ADC and 24bit DAC.
- ✓ The input processing includes: port input, AES/EBU input, noise gate, gain control, mute, phase,16-band EQ, Xover, input compression, 2000ms input adjustable delay, linkage adjustment and other processing functions.
- ✓ The output processing includes: FIR filter, 16-band EQ, 2000ms adjustable output delay, gain control, mute, phase, Xover, output limit, port output, AES/EBU output, linkage adjustment and other processing functions.
- ✓ All inputs and outputs channels can be freely routing, and the name of each input and output channel can be changed.
- ✓ All input and output channels have independent phase curve adjustment function.
- ✓ Parameter settings between any channels can be freely copied, and any channel can be linked adjustment.
- ✓ All input and output channels can be auto EQ by importing channel parameters
- ✓ There are six types of equalization: peaking, notch, all pass 1st, all pass 2nd, high shelf and low shelf.
- ✓ All input and output channels have HPF and LPF, the types are: Butterworth, Bessel, Linkwitz-Riley, the slope is optional from 6/12/18/24/30/36/42/48 dB/Oct.
- ✓ All input compressor have thresholds, ratios, start-up time, recovery time and compensation gain adjustable.
- ✓ All output limiter are included: threshold, recovery time can be adjusted, brick wall type limiter, starting time is 0, absolute limit.
- ✓ The delay of all input and output channels with 2000ms delay.
- ✓ Built-in signal generator, input mode can choose pink noise, white noise, sweep frequency and 20Hz-20KHz sine wave adjustable, signal amplitude adjustable.
- ✓ The front panel has USB control port for serial port connection, input and output level indicator and function buttons, the rear panel has RS232 control port and the

100M network port connection. One-button connection makes the user's operation easier and faster.

- ✓ It supports 128 preset scenes to be saved, and the device scene algorithm and each preset can be saved and loaded separately. It also has the permission management function to make the device more secure.
- ✓ Application: professional performances, bar, conference rooms, courtrooms, auditoriums, multi-function halls, etc.

TECHNICAL PARAMETERS

Sample Rate: 96KHz

Analog Input: 4 x XLR Balanced

Analog Output: 8 x XLR Balanced

Input Impedance: 10KΩ Balanced / 20KΩ Unbalanced

Output Impedance: 50Ω Balanced / 100Ω Unbalanced

Frequency Response: 20Hz – 20KHz±0.1dB

AD & DA Converter: 32bit ADC and 24bit DAC

Default Output Level: 0dBu

Maximum Input/Output Level: 22dBu

THD+N Distortion: ≤0.002% @ 4dBu 20Hz-20KHz

System Delay: ≤3.15ms

Noise Floor: -90dBu

Input To Output Dynamic Range: ≥112dB

S/N Ratio: ≥112dB

DSP PROCESSING TECHNICAL PARAMETERS

Use TI OMAP-L138 high-performance DSP, with a computing capacity of up to 3648 MIPS and 2746 MFLOPS per second, built-in multiple audio algorithms, 96KHz sampling rate.

Parametric equalizer： input and output channels with 16-band EQ adjust

6 filter types： peaking, notch, all pass 1st, all pass 2nd, high shelf, low shelf

The filter gain range： -30dB to +30dB, the step accuracy is 0.1dB

Input and output gain range： -72dB to +12dB, step accuracy: 0.1dB

Center frequency： adjustable within the frequency range of 20Hz~20KHz with step accuracy of 1Hz

Filter Q value Bandwidth： Peaking, Notch, All-pass 2nd, Q value: 0.18 ~ 144.27

High and low pass filter: Butterworth slope: 6/12/18/24/30/36/42/48dB/Oct

Bessel slope: 6/12/18/24/30/36/42/48dB/Oct

Linkwitz-Riley: 12/24/36/48dB/Oct

Input noise gate： Threshold range: -100dBFS ～ 0dBFS

Start control time: 1ms ~ 1000ms;

Recovery time: 1ms ~ 1000ms

Input compressor： Threshold range: -84dBFS ~ 0dBFS

Ratio: 1 ~ 20

Start-up time: 1ms ~ 1000ms

Recovery time: 1ms ~ 10000ms

Compensation gain: -24dB ~ 30dB

Output limiter： Threshold range: -84dBFS ~ 0dBFS

Recovery time: 0-100000ms

Delay： Each input and output channel has up to 2000ms delay adjustable

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