SIP2401V Network Audio Module Specifications

Summary	This article briefly introduces the general introduction of the SIP2401V network audio module. For more information, please refer to the following documents: "SIP serial protocol.docx" "Network Audio Device Configuration Tool Instructions" "Network Audio System Software Instructions"

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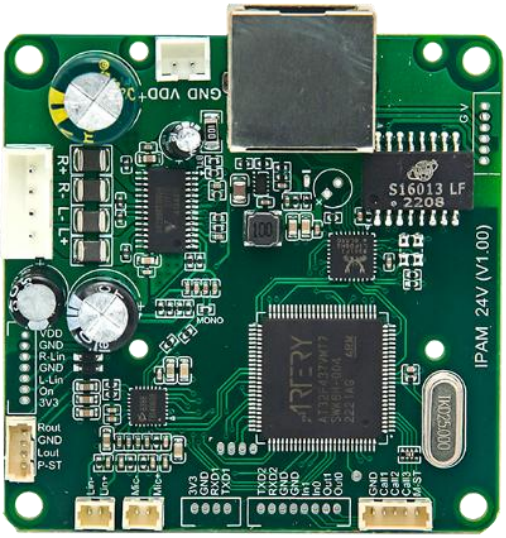
1. Module introduction



SIP2401V network audio module is a general-purpose independent SIP audio function module with 2*15W power amplifier audio output, which can be easily embedded into OEM products. This module encodes and decodes the SIP protocol and RTP audio stream from the network.

The module supports a variety of network protocols and audio codec protocols, and can be used in applications such as VoIP and IP paging and high-quality music streaming media playback. At the same time, SIP2401V also provides a serial port, allowing users to control through commands.

For hardware product R&D integration, our company provides suggestions for using development specifications, including schematic diagrams, pin distributions and custom carrier PCB layouts.

SIP2401Vand**SIP2101V**and**SIP2701V**Compatible, it is a playback and intercom module, and does not support GPIO and serial port transparent transmission functions.

SIP2402Vand**SIP2102V**and**SIP2702V**Compatible, it is a playback module that only supports playback and passive call reception.

SIP2403Vand**SIP2103V**and**SIP2703V**Compatible, it is a full-featured module that supports intercom, playback, GPIO and serial port transparent transmission functions.

This series of modules can be applied in the following fields:

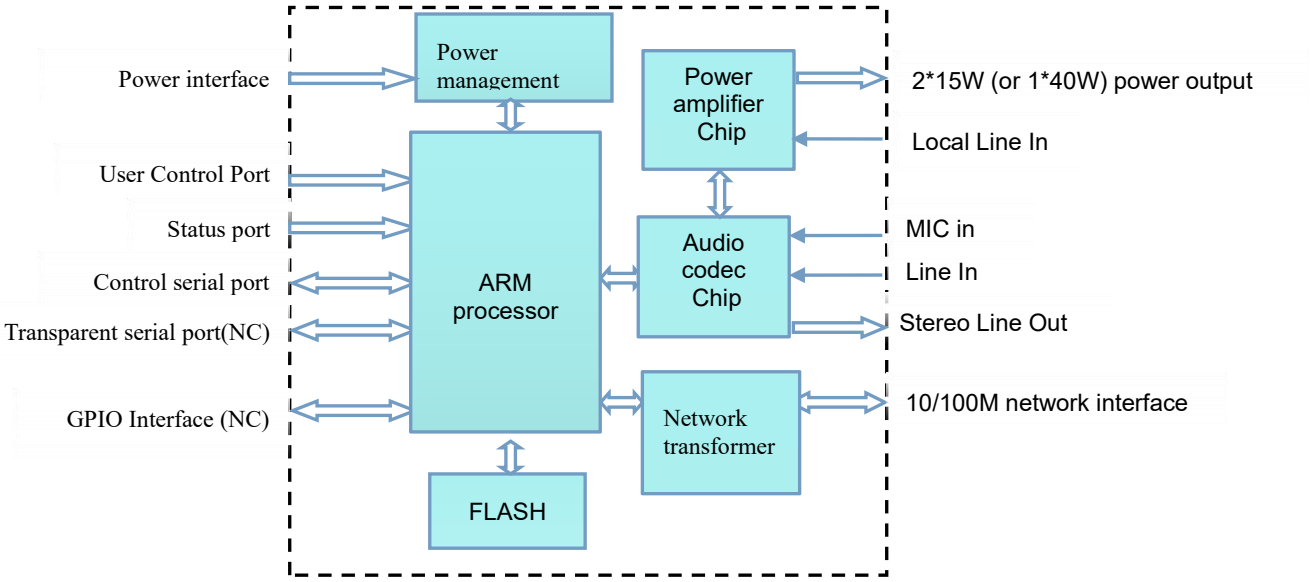
- **Various commercial network audio streaming applications**
- **Network alarm, network player**
- **Universal two-way full-duplex VoIP intercom for paging and intercom applications**
- **Source Encoder for IP Audio Distribution Applications**
- **Background Music and Music On Hold Encoder for VoIP Applications**
- **Compatible with various low-latency IP audio encoders**
- **Unmanned parking lot, self-service charging station intercom terminal**
- **Expressway tunnel intercom, toll station intercom terminal**

The SIP network audio module works with other SIP endpoints in a standard SIP system. In addition to supporting the standard sip protocol, this series of modules also has our proprietary protocol, so it can be used with our standard broadcast software. Implementations include additional features such as real-time paging, scheduled broadcasts, background music and emergency broadcasts. The corresponding software package can be downloaded from the SINREY website. At

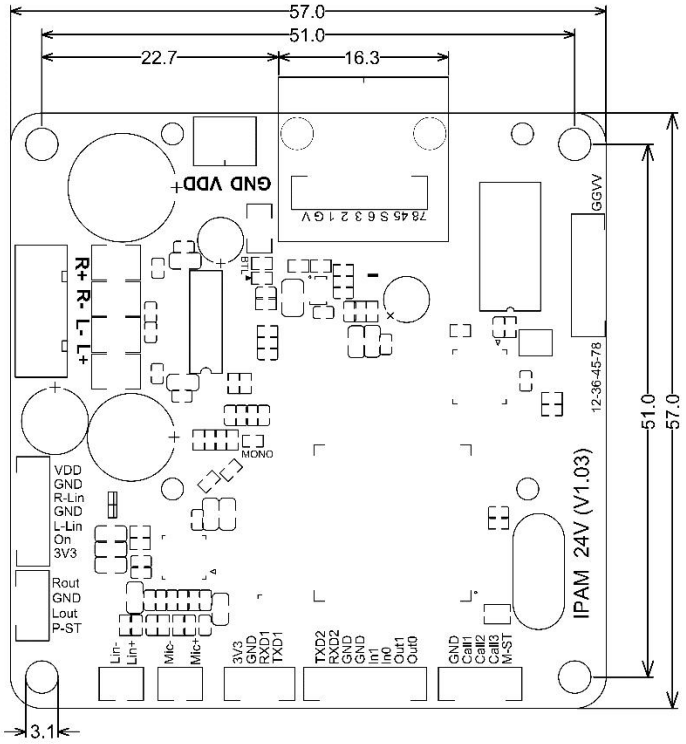
the same time, SINREY also provides Manager configuration software to configure the IP address and other parameters of the module.

2. Module structure

The SIP2401V module uses an ARM processor architecture with a speed of up to 400MHz plus a professional two-way audio Codec codec, The ARM processor is responsible for data transmission, the analysis and execution of user commands, and the control of the power amplifier interface, and the professional audio Codec is responsible for audio input and output. Its internal structure is as follows:



Its shape and structure are as follows:



For easier use, the SIP2401V module uses a connector as an external interface and integrates a network interface.

3. Functions and Features

- How audio works:

Decode: That is, audio playback. Receive the audio stream from the network, and output high-quality audio signal through the line after decoding by the module. Currently supports the following audio formats:MP3.WAV (PCM +IMA ADPCM), G.711, G.722, etc. Can play audio streams up to 48khz sampling rate 320kps bit rate.

Receive the audio stream sent by TCP, UDP, RTP, multicast, broadcast, etc. in the network, and play it after local decoding.

Encode: Support G.711a/u, G.722 and RTP multicast mode.

- Control method:

Button control: Directly through the IO port on the module, control the module SIP intercom or initiate RTP multicast

Serial port control: Through the control serial port on the module, control the module SIP intercom or initiate RTP multicast

Network Control: Through network commands on other devices in the network, control the control module SIP intercom or initiate RTP multicast.

- System status output:

Audio input (M-ST), output status (P-ST): You can connect indicator lights or control relays.

- Amplifier output:

  With 2*15W power amplifier output, it can also be changed to 1*40W power amplifier output by adjusting the circuit

  With 1 channel local audio input to realize the local sound amplification output function

- Extensions:

  POE power supply: Provide POE power receiving module interface

  Real Time Clock: The module is powered off, or the time is not adjusted by the server to provide accurate time, which can be obtained through the serial port or network, or the audio file can be played regularly.

4. Technical parameters

- power input: Voltage DC 12 ~ 24V
  Maximum working current, 2000mA
- temperature: Operating temperature range Industrial grade: -40～85℃
  storage temperature range -40～85℃
- Network : 10/100M Base adaptive Ethernet interface
- Mic input: Typical amplitude 50mVrms, signal-to-noise ratio 95dB
- LineIn input: Typical amplitude 1000mVrms, SNR 95dB
- Line Out output:Load 10KΩ, typical 1000Vrms, signal-to-noise ratio 95dB (in playback mode)
- RL-Line input: Local PA input, typical amplitude 1000mVrms, signal-to-noise ratio 95dB
- Power output: Power amplifier output, output power 2*15W or 1*40W, connected to 8 ohm constant resistance speaker
- Decoding : Provide stereo playback, up to 48kHz, 320kbps audio stream, support MP3.WAV (PCM +IMA ADPCM), G.711 a/u, G.722 and other formats.
  The minimum delay is50ms
- Encoding : Support G.711a/u, G.722 and RTP mode
  The minimum delay is 30ms
- Bidirectional : Two-way intercom, with high-performance echo suppression algorithm, support G.711 a/u, G.722 encoding
  The minimum delay is 80ms
- Intercom control port: There are three control interfaces of Call1, Call2 and Call3 in total, and this interface has two working modes: [Pulse Mode] and [Hold Mode].
- control serial interface: Baud rate 115.2kbps
- module size: Length x width x height: 5.7 x 5.7 x 1.7cm

5. Hardware interface description

SIP2401V network audio modules, all external connections use terminals, the power supply uses 2.0mm terminals, the network uses standard RJ45 connectors, and the others are 1.25mm connectors.

port type definition

P ——— power supply

AI ——— Analog input (in this case, audio input)
AO ——— Analog output (in this case, audio output)
I ——— Digital input port, TTL level, withstand voltage 5V
o ——— Digital output port, TTL level
I/O ——— Digital input and output port, TTL level
N ——— network port

Because many of the hardware interfaces mentioned later are directly connected to the processor. In most cases, the digital input and output terminals are connected to the outside, and an external drive circuit is required to protect the processor from damage. Analog input/analog output also need to increase the amplifier circuit and input protection circuit.

1. Power input port

The network audio module adopts DC power supply mode, wide voltage input, power supply range DC12 ~ 24V. Because it involves audio and a processor up to 400MHz, the input power supply voltage must be as stable as possible, and the minimum current is guaranteed to be 500mA.

Interface description:

	definition	type	describe
1	VDD	P	Power input, the voltage is DC 12-24V
2	GND	P	power ground

2. Control serial port (Uart1)

The network audio module has a control serial port, and the working information and parameters of the network audio module can be queried through the control serial port, and the work of the network audio module can also be controlled. You can even communicate with any device on the network. Generally, we use this port as a keyboard or monitor interface. Our company also provides display and keyboard kits, which can be used as a supporting device for video-on-demand terminals and paging microphones.

For details, see **【Network audio module serial port】**

Interface description:

	interface	type	describe
1	TXD1	o	Control the output of the serial port and connect the input of the external serial port
2	RXD1	I	Control the serial port input, connect the output of the external serial port
3	GND	P	ground wire
4	3V3	P	3.3V power output. Because the power supply is directly connected to the working power supply of the module, in order to ensure the performance of the module, it is not recommended to connect 3V3 to the external circuit

3. Line Out line output port

Line output is the audio output interface when the module is playing or talking.

Interface description:

	definition	type	describe
1	Routing	AO	right channel output
2	GND	P	land
3	Lout	AO	left channel output
4	P-ST	o	Audio output indication. Output low level when there is no sound, output high level 3.3V when talking or playing

If a stereo signal such as MP3 is played, a stereo output will be provided; if the audio source from the network is detected to be a mono audio format such as mono ADPCM/PCM, the left and right channels will output the same signal.

As an audio output indicator, P-ST can control the enable terminal of the digital power amplifier to ensure that the speaker has no noise when there is no sound output, and save energy consumption at the same time. P-ST can also be used as the control terminal of a relay to control the power supply of the power amplifier circuit of the rear stage.

The change of the P-ST signal is determined by whether there is an audio stream in the network, regardless of the assignment of the output audio signal. At the same time, the P-ST will return to low level when no audio signal is received for more than 1 second. Sometimes, we want P-ST to become high level without sound output, can play audio stream without sound.

4. Amplifier output port

SIP2401V has 2*15W power amplifier output, and can also be changed to 1*40W power amplifier output by adjusting the circuit

Interface description:

	definitio n	type	describe
1	R+	AO	Power output of the right channel, connected to the positive pole of the passive speaker
2	R-	AO	Right channel power output, connect passive speaker negative pole
3	L-	AO	Left channel power output, connect passive speaker negative pole
4	L+	AO	Left channel power output, connect to positive pole of passive speaker

5. Line In Line/Mic input port

Line In/Mic input is the audio input interface when the module is broadcasting or intercom. This module can provide 1 line input or directly connect to a condenser microphone. Mic input 180 mVpp (maximum), Line In audio line input 3.3 Vpp (maximum), the gain can be adjusted.

Mic can be directly connected to a condenser microphone, Line In can be single-ended input

Line In and Mic have two working modes: ①Mic is valid/Line In is invalid; ②Mic is

invalid/Line In is valid; the working mode can be selected through Manager or serial port or network.

Interface description:

	definition	type	describe
1	Mic+	AI	Connect the positive pole of the condenser microphone
2	Mic-	AI	Connect the negative pole of the condenser microphone
3	Lin+	AI	Connect Line In Positive
4	Lin-	AI	Connect the line input negative pole, or connect the line input ground

6. Local PA input RL-Line in

1	Rin	AO	right channel output
2	GND	P	land
3	Lin	AO	left channel output
4	On	I	Power amplifier control interface, input high level 3.3V, you can start the power amplifier work

If On is not connected to a high level, the power amplifier chip is turned off when the module is idle and standby, and the power amplifier chip is automatically turned on when the module is playing intercom.

When the module needs external input (such as CD player, mixer) for local sound amplification, connect On to 3.3V high level to make the power amplifier chip enter the start-up playback state. Even when the module is idle and standby, local sound amplification can also be performed.

7. Intercom/collection and broadcast control port

For simple application scenarios, you only need to connect a few key switches to realize two-way intercom or broadcasting. For this reason, SIP2403V is specially designed with a control terminal, which is defined as follows.

Interface description:

	definition	type	describe
1	GND	P	ground wire
2	Call1	I	The first call/collection and broadcasting control
3	Call2	I	The second call/collection and broadcasting control
4	Call3	I	The third call/collection and broadcasting control
5	M-ST	o	Intercom/collecting and broadcasting work instructions, usually output low level, when intercom or broadcasting, output high level 3.3V

The call port is active at low level, it can be directly connected to a switch or button, and a 10/100k pull-up resistor (optional) can be added, and it cannot be directly connected to a power supply (including 3.3V).

The input port has two working modes: [pulse mode], that is, press once to call, and press again to disconnect. [Hold mode], it needs to be pressed all the time to work, and the port is released when it is released. The calling end can control the module for intercom, or broadcast (unicast, multicast or broadcast).

Various parameters corresponding to the calling end can be configured through the Manager.

8. Network Interface

The network interface adopts two connection methods:

1) Adopt RJ45 standard 10/100M adaptive network interface. There are two indications on the RJ45 socket, one for connection and one for data sending and receiving.
2) With the tail wire method, most of the camera tail wires can be used directly. In this way, the network audio module can be placed inside the device.

Interface description:

	definitio n	type	describe
1	V	P	Power input, the voltage is DC 12 ~ 2V
2	G	P	power ground
3	1	N	Network cable 1
4	2	N	Network cable 2
5	3	N	Network cable 3
6	6	N	Network cable 6
7	S	o	network connection indicator
8		N	Network cable 7 and network cable 8 can supply power to the POE module (can be disconnected)
9		N	Network cable 4, network cable 5, can supply power to the POE module (can not be connected)

Use the network tail cable, you can choose 9-core or 11-core tail cable, if you need to use the POE power receiving module, you must use the 11-core tail cable. Among them, 7 cores (or 9 cores) are connected to this terminal, and the other two cores are connected to the power terminal.

Among them, 1, 2, 3, and 6 can be exchanged with each other, as long as 1, 2 or 3, 6 are paired. 8 and 9 can also be interchanged.

9. POE powered (PD) interface

Through this interface, an external POE power receiving module can be connected to achieve the function of network power supply. After the POE power receiving module is installed, DC power supply can also be used.

Notice: The POE power receiving module outputs 12V voltage, which is connected to the power input of the module. If a DC external power supply is used at the same time, unpredictable conditions may result.

interface description:

	definitio	type	describe

	n		
1	V	P	Power input, the voltage is DC 12 ~ 24V
2	G	P	power ground
3	1-2	N	Network cable 1, 2 network cable middle tap
4	3-6	N	Network cable 3, 6 network cable middle tap
5	4-5	N	Network cable 4, 5 network cable (parallel connection)
6	7-8	N	Network cable 7, 8 network cable (parallel connection)

6. product selection

1. SIP2400V series module model list

The SIP2400V series modules adopt the socket structure, and the models and corresponding functions are as follows:

	play	interc om	multic ast	Intercom broadcast control interface	Line Out	Line In Mic	control serial port	GPIOs Universal serial port	2×15W amplifier
SIP2401V	√	√	√	√	√	√	√	x	√
SIP2402V	√	√*	x	x	√	x	x	x	√
SIP2403V	√	√	√	√	√	√	√	√	√

*SIP2402V does not support active calling, but it can passively receive calls. In the intercom state, audio can only be received but cannot be sent.

2. List of other compatible module models

SIP2100VThe series of modules have a lot of functions, but not every application needs all the interfaces, the models and corresponding functions are as follows

	play	interco m	multica st	Intercom broadcast control interface	Line Out	Line In Mic	control serial port	GPIOs Universal serial port
SIP2101V	√	√	√	√	√	√	√	x
SIP2102V	√	√*	x	x	√	x	x	x
SIP2103V	√	√	√	√	√	√	√	√

*SIP2102V does not support active calling, but it can passively receive calls. In the intercom state, audio can only be received but cannot be sent.

The SIP2700V series modules adopt pin structure, and the models and corresponding functions are as follows:

	play	interco m	multica st	Intercom broadcast control interface	Line Out	Line In Mic	control serial port	GPIOs Universal serial port
SIP2701V	√	√	√	√	√	√	√	x

SIP2702V	√	√*	x	x	√	x	x	x
SIP2703V	√	√	√	√	√	√	√	√

*SIP2702V does not support active calling, but it can passively receive calls. In the intercom state, audio can only be received but cannot be sent.

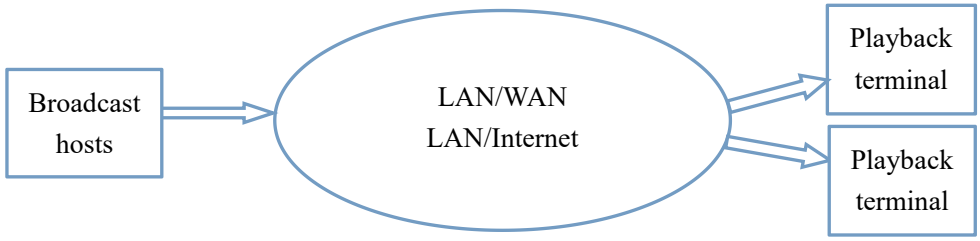
7. Application scenarios

Whether it is used for audio broadcasting, two-way intercom, environmental sound collection,SIP2401V network audio module can be used with very few peripherals. For example, it can be integrated into active speakers as IP network speakers, integrated into traditional constant-voltage power amplifiers as network constant-voltage power amplifiers; it can also be integrated into industrial robots to add playback, intercom and other functions.

1. Broadcast application

Network audio equipment can accept audio streams from the network and convert them into analog audio signals for output after decoding. The figure below is a standard playback application scenario.

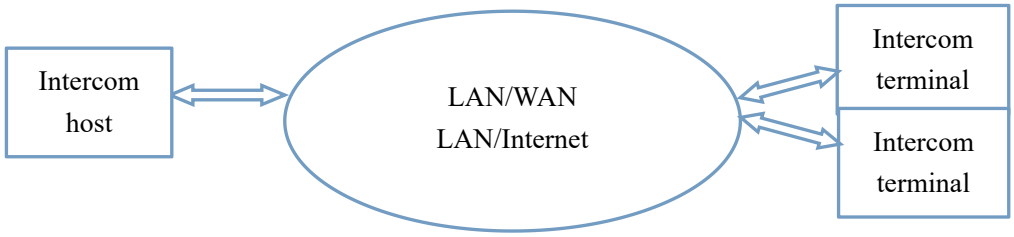
SIP2401V network audio module is used as a playback terminal, and only needs to increase the power supply and external power amplifier. It can be used as a single device, or integrated in other audio devices (power amplifier, or active speakers, etc.), making it a network audio playback terminal.



The broadcast host can use our company's broadcast system control software, broadcasting equipment, network microphone, etc., or the user can develop his own broadcast software based on our company's DLL dynamic library. Both the playback software and the playback development kit use our company's private protocol, and only support the playback of our company's equipment modules. As a playback terminal, the network audio module uses our proprietary protocol for playback control, and can only be used in a LAN or VPN intranet. In fact, customers can also choose to develop their own playback software of SIP protocol, which can be used in LAN or Internet.

2. Intercom application

Network audio devices can be used in a two-way SIP intercom system.



SIP2401V network audio module is used as an intercom broadcast terminal, only need to add power supply, external power amplifier, speaker and microphone. For example: the intercom monitoring function is added to the parking lot system, teaching central control, robot and other equipment.

The SIP2401V network audio module can also be used as a SIP intercom broadcast host. In addition to adding a power supply, external power amplifier, speaker and microphone, a keyboard display controlled by a serial port is also required.

The Intercom host can use our company's broadcast system control software, network microphone, etc., or the user can develop his own broadcast software based on our company's DLL dynamic library.

Our network audio module can be used as a broadcast intercom in a local area network or in the Internet.