SINREY

SIP2102V Network Audio Module Specifications

Summary	This article briefly introduces the general introduction of the SIP2102V 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

The SIP2102V network audio module is a general-purpose stand-alone SIP audio playback module that can be easily embedded into OEM products. This module decodes and plays the SIP protocol and RTP audio stream from the network.

The module supports multiple network protocols and audio decoding protocols, and can be used in applications such as VoIP and IP paging and high-quality music streaming.

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

SIP2101V and **SIP2401V** and **SIP2701V** Compatible, it is a playback and intercom module, and does not support GPIO and serial port transparent transmission functions.

SIP2102V and **SIP2402V** and **SIP2702V** Compatible, it is a playback module that only supports playback and passive call reception.

SIP2103V and **SIP2403V** 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 SIP2102V network audio module can be used with other SIP terminals in the standard SIP system. In addition to supporting the standard sip protocol, the module also supports the RTP multicast protocol, which can receive RTP multicast audio data and decode and play it. At the same time, it can be used together with the RTP multicast SDK switch package we provide. SIP2102V can be applied in scenarios such as real-time paging broadcast, scheduled broadcast, background music
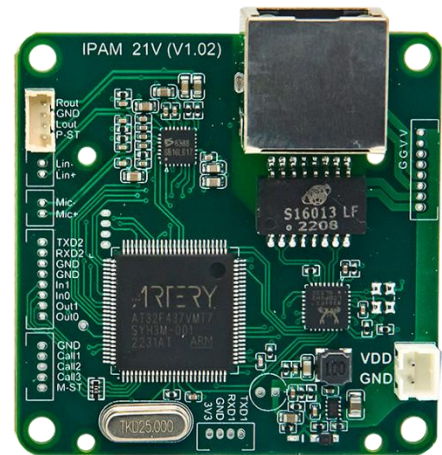
and emergency broadcast. The corresponding software package can be downloaded from the SINREY website. At the same time, SINREY also provides Manager configuration software to configure the IP address and other parameters of the module.

2. Module structure

SIP2102V uses an ARM processor architecture with a speed of up to 400MHz plus a professional audio Codec decoder. 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 the decoding and output of audio. Its internal structure is as follows:
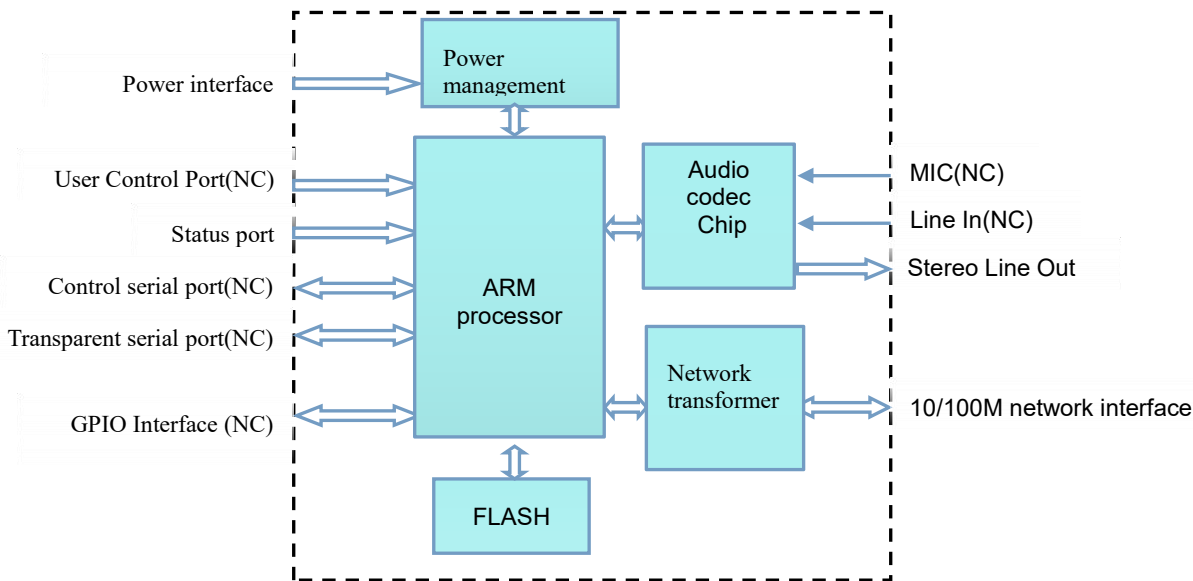
Its shape and structure are as follows:
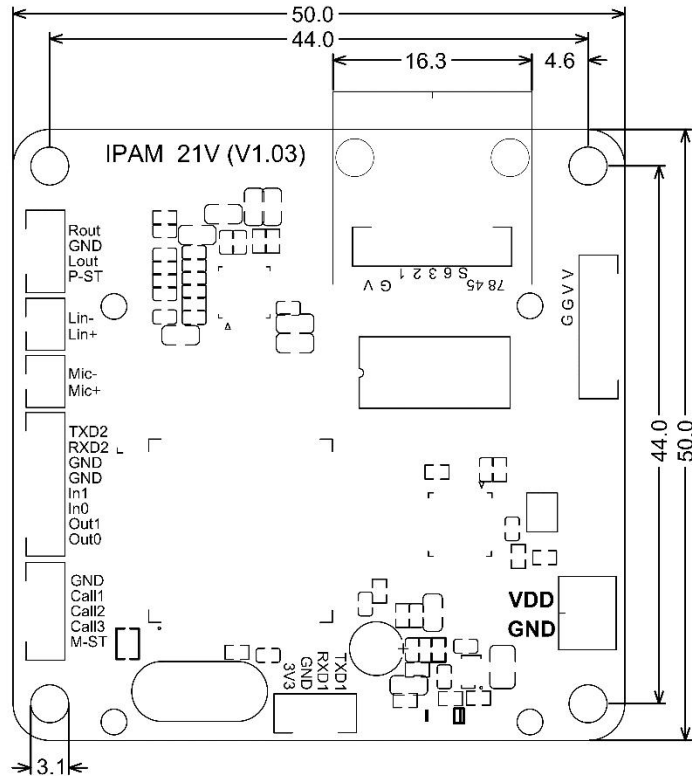
In order to be easier to use, SIP2102V uses a connector as an external interface and integrates a network interface.

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.

● Control method:

SIP2102V is a playback module that can only passively receive intercom or broadcast audio signals. Control via external devices is not supported.

● System status output:

Output status (P-ST): Output high level during playback or intercom, which can be connected to indicator lights or control relays.

● Extensions:

POE power supply:Provide POE power receiving module interface

4. Technical parameters

- Power input: Voltage DC 4.7～16V

    Maximum working current, 200mA
- Temperature: Operating temperature range Industrial grade: -20～55℃

    storage temperature range -40～85℃
- Network: 10/100M Base adaptive Ethernet interface
- Line Out output:Load 10KΩ, typical 1000Vrms, signal-to-noise ratio 95dB (in playback mode)
- 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
- Unidirectional mode: One-way intercom(reception only), with high-performance echo suppression algorithm, 8kHz sampling, G.711a/u, G.722 encoding

    The minimum delay is 80ms
- Module size: Length x width x height: 5 x 5 x 1.7cm

5. Hardware interface description

SIP2102VFor the network audio module, all external connections use terminals, the power supply uses 2.0mm connection terminals, the network uses standard RJ45 connectors, and the others are all 1.25mm connection terminals.

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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
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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, and the power supply range is DC4.7～16V. 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.

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Interface description:
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	definition	type	describe
1	VDD	P	Power input, the voltage is DC 4.7-16V

2	GND	P	power ground

2. 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.

3. 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:

	definition	type	describe
1	V	P	Power input, the voltage is DC 4.7 ~ 16V
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.

4. 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:

	definition	type	describe
1	V	P	Power input, the voltage is DC 4.7 ~ 16V
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. SIP2100V series module model list

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	intercom	multicast	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.

2. List of other compatible module models

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

	play	intercom	multicast	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.

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

	play	intercom	multicast	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

SIP2102V 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 for added playback 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.

SIP2102V 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.
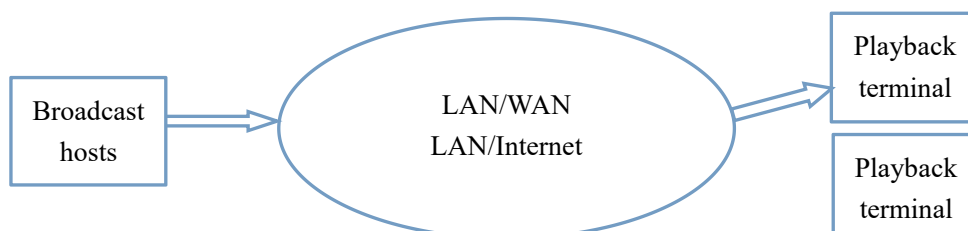
The broadcast host can use our company's broadcast system control software, broadcasting equipment, network microphone, etc., or the user can develop his own broadcast software based on our company's DLL dynamic library. Both the playback software and the playback development kit use our company's private 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.