SIP Audio Encoder
SIP-900ED

Features:

- Server-less SIP encoder work with SIP decoder
- Point to point analog sources encoding and decoding audio over IP
- BGM, general /live /scheduled & emergency announcement
- Compatible with any brand SIP device and VoIP devices.
- Use VoIP & SIP phone to make announcement to IP PA system
- Easy integration with Cisco, Avaya, Huwai & Polycom
- With one line input, microphone input and line output
- Based on ARM + DSP architecture, high-speed industrial-grade chips. Playing delay is less than 130ms
- Provide a variety of decoding MP3, WMA, WAV, etc. Supports up to 48K sampling, 192kbps streams
- Network Interface: 10 / 100Base-TX adaptive network, RJ45 port, 1.5KV electrical isolation
- 12-24V DC power supply connector (No DC power adapter included)
- With relay output to integration with third party system like emergency lamp or emergency button or fire alarm system

Description:

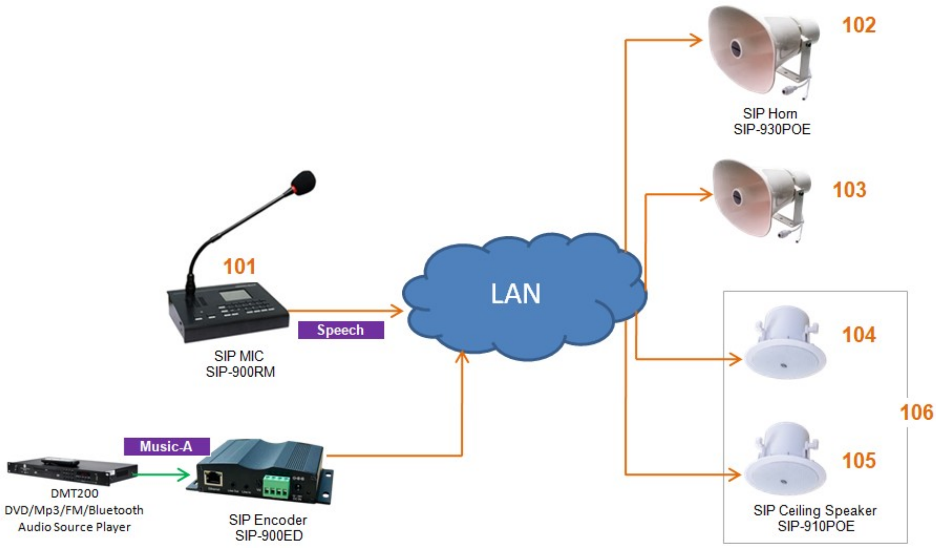
SIP Audio Encoder SIP-900ED is designed to encode analog audio sources & microphone line input into SIP signal over IP transmission. By working with SIP-900NA SIP audio decoder to set up the basic audio over IP solution, thus you can assign the analog sources to pre-set SIP decoder for general announcement, live announcement or scheduled announcement or emergency announcement. Moreover, by using the SIP encoder and SIP decoder, you can easily upgrade the existing analog public address system into audio over IP without changing the existing amplifier and loudspeakers. The SIP-900 SIP audio system compatible with any brand name IP PABX, SIP & VoIP phone system and use it to make announcements via your SIP or VoIP phone. Point to point SIP audio encoding and decoder plus paging system no need software nor SIP server. SIP server is needed for video/audio intercom, IP PABX paging and SIP voice alarm etc functions.

Specifications:

Model	SIP-900ED
Description	SIP Audio Encoder
Power Supply	12-24V DC (No DC power adapter included)
Power Consumption	<300mW
Protocol	SIP2.0, SIPS (TLS), DHCP opt. 66, SMTP, 802.1x, RTSP, RTP, SRTP, TFTP, HTTP, HTTPS, Syslog, ONVIF, TCP/IP, ARP, ICMP, DHCP, DNS, IGMP
Network Speed	10 / 100 Base-TX adaptive network, RJ45 port, 1.5KV electrical isolation
Audio Format	G.711 (PCMA, PCMU), G.729 (Annex A, B), G.722, L16 /16 kHz
S/N Ratio	≥90dB
Frequency Response	20Hz-20KHz
Connector	line input: 2000mVpp; Mic input 70～12.KHz, 50mVpp, S/N Ratio: 68dB; line output: 2200mVpp, Control input & output: closed contact or open contact.
Working Temperature	-10℃ to +65℃
Humidity	10% ~ 90%
Dimensions	130×85×30mm
Weight	0.2kg

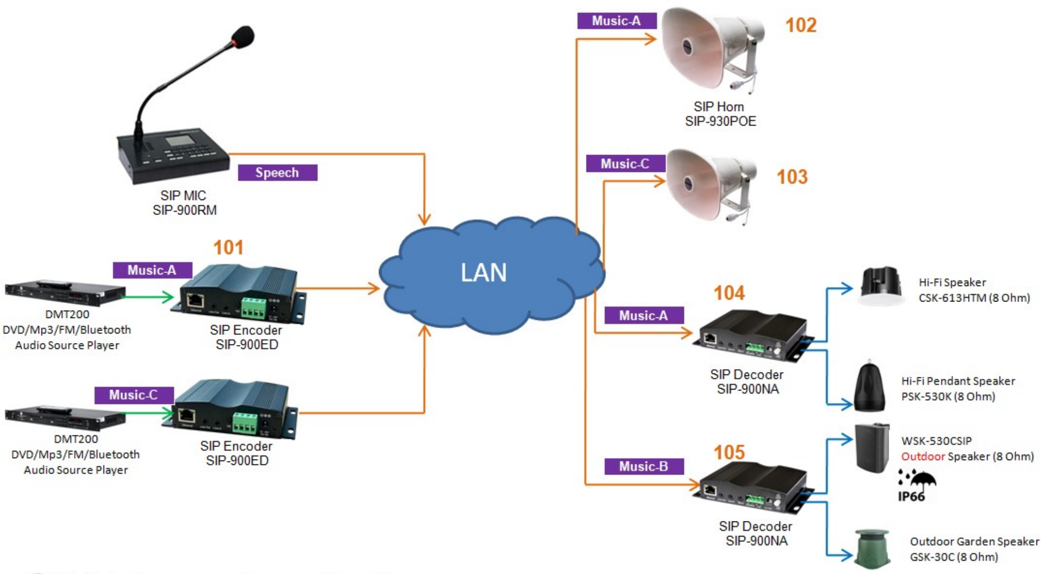
SIP Multiple Zone Paging & BGM System

（Server-less System）

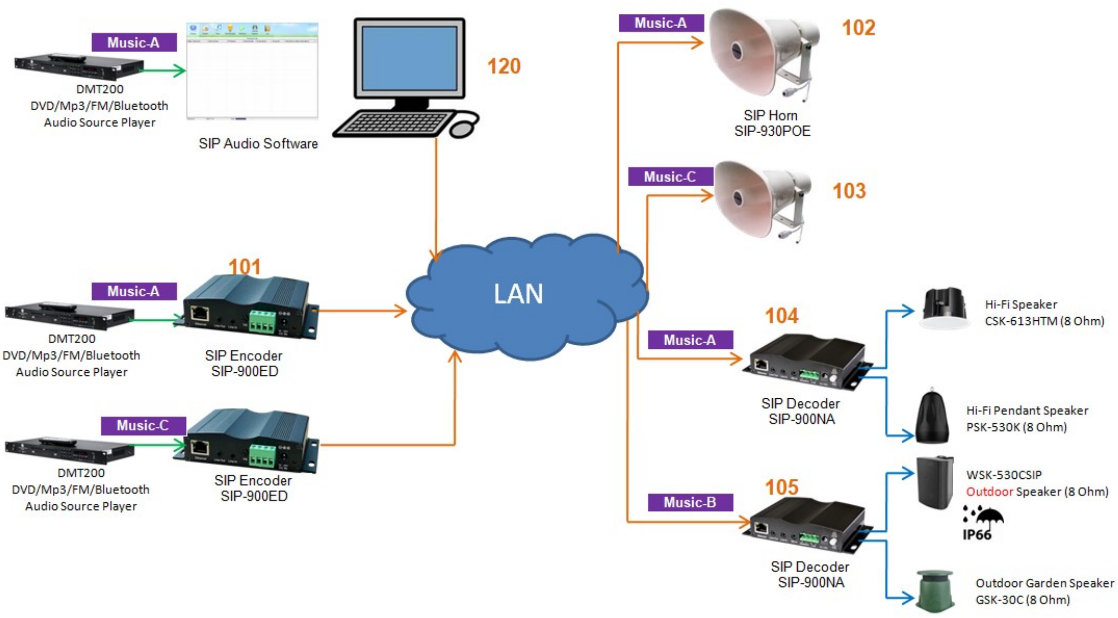


SIP Audio Encoder & Decoder

（Server-less System）



SIP Encoder & Decoder



www.cmxaudio.com, info@cmxaudio.com