Hanwha Vision

SPA-D1000
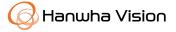## IP Audio Module

Key Features

- Output Level: +6dBV ± 3dB
- Frequency Response: 20Hz ~ 20kHz
- Amplifier: Built in 10W Class D
- Power: PoE (IEEE 802.3 af type 1 Class 3), PoE+(IEEE 802.3 at type 2 Class 4)
- Internal Memory: 1 GBytes
- Supported Protocol: IPv4, HTTP, SIP, mDNS, DNS, NTP, TCP, UDP, DHCP, ARP, ICMP
- TTS: English(US, UK), German, French, Spanish, Russian
- Audio Mornitoring: Speaker, Connection test

SPA-D1000
IP Audio Module

Specifications

Product	
Type	IP Audio Module

Line Output	
Output Level	+6dBV ± 3dB
Frequency Response	20Hz ~ 20kHz
THD + N Ratio (AES17 LPF)	Less than 0.1%
S/N Ratio (20kHz LPF, A-WTD)	Greater than 91dB

Amplifier	
Description	Built in 10W Class D

Network	
Ethernet	10/100 Base-T

Memory	
Internal Memory	1 GBytes
External Memory (Micro SD)	SDHC upto 32GB (SANDISK)

Contact	
Contact Input. Dry contact	One channel
Contact Output. Dry contact (NO)	One channel

General	
Operating Temperature	-20 ~ +50°C (-4°F ~ +122°F)
Operating Humidity	10~85% RH (Non-Condensing)
Weight	0.24Kg
Size	123(W)*80(H)*30.6(D)mm
Color	White
Certificate	EMC : KS C 9832/9835, EN 55032/55035, FCC Part 15 Subpart B, ICES-003 Safety : KC 62368-1, UL 62368-1, CAN/CSA 62368-1

Power	
PoE	PoE (IEEE 802.3 af type 1 Class 3)
PoE+	PoE+(IEEE 802.3 at type 2 Class 4)

Embedded MIC	

Input Sensitivity	Sound Level Meter
Frequency Response	Sound Level Meter

Audio

Supported Audio Format	WAV, MP3 in mono/stereo from 64 kbps to 320 kbps. Sampling rate from 16 kHz up to 48 kHz

Network Protocol

Security	Password protection : admin,setup,user,guest (sha-2, Digest authentication, User access log) Digest authentication, User access log
Supported Protocols	IPv4, HTTP, SIP, mDNS, DNS, NTP, TCP, UDP, DHCP, ARP, ICMP

System Integration

API (Application Programming Interface)	SUNAPI
Multi-source Dynamic PA control	Multi-source up to 48 (Multicast) (Audio 24CH + Mic 24CH) Up to 256 Zone Control (Multicast) Up to 20 Zone Control (Unicast) Up to 255 Groups
VoIP	Tested with SIP clients such as Grandstream, Yealink, Cisco, MicroSIP softphone and PBX supplier such as Asterisk Tested with SIP client such as Grandstream and PBX supplier such as Grandstream Supported SIP features: DTMF (RFC2833) Supported codecs: PCMU, PCMA, speex/8000, speex/16000
TTS	English (US, UK), German, French, Spanish, Russian
Event & Preset	Virtual Contact, Dry contact
Functional Monitoring	Connection verification, Built-in system logging