MAS-A100

A hands-free lecture capture and presentation solution

Overview

Beamforming ceiling microphone with advanced speech reinforcement technology—enabling truly hands-free lectures and presentations

The MAS-A100 is a state-of-the-art speech reinforcement* and recording solution. Attaching and positioning lavalier microphones is time-consuming, and handheld microphones restrict movement. However, the Beamforming Microphone is simply affixed to the ceiling, then ready to use—leaving the orator location-free, hassle-free and hands-free to focus on presenting.

Our comprehensive solution employs audio volume stabilization and clear audio processing to produce sharp recordings. The MAS-A100 has two channel outputs. The dual-output channel enables speech reinforcement and recording simultaneously. The dedicated rec channel records with a wider capture range and optimal intelligibility—perfect for lecture capture.

Supporting Dante and PoE, MAS-A100 links with third-party systems via a single cable connection. Auto-calibration streamlines system configuration, and settings can easily be adjusted via the network using free MASM-1 management software. With simple settings, central management and no need to monitor battery use, MAS-A100 is a convenient classroom audio solution.

*A loudspeaker system is required for speech reinforcement

Features

Beamforming and Intelligent Feedback Reducer

The integration of beamforming technology and our market-leading Intelligent Feedback Reducer produces clear audio with hands-free speech reinforcement, leaving the presenter hands-free. Sony's high-performance digital signal processing and unique algorithms extract speech sound while suppressing unwanted feedback (howling).

Noise reduction

Our Beamforming Microphone incorporates advanced noise-reduction technologies to minimize ambient noise. Stationary background noise arising from projectors and air conditioning are automatically detected and reduced to achieve clear sound resulting in better intelligibility and, ultimately, better understanding for listeners.

In addition to this, the Ambient Noise Filter* feature automatically detects and filters out intermittent and distracting sounds, such as the typing of a keyboard, click of a mouse or the turning of pages. (Version 1.1)

*The Ambient Noise Filter feature is scheduled to be available in March 2021.

Automatic gain control

The MAS-A100 has a useful inbuilt volume normalization feature to maintain stable volume regardless of the distance between the presenter and the microphone, leaving the presenter free to move around the space without the audio quality deteriorating. Automatic gain control adjusts the strength of the signal received to ensure clear and consistent sound for easy listening.

Automatic calibration

The Beamforming Microphone optimizes the parameters of audio processing for speech reinforcement automatically. The microphone generates a test signal, which is captured by the loud speakers. The system then calculates and configures the parameters for speech. This calibration is required only on initial installation, minimizing time-consuming set-up and system maintenance.

Bass Boost

Enhance audio quality for clear and easy listening with adjustable levels of bass boost. (Version 1.1)

Scheduled to be available in March 2021.

Dante and PoE compatibility

The MAS-A100 is compatible with third-party Dante mixers, converters and other devices, as well as power over Ethernet (PoE). There's no complicated wiring—a single cable can connect the microphone to the system.

Status LED

The microphone has an inbuilt status LED, enabling both instructor and operator to easily recognize the microphone status.

API

Our Beamforming Microphone can be operated by API, permitting simple external control and customization, as well as linking with existing systems.

Specifications

Networking	
Cable requirements	Cat5e UTP or higher

SONY

Audio	
Frequency response	100-10,000 Hz *
Sensitivity	0dBFS/Pa at 1 kHz*
Maximum SPL	94dBSPL
Signal to noise ratio	75 dB (A-weighted, 1 kHz, 1 Pa)*
Latency	Under 24 ms (not including Dante latency)
Self noise	19dBSPL (A-weighted)*
Dynamic range	75 dB*
Dante digital output	Channel count: 2 channels (Main, Rec) with Dante Sampling rate: 48 kHz Bit depth: 24 Audio interface: Dante, AES67

Digital Signal Processing	
Audio technologies	Beamforming Feedback reducer Automatic gain control Noise reduction Noise gate Automatic calibration with test signal

	Equalizer API for external control system
Performance	Maximum noise reduction: 24 dB Maximum feedback reduction: 32 dB
Configuration	Mic gain (output volume): -60 dB to +12 dB Capture range: Narrow / standard / wide Feedback reduction: Low/middle/high Noise reduction: Low/middle/high Noise gate: Off/low/middle/high Equalizer: Fixed 5-band, ±12 dB (1 dB Step)

General

Connector type	RJ45
Power requirements	Power over Ethernet (PoE), Class 0
Power consumption	13 W, maximum 9 W @25°C, typical
Weight	Approx. 2 lb 3.2 oz (1 kg) (main unit only) Approx. 2 lb 14 oz (1.3 kg) (with ceiling bracket)

Dimensions	Φ: Approx. 240 mm (9 1/2 inch) Height: Approx. 50 mm (2 inch)
Control browser (MAS-A100)	Google Chrome
Operating temperature range	0–40°C
Operating humidity	20% to 80% (no condensation allowed)
Storage temperature	-20 °C to +60 °C (-4 °F to 140 °F)
Storage humidity	20% to 80% (no condensation allowed)
Supplied accessories	Ceiling bracket (1) Safety regulations (3) Template (1)

Notes	
*	Not including signal processing

Gallery