

Microphone Array System

System Integration Guide

Version 1.10

MAS-A100

MASM-1

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1. Introduction

This document describes the design of a system that uses MAS-A100 Beamforming Microphones and an MASM-1 Microphone Array System Manager, and the required information for setup and operation. Refer to the MAS-A100 Operating Instructions and the MASM-1 Help Guide.

The MAS-A100 Beamforming Microphone is a beamforming type, ceiling-mounted microphone with a speech reinforcement and clear audio recording system for hands-free, hassle-free operation covering locations over a wide area.

Lecturers and presenters can use both hands freely when giving lectures and presentations, without troublesome preparation and equipment operation.

The unit also supports Dante audio transmission and PoE (Power Over Ethernet) power supply. It can be connected to various Dante-compatible audio systems using a single network cable.

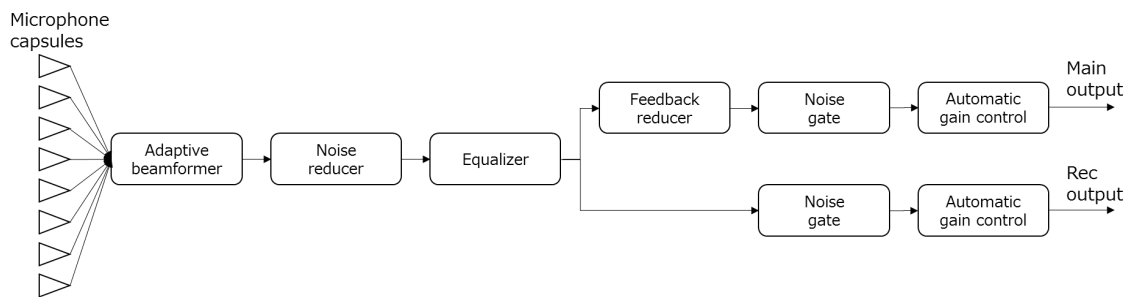
The MASM-1 Microphone Array System Manager is Windows application software for configuring and operating an array of MAS-A100 Beamforming Microphones collectively by remote control via a network.

2. MAS-A100 Overview

This section provides an overview of the MAS-A100 Beamforming Microphone.

Audio capture function

The MAS-A100 integrates eight directional microphone capsules and the signal processing functions, as shown in the following diagram. It enables audio capture features for speech reinforcement and for recording, corresponding to the Main output and the Rec output, respectively.



The Main output exploits a unique feedback reduction technique, and can be connected to a speech reinforcement system and played back by loudspeakers in a room. The Rec output is optimized for recording of clearer audio over a wider area.

Audio output function

The MAS-A100 is compatible with the Dante system and achieves long-distance transmission of low-latency, high-quality, digital audio. Main and Rec (2) outputs are multiplexed and output using a single Cat 5e LAN cable. Also, the power supply for the MAS-A100 is supplied by the same cable using PoE.

Setup and management function

The MAS-A100 has two IP addresses. One IP address is for output of the captured audio using the Dante system. The other IP address is for MAS-A100 setup using the web app or external API.

MASM-1 and Dante Controller software by Audinate (referred to as "Dante Controller") on a PC connected to the same segment (same layer 2 network) are used to initialize the MAS-A100 units and route Dante audio, respectively.

3. System Configuration Example

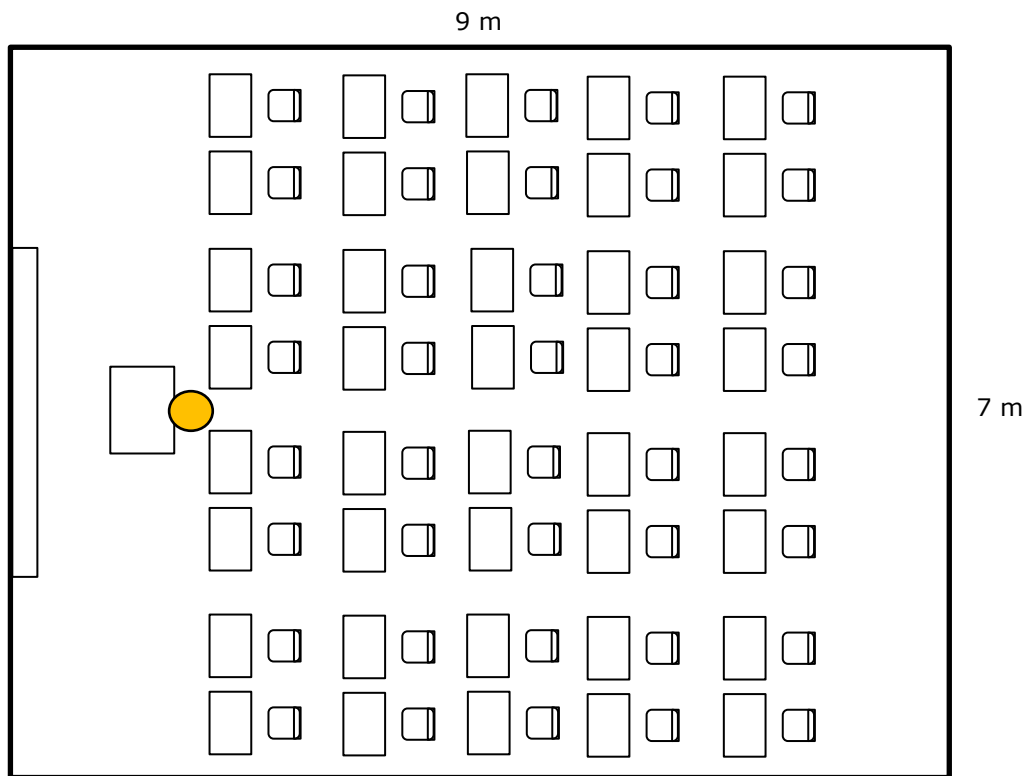
This section shows an example system configuration using the MAS-A100.

3.1. Recording a lecture in a small room (no speech reinforcement)

The following system configuration shows the unit installed for recording a lecture in a small room with a width of 7 m, a depth of 9 m, and a ceiling height of about 3 m. One MAS-A100 unit mounted on the ceiling in the center of the room above the lecturer is used to capture audio.

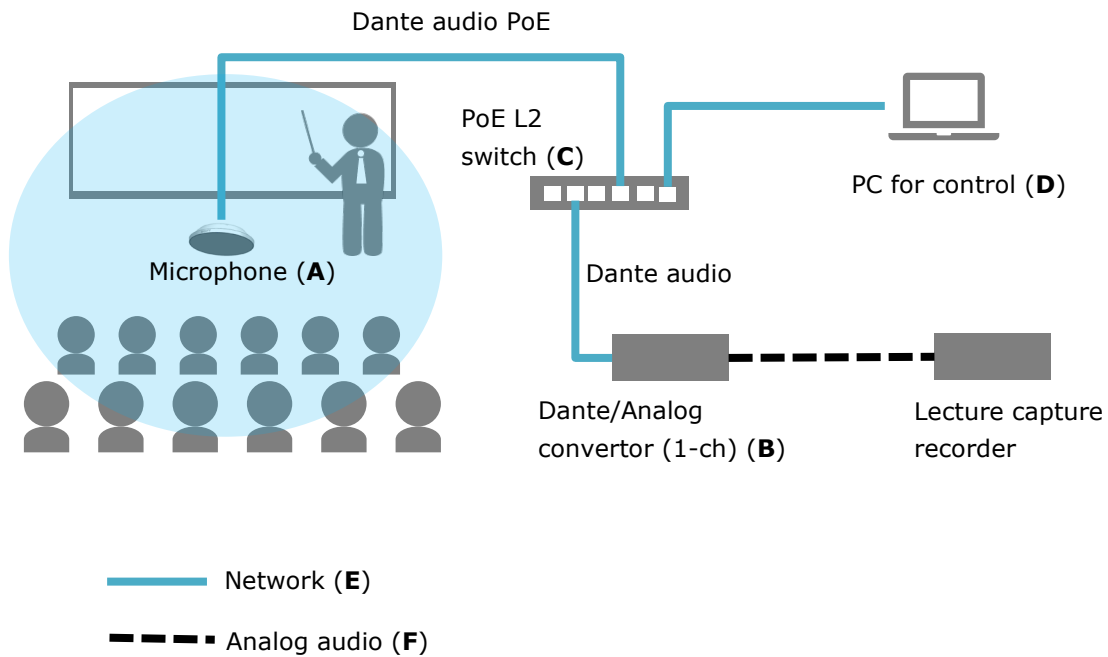
The capture range of the Rec output is a hemisphere of 7 m radius (14 m diameter), and one unit can cover the recording of the lecturer and audience. The Rec output of the MAS-A100 connects to the recording system via a Dante/Analog converter (Audinate).

Room floor plan



 : MAS-A100

Connection diagram



Required equipment

Label in figure above	Name	Qty.	Remarks
A	MAS-A100	1	
B	Dante DAC ADP-DAO-AU-0X1	1	Audinate
C	PoE HUB GS108PEv3	1	NETGEAR
D	Windows PC	1	For setup
E	LAN cable (Cat 5e or higher)	As required	
F	Audio cable	1	

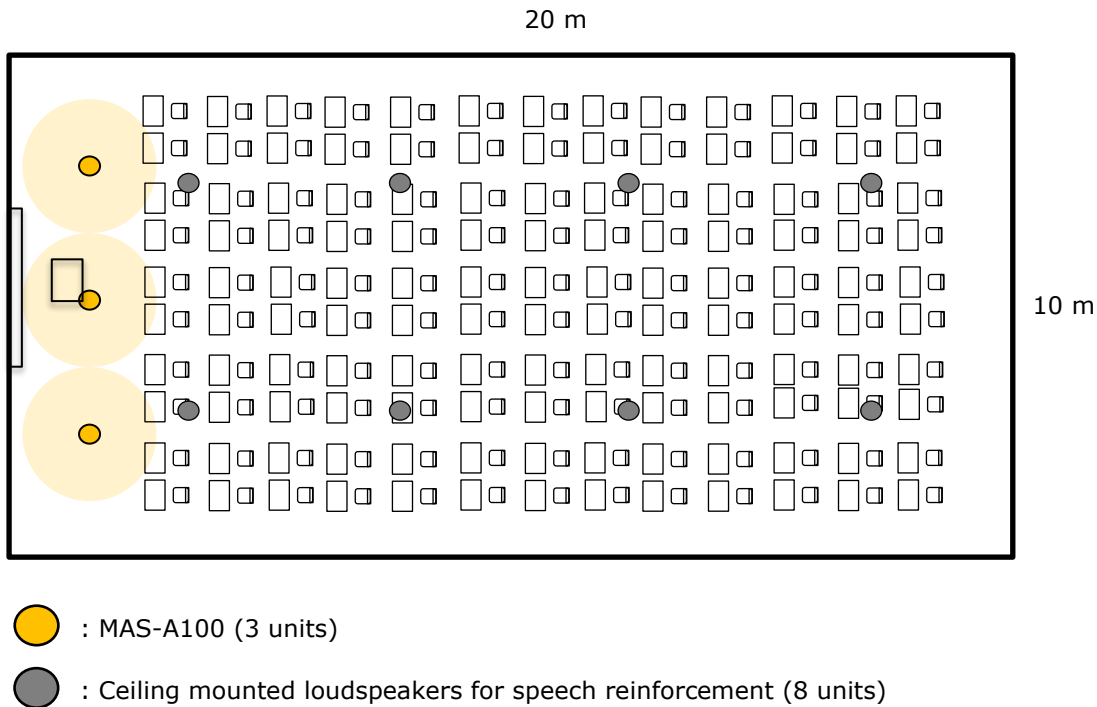
3.2. Reinforcing/capturing a lecture in a medium-sized room

The following system configuration shows units installed for reinforcing speech and recording a lecture in a medium-sized room with a width of 10 m, a depth of 20 m, and a ceiling height of about 3 m. Three MAS-A100 units mounted on the ceiling at intervals of 2.7 m above the lecturer are used to reinforce speech and capture audio.

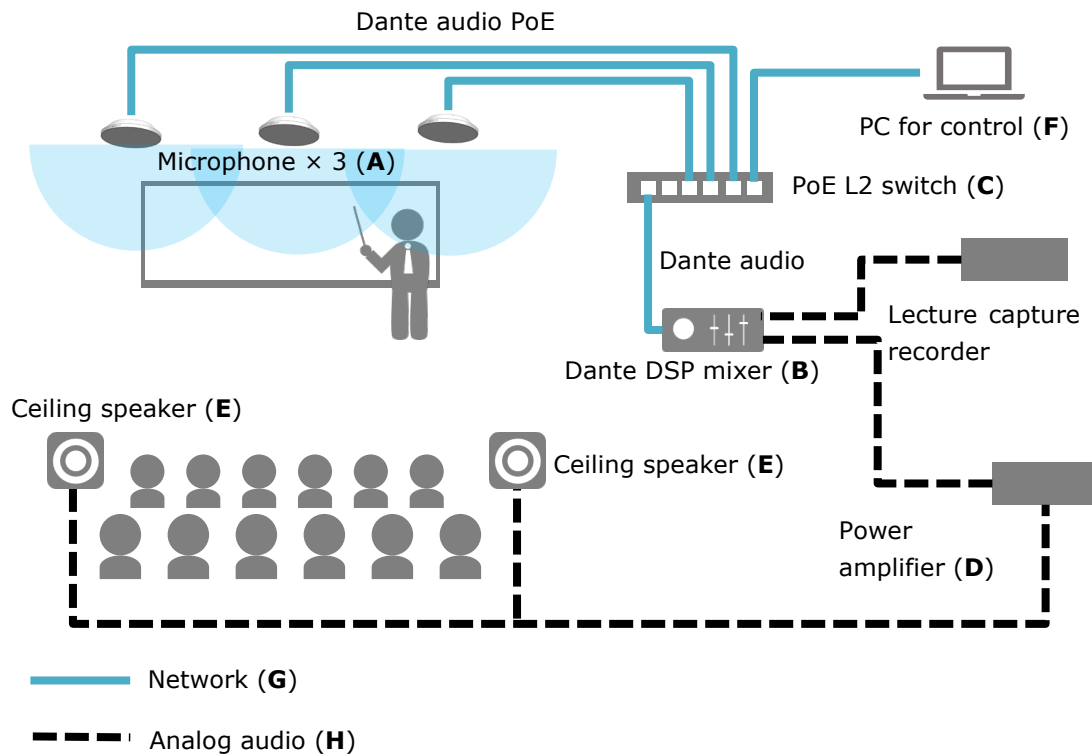
The Main output and the Rec output of these MAS-A100 units are mixed by a Dante DSP mixer and connected to a speech reinforcement system and a recording system, respectively.

The speech reinforcement quality and volume can be increased by not installing loudspeakers near the front of the room.

Room floor plan



Connection diagram



Required equipment

Label in figure above	Name	Qty.	Remarks
A	MAS-A100	3	
B	Dante Mixer Prism 4x4	1	Symetrix
C	PoE HUB GS108PEv3	1	NETGEAR
D	Power amplifier	1	
E	Ceiling speaker	8	
F	Windows PC	1	For setup
G	LAN cable (Cat 5e or higher)	As required	
H	Audio cable	As required	

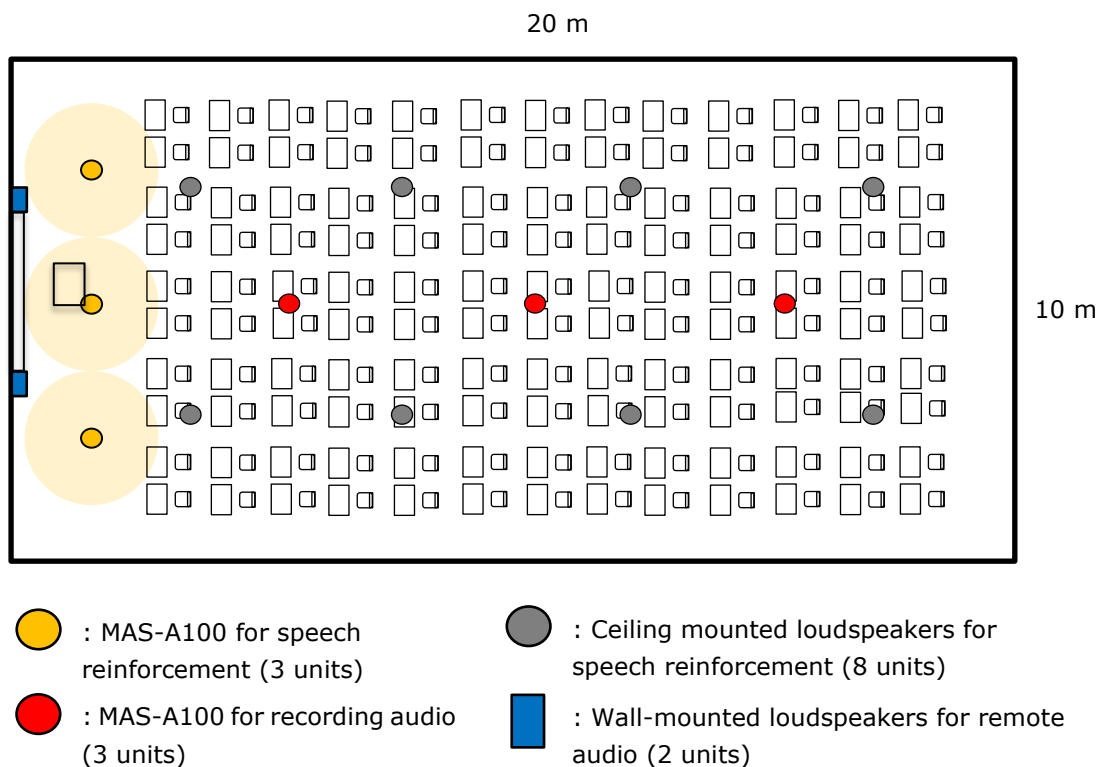
3.3. Remote lecture

The following system configuration shows units installed for reinforcing speech and recording a lecture with a remote classroom, using bidirectional communication, in a medium-sized room with a width of 10 m, a depth of 20 m, and a ceiling height of

about 3 m. Three MAS-A100 units mounted on the ceiling at intervals of 2.7 m above the lecturer are used to reinforce speech and record audio. Three MAS-A100 units mounted on the ceiling above the audience are used to record audio. The recorded audio is sent to the remote room.

The Main output and the Rec output of these MAS-A100 units are mixed by a Dante DSP mixer and connected to a speech reinforcement system and a remote conferencing system (PCS-XG100), respectively. In order to improve the speech reinforcement performance, only the remote sound is output from the loudspeakers close to the MAS-A100 units near the instructor, and both the reinforced speech of the lecturer and the remote sound are output from the loudspeakers on the audience side.

Room floor plan



The diagram illustrates a Dante-based audio system for a lecture hall, showing the following components and connections:

- Components:**
 - (A) Microphone x 3:** Two sets of three microphones, one for the presenter and one for the audience.
 - (B) Teleconference system:** Connected to the Dante DSP mixer via remote audio.
 - (C) Dante DSP mixer:** The central processing unit for the Dante audio network.
 - (D) PoE L2 switch:** Provides power and network connectivity to the microphones and the DSP mixer.
 - (E) Power amplifier:** Amplifies the audio signals for the speakers.
 - (F) Loudspeaker:** Two main loudspeakers for the room.
 - (G) Ceiling speaker:** Two ceiling-mounted speakers.
 - (H) PC for control:** A laptop connected to the PoE L2 switch for network management.
- Connections:**
 - Network (I):** Solid blue lines represent the Dante audio network, connecting the microphones, DSP mixer, and PoE switch.
 - Analog audio (J):** Dashed black lines represent analog audio paths from the DSP mixer to the power amplifier and then to the loudspeakers and ceiling speakers.
 - Remote audio (bidirectional):** Blue double-headed arrows represent the bidirectional audio link between the DSP mixer and the teleconference system.
 - Dante audio PoE:** A label at the top indicates the power and data lines for the microphones.

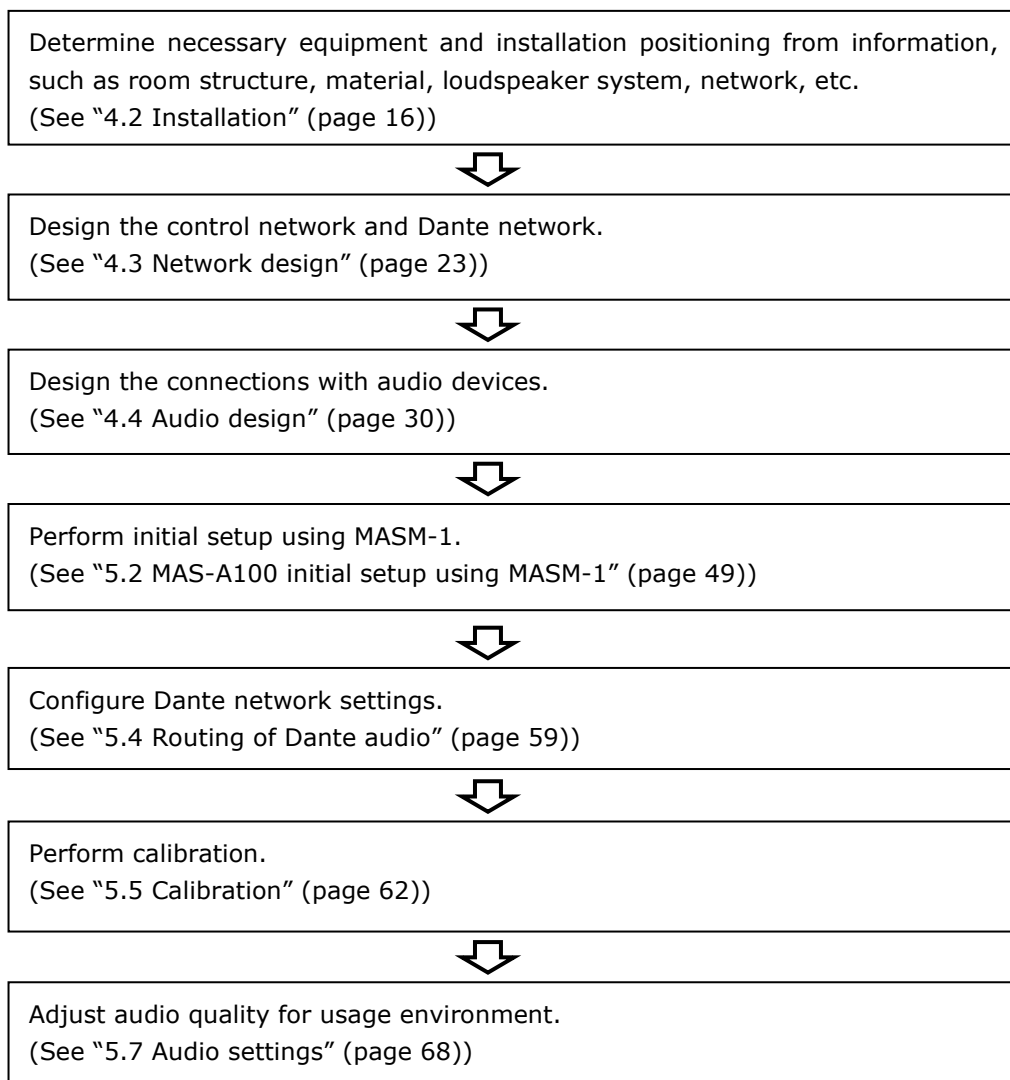
Label in figure above	Name	Qty.	Remarks
A	MAS-A100	6	3 units each for speech reinforcement and audio capture
B	PCS-XG100	1	
C	Dante Mixer Prism 4x4	1	Symetrix
D	PoE HUB SG350-28P	1	Cisco
E	Power amplifier	2	
F	Loudspeaker	2	
G	Ceiling speaker	8	
H	Windows PC	1	For setup

Label in figure above	Name	Qty.	Remarks
I	LAN cable (Cat 5e or higher)	As required	
J	Audio cable	As required	

4. System Design

4.1. System design flow

The following shows the procedure for MAS-A100 system design, installation, connection, and configuration.



4.2. Installation

MAS-A100 installation

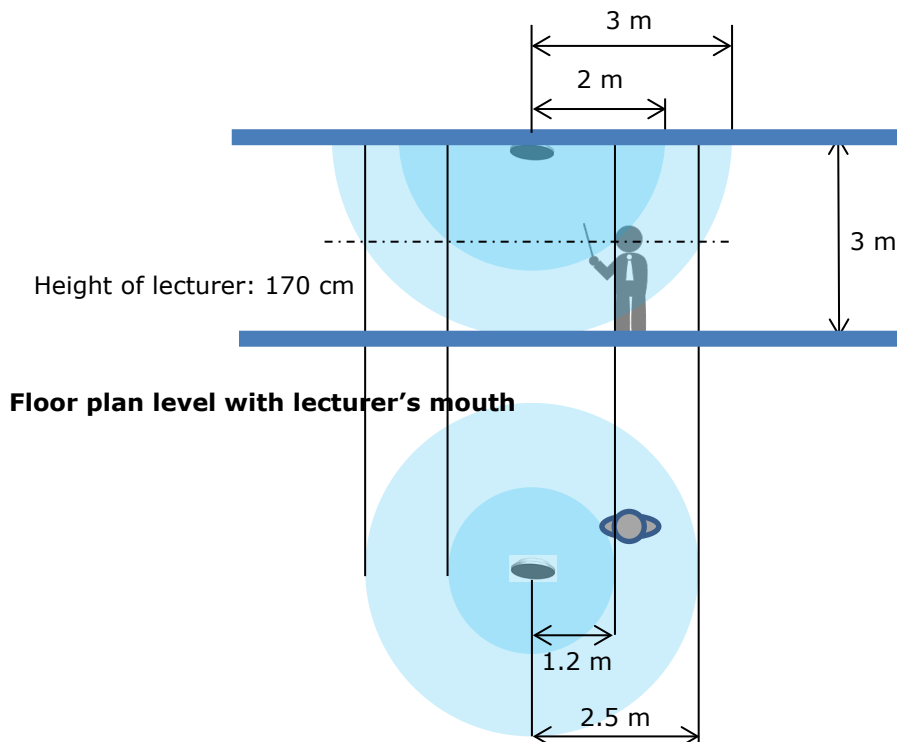
When using speech reinforcement loudspeakers (Main output)

The audio capture range of an MAS-A100 for speech reinforcement is a hemisphere of 2 to 3 m radius (4 to 6 m diameter) with the MAS-A100 at the center. Consider the height of the ceiling, the posture of the lecturer (sitting or standing), and the position of the lecturer's mouth, then design the installation so that the lecturer is within the audio capture range.

It is recommended that the MAS-A100 be within 2 m distance from the lecturer.

Audio capture range for speech reinforcement

Side view



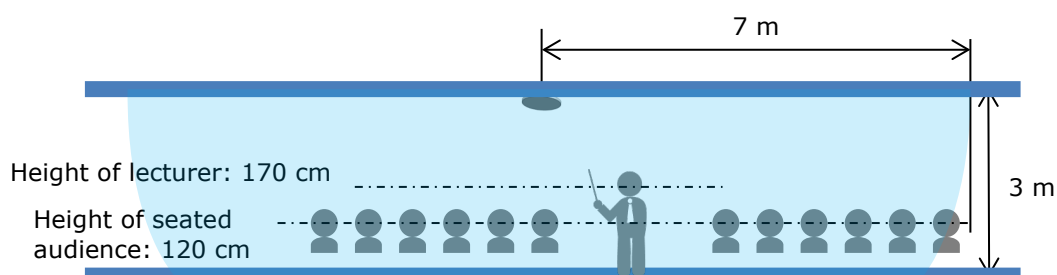
When using recording output (Rec output)

The audio capture range of an MAS-A100 for recording is a hemisphere of 7 m radius (14 m diameter). Consider the height of the ceiling, the posture of the audience

(standing or sitting), and the position of the audience's mouths, then design the installation so that the audience is within the audio capture range.

Capture range when recording

Front view

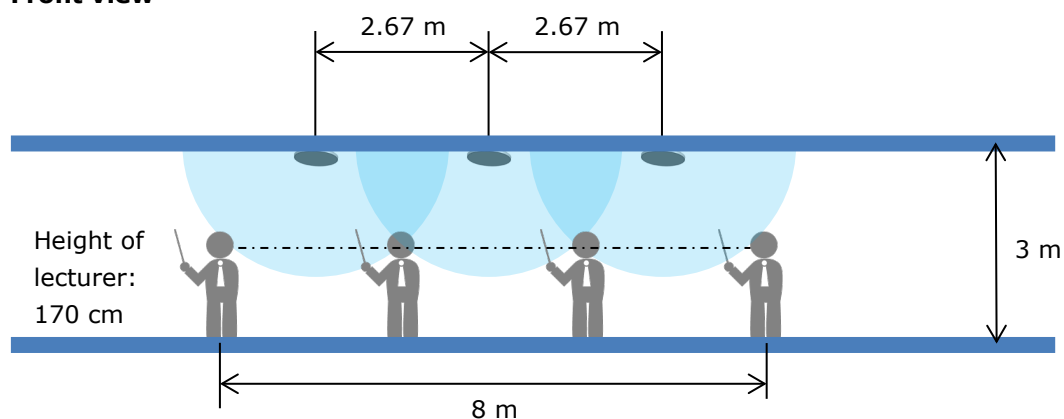


Multiple unit installation

To capture a wider area, install multiple MAS-A100 units separated at fixed intervals, and mix the output of the units at the same volume. In such an installation, the MAS-A100 units capture audio with less noise, so sound quality abnormalities due to interference between the MAS-A100 units are less likely to occur, making the audio capture clearer.

Multiple MAS-A100 installation example

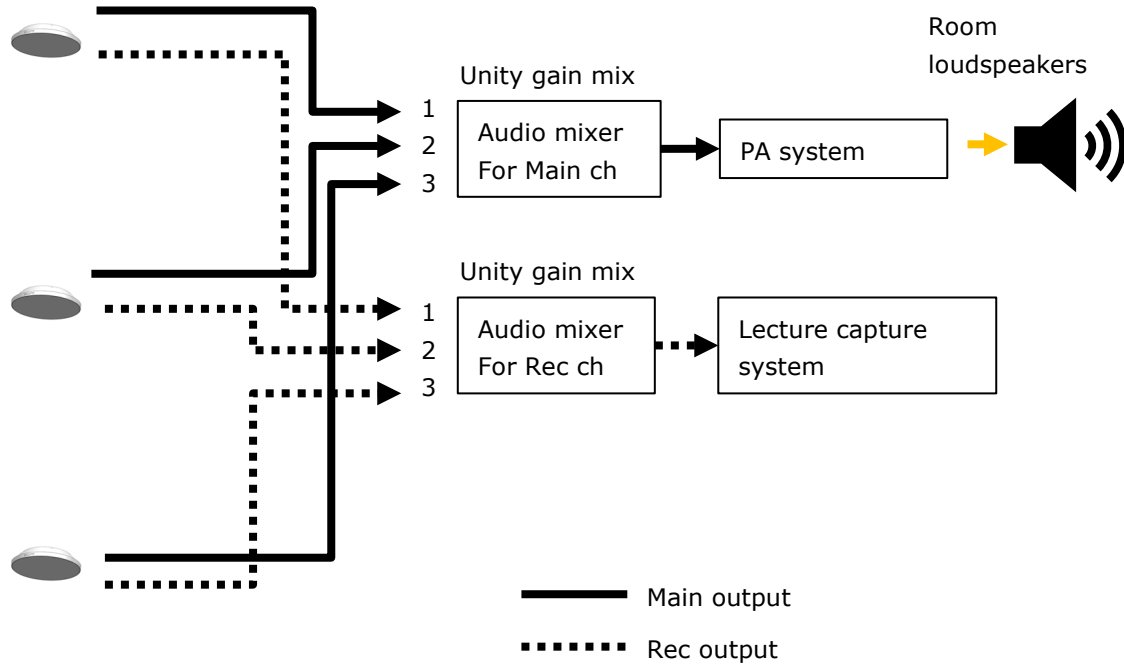
Front view



Make sure to mix the audio channels of the Main output and audio channels of Rec output separately.

Do not mix the Main output and Rec output together. If the Main output is connected to a mixer that has an automatic gain control function, the system will not work properly.

Multiple MAS-A100 units connection diagram



Speech reinforcement level adjustment

By exploiting the beamforming function and feedback reduction technique, the Main output of an MAS-A100 unit can be connected to a speech reinforcement system and played back to achieve hands-free speech reinforcement.

When using the Main output of the MAS-A100, a test signal generated by the MAS-A100 is output from the loudspeakers and is used to optimize the mixer settings, power amplifier settings, and acoustic positional relationship with the speech reinforcement loudspeakers. The audio captured by the MAS-A100 is used for calibration. For details about the calibration process, see "5.5 Calibration" (page 62). After calibration is finished, take care not to change the mixer settings (gain, equalizer), power amplifier settings (gain), and loudspeaker position or orientation. To adjust the speech reinforcement level, adjust the volume using the speech reinforcement adjustment function of the MAS-A100, not by changing the gain of the mixer or amplifier.

Installation environment for speech reinforcement

The speech reinforcement performance of the MAS-A100 is affected by the quality of the loudspeaker, installation location, shape of the room in which it is installed, materials of the room, and ambient noise environment.

This section describes the recommended installation environment for the MAS-A100.

Speech reinforcement loudspeakers

Use speech reinforcement loudspeakers that have sufficient output margin. Proper speech reinforcement cannot be achieved if distortion, crackling, or chattering occurs. Due to problems with volume, the use of small loudspeakers for computers or built-in loudspeakers, such as in TVs, radio cassette recorders, or karaoke players, are not recommended.

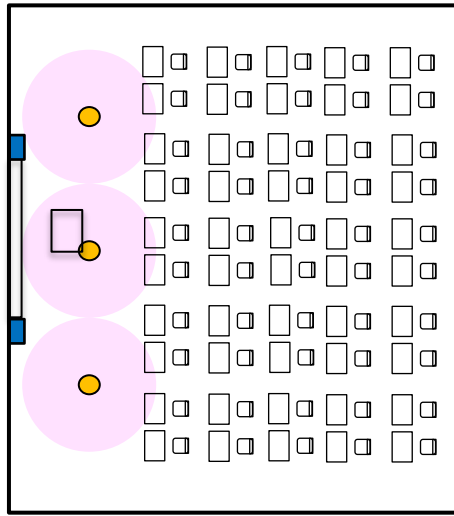
Install speech reinforcement loudspeakers such that their directivity field does not point towards the MAS-A100, and with a separation of 2 m or more from the MAS-A100. Installing 3 m or further away is recommended.

When installing MAS-A100 units on the ceiling above a classroom dais, it is recommended that the loudspeakers be placed close to the audience.

By using loudspeakers with narrow directivity, such as line array loudspeakers, you can increase the sound pressure on the audience side and decrease the sound pressure returning from the loudspeakers to the MAS-A100, allowing you to increase the loudspeaker volume.

If the loudspeaker installation position cannot be separated from the MAS-A100, the sound quality will deteriorate because of the narrower audio capture range or the application of strong feedback control, although this will suppress the occurrence of feedback.

Example of loudspeakers in non-recommended locations



● : MAS-A100 for speech reinforcement (3 units)

● : Loudspeakers non-recommended locations (2 m radius)

Room acoustic characteristics

It is recommended to use a room with low reverberation and low reflection acoustic characteristics. In an environment with reverberation or reflection, the intelligibility of the sound may be poor and the sound quality may make hearing difficult.

The upper limit of speech reinforcement may be reduced in the following cases if audio re-enters the MAS-A100 with little attenuation from the loudspeakers.

- Small rooms (e.g., smaller than 5 m × 5 m)
- Rooms with reflective interiors (glass, tile, or stone walls and floors)
- Rooms with high-reflectivity ceilings

Measures to prevent general feedback (use of sound absorbing material, curtains, diffuser on flat surfaces, etc.) are effective.

Noise

The MAS-A100 has two functions to reduce noise.

One is a noise reduction function that reduces continuously occurring noise. This function makes fan sound from projectors and air conditioners, and sources that generate continuous sine wave sounds less likely to be captured.

The other is an ambient noise filter function that reduces sporadically occurring noise. This function makes keyboard type sounds and paper rustling sounds less likely to be captured.

Both functions cannot be used in environments where there is noise louder than the voice of the speaker requiring reinforcement. There are limits on the ability of the functions to reduce noise. It is recommended that units be installed as far as possible from sources of noise.

When not installing on a ceiling, use cushioning material or sound absorbing material so that the sound of footsteps and desk noises are not captured.

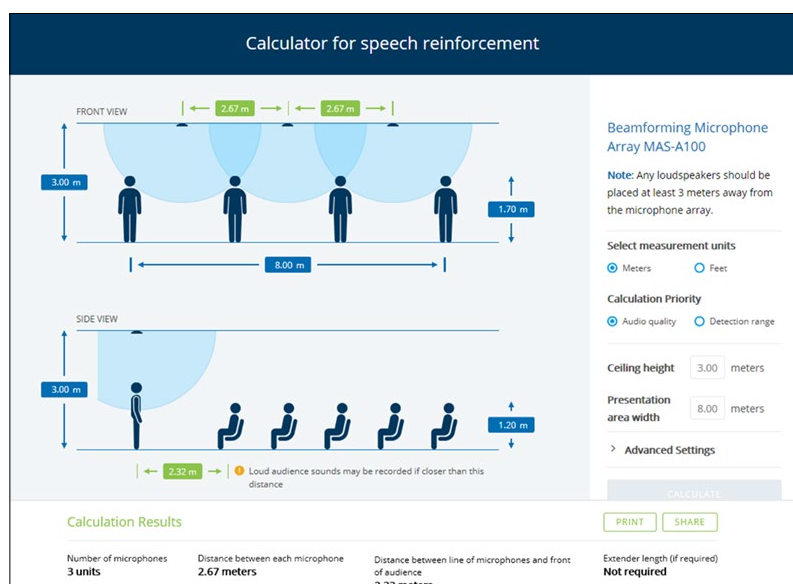
You can configure the capture range on the MAS-A100 setup screen. Noise can be reduced by reducing the audio capture range of MAS-A100 units that are close to noise sources to match the movement of the lecturer.

Calculator for speech reinforcement

The “Calculator for speech reinforcement” web site can be used to calculate the coverage of the MAS-A100 units and the intervals between multiple units so that the mouth of a lecturer walking around the lecturer’s dais is always within the radius of the coverage.

Visit the following web site.

<https://www.sonycreativesoftware.com/mas-a100/calculator>



Installation environment for audio capture

The audio capture performance of the MAS-A100 is affected by the use of the speech reinforcement function, installation location, shape of the room in which it is installed, materials of the room, and ambient noise environment. This section describes the recommended installation environment for the MAS-A100 for audio capture.

Relationship with speech reinforcement function

To use only the audio capture function, calibration is not required and you can use the unit immediately after initial setup.

To use the speech reinforcement output and audio capture output at the same time, the operation will be affected by the calibration of the speech reinforcement function. If the results of calibration for speech reinforcement are poor, it may affect the sound quality of the audio capture function.

When the results of calibration for speech reinforcement are poor, use separate MAS-A100 units for audio capture and speech reinforcement.

If the installation location is changed or if the intended use is changed, such as from speech reinforcement to audio capture, reset the unit to factory defaults (see "6.3 Factory reset" (page 81)) and perform the initial setup to initialize the calibration state.

Noise

The MAS-A100 has two functions to reduce noise.

One is a noise reduction function that reduces continuously occurring noise. This function makes fan sound from projectors and air conditioners, and sources that generate continuous sine wave sounds less likely to be captured.

The other is an ambient noise filter function that reduces sporadically occurring noise. This function makes keyboard type sounds and paper rustling sounds less likely to be captured.

Both functions cannot be used in environments where there is noise louder than the voice of the speaker requiring reinforcement. There are limits on the ability of the functions to reduce noise. The audio capture output function can capture audio over a wider range than for the speech reinforcement function. It is recommended that units be installed as far as possible from sources of noise. Depending on the installation location, noise from corridors and air conditioning ducts, etc. may be captured.

When not installing on a ceiling, use cushioning material or sound absorbing material so that the sound of footsteps and desk noises are not captured.

You can configure the capture range on the MAS-A100 setup screen. Noise can be reduced by reducing the audio capture range of MAS-A100 units that are close to noise sources to match the movement of the lecturer.

Room acoustic characteristics

It is recommended to use a room with low reverberation and low reflection acoustic characteristics. In an environment with reverberation or reflection, the intelligibility of the sound may be poor and the sound quality may make hearing difficult.

Measures to prevent general audio issues (use of sound absorbing material, curtains, diffuser on flat surfaces, etc.) are effective.

4.3. Network design

This section describes the design of networks for use with the MAS-A100 system.

It describes the network function of the MAS-A100, power supply using PoE, and the specifications required for the network switch.

MAS-A100 network function overview

The MAS-A100 connects to a network using a single LAN cable (Cat 5e or higher), but internally it has two network interfaces.

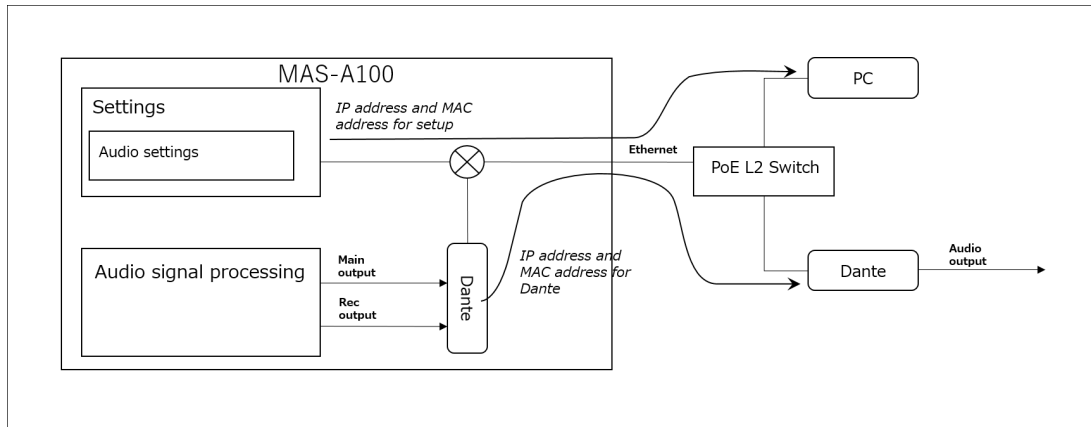
Each network interface has a separate MAC address.

Also, the IP address must be different for each network interface.

The name and functions of the two network interfaces are shown below.

Network interface	MAC address	IP address	Function
Network interface for setup	MAC address for setup	IP address for setup IPv6 address for setup	MAS-A100 setup and operation
Network interface for Dante	MAC address for Dante	IP address for Dante	Dante audio output and setup

MAS-A100 network system structure



IP address design

Operation on an IPv4 network is recommended. Two IP addresses (an IP address for setup and an IP address for Dante) must be assigned to each MAS-A100 unit.

There are three methods supported for obtaining an IP address. To prevent conflicts, it is recommended that the same method be used for both network interfaces.

Also, when installing equipment, it is recommended that addresses be set manually for ease of troubleshooting.

IP address assignment method

Name	Address range	Assignment method	Remarks
Manual	Arbitrary	MASM-1	Configured on device search screen. (See "MAS-A100 discovery and IP address setting" (page 50))
		Dante Controller	Configured on Device View screen. (See "5.4 Routing of Dante audio" (page 59))
DHCP	Arbitrary	DHCP server	Default
AutoIP	169.254.0.0 to 169.254.255.255	Automatic	Use if a DHCP server is not found when DHCP was selected.

Configure unique addresses for the IP address for setup and the IP address for Dante. Configure devices such that all devices have a unique IP address within the same segment (same layer 2 network).

If the default gateway is configured correctly, the IP address for setup can communicate across routers and can be used for remote management using MASM-1. However, when using AutoIP, communication across routers is not possible. The network interface for setup can also operate using IPv6, but in most cases there is no need to use it. To prevent inadvertent access, it is recommended to disable IPv6 routing on the router of the network to which the MAS-A100 is connected.

Dante does not support communication across routers or IPv6.

Setup network communications protocols and ports

The following ports on the setup network interface are open.

Allow access to the following ports as necessary, such as for remote management by communicating across routers.

The network for setup supports HTTPS, but the MASM-1 does not support HTTPS. It is recommended to operate using HTTP on a secure network isolated from the Internet. API Control, using the external control API, does not have a security function. When using API Control, it is recommended to operate on a secure network isolated from the Internet.

Application	Protocol	Port
Web UI HTTP	TCP	80 (can be modified)
Web UI HTTPS	TCP	443 (can be modified)
API Control	TCP	42382
Device discovery	UDP	42380

Dante network communications protocols and ports

The following ports on the Dante network interface are used.

Blocking the above network protocols and ports may disrupt the transmission of Dante audio. Be sure not to block the network protocols and ports using functions of a network switch, such as "Block unknown multicast address" or Internet Group Management Protocol (IGMP) snooping.

For details, visit the following Audinate website.

<https://www.audinate.com/faq/which-network-ports-does-dante-use>

Application	Protocol	Remote port	Remote IP
PTP	UDP	319, 320	224.0.1.129 to 132
mDNS	UDP	5353	224.0.0.251
Dante control and monitoring	UDP	8700 to 8708	224.0.0.230 to 232
Dante Multicast audio	UDP	4321	239.255.0.0/16
Dante Unicast audio	UDP	14336 to 14600	Unicast destination

PoE

The MAS-A100 must be powered by a PoE class 0 equivalent power supply function that complies with IEEE 802.3af. The MAS-A100 power consumption is 9 W (typ) and 13 W (max). When a PoE hub connects to multiple MAS-A100 units or other PoE-compatible devices, be sure not to exceed the PoE power budget of the PoE hub. Also, connect the frame ground of the PoE hub to the ground (Earth) connection of the building. PoE power may not be supplied normally and the MAS-A100 may not start due to the potential difference between the ground connection of the PoE hub installation location and the MAS-A100 installation location.

Ethernet power-saving function

Power saving may disrupt normal operation of a Dante network. Disable power saving, such as IEEE 802.3 Energy Efficient Ethernet (EEE) or Green Ethernet, or use a network switch without a power saving function.

IGMP snooping

Dante networks always use multicast communication.

IGMP snooping of a network switch may disrupt Dante network communications if it is not configured appropriately for a Dante network.

For a network that uses only unicast flows as the Dante transmission method, it is recommended that you disable IGMP snooping. It is not recommended to use network switches (such as TPLINK TL-SG1005P) in which IGMP snooping cannot be disabled and cannot be configured appropriately for a Dante network.

On a network that uses multicast flows as the Dante transmission method, or that has other multicast traffic on the same segment (same layer 2 network), IGMP snooping may be necessary to avoid multicast flooding. In that case, select a managed switch and configure it appropriately for a Dante network observing the following points.

- Do not use fast leave (immediate leave)
- Do not use IGMP proxy
- Do not use the communications ports used by the Dante network.

For details about Dante networks and network switches, visit the following Audinate website.

<https://www.audinate.com/networks-and-switches>

Network sharing with other network devices

Dante audio networks use LAN cables (Cat 5e or higher) to transfer real-time audio signals with low latency. It is recommended that you do not connect network devices, other than Dante network devices and related devices, to the network to which MAS-A100 and Dante devices are connected.

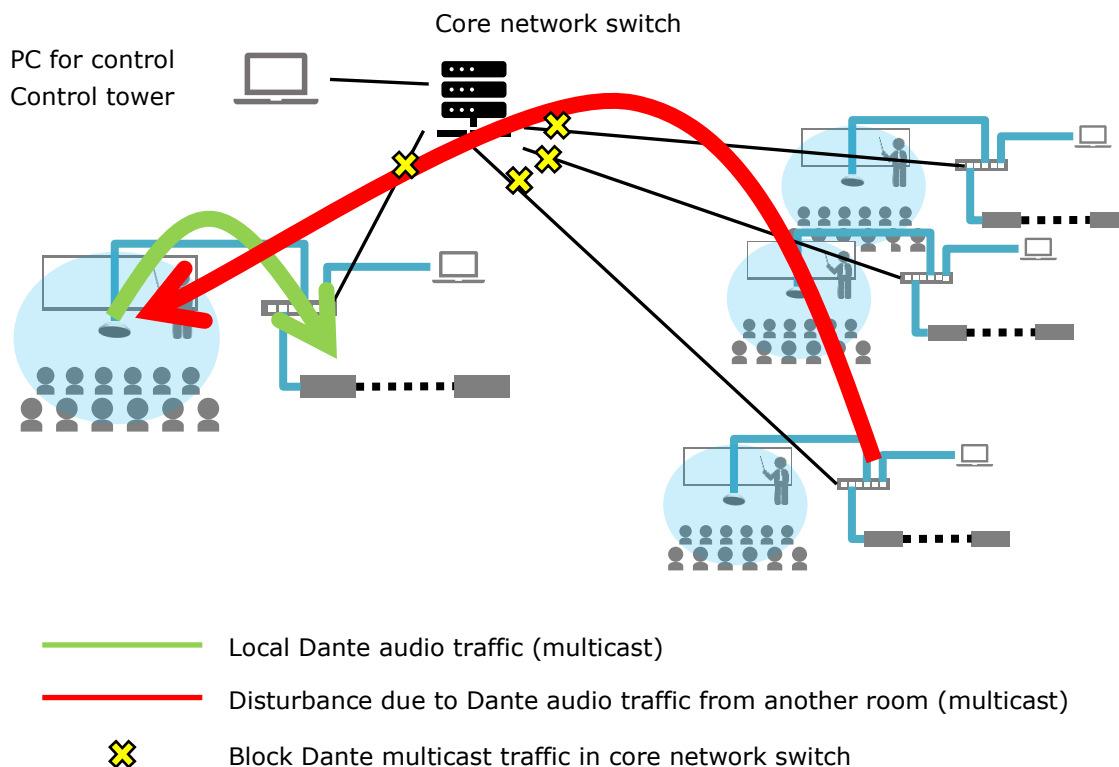
If other devices use the network, the network bandwidth may be exceeded, which may cause audio interruptions and noise.

Connecting multiple systems to the same network

When using MAS-A100 units and speech reinforcement/audio capture devices independently in multiple rooms, design the network so that the multicast traffic is completely contained within each room. If a Dante audio device is connected to a non-Dante compatible network switch or a network switch that is not configured to support Dante, devices in other rooms may be connected or disconnected and other Dante audio network reorganization or conflicts may occur, causing audio interruptions and other issues.

Place a broadband router in each room to separate the network segments or disable multicast communication on the core switch.

Multicast communication is also used for the device search function of the network interface for setup. Allow multicast communication only when searching for devices in the control room, and then block multicast after device registration has been completed.



Compatible third-party products

The following products have been confirmed to be compatible. A PoE hub will not operate correctly and stably using the factory default settings. Follow the procedure in the next section to change the settings before use.

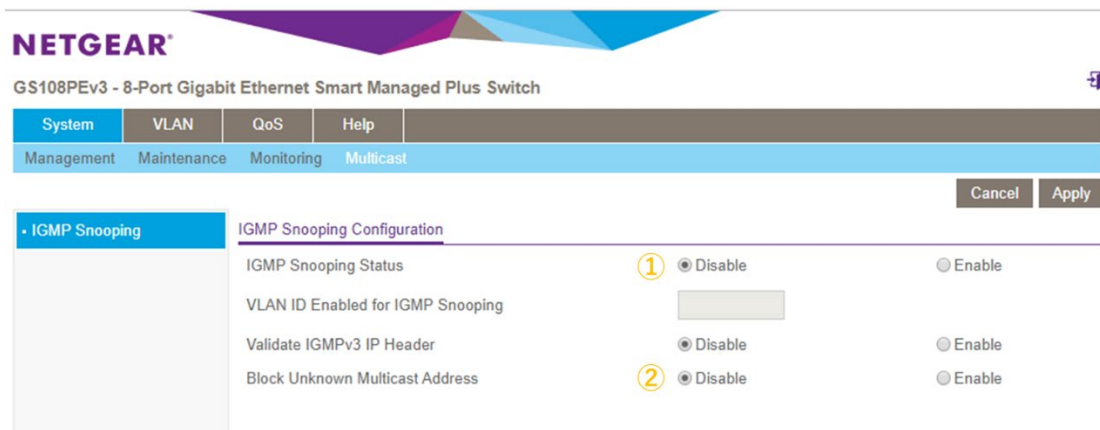
Category	Manufacturer and model name	Version
PoE hub	NETGEAR GS108PEv3	2.06.10
	Cisco SG350-28P	2.5.0.90

NETGEAR GS108PEv3 setup

Log in to the PoE hub web page and make the following settings. If you do not know the IP address of the PoE hub, use the ProSAFE Plus Utility from the NETGEAR website for search and configuration.

<https://www.netgear.com/support/product/ProSafe%20Plus%20Utility#download>

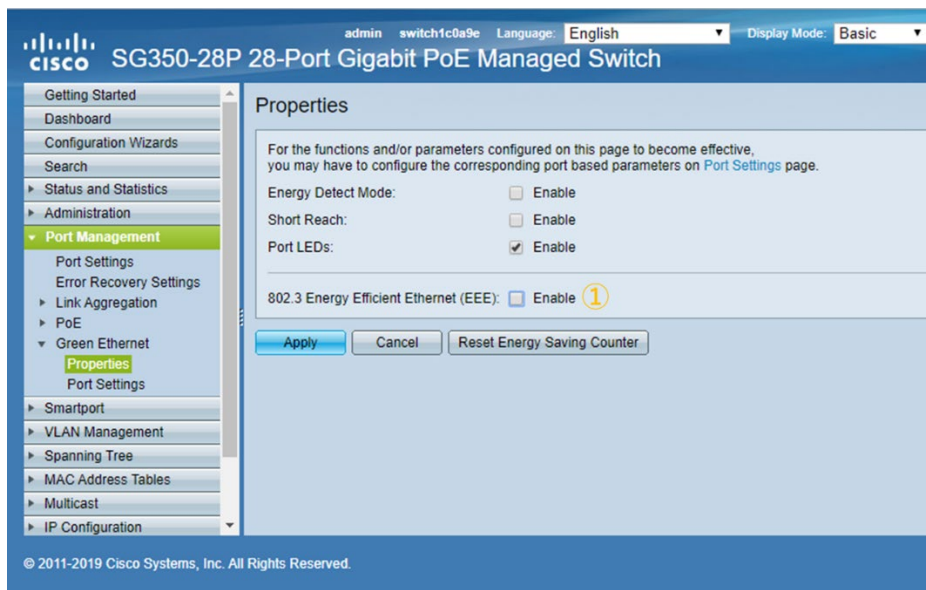
1. Set [IGMP Snooping Status] to [Disable]. (Factory default setting is [Enable].)
2. Set [Block Unknown Multicast Address] to [Disable].



IGMP snooping of the PoE hub must be disabled since it disrupts Dante networks. It is recommended to use only unicast flows as the Dante transmission method.

Cisco SG350-28P setup

Clear the check mark from [802.3 Energy Efficient Ethernet (EEE)], and apply the setting.



If IGMP snooping is used with a PoE hub for Dante or other multicast traffic, configure the IGMP snooping appropriately for a Dante network (e.g., configure the IGMP filter not to prevent Dante traffic).

4.4. Audio design

This section describes the design for connecting MAS-A100 units to the system.

MAS-A100 audio output overview

The MAS-A100 functions as a Dante transmitter device with two channels of audio outputs (Main output and Rec output) on a Dante network. Design how to route the separate outputs to the speech reinforcement system and recording system, respectively.

A wide range of Dante peripherals are available, from simple D/A converters and Dante-compatible DSPs through to Dante receiver software. The MAS-A100 supports the Dante audio network and is compatible with these products.

Routing

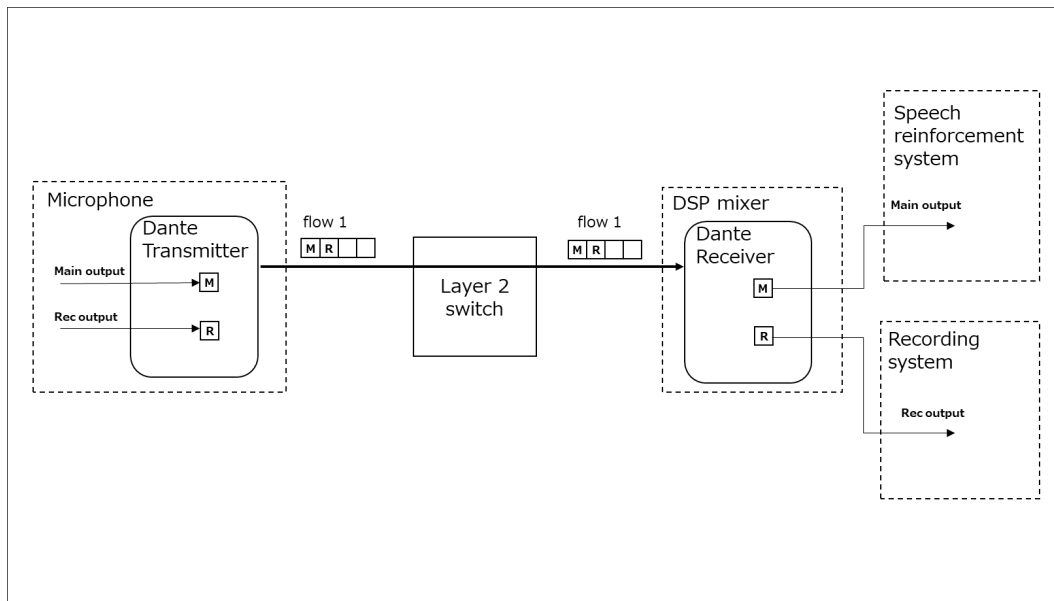
Routing of Dante signals for Dante receivers within the same segment (same layer 2 network) is used to transfer audio.

Dante Controller and Dante-compatible DSPs are used to configure Dante devices and route audio between Dante devices. For details, see “5.4 Routing of Dante audio” (page 59) and refer to the Dante Controller user guide or the operating instructions for the Dante-compatible DSP.

Transmission method

There are two kinds of “flows,” which are the units of transmission for audio data on a Dante network. One is a unicast flow, where a Dante device transmits a flow directly to another Dante device. The other is a multicast flow, where a Dante device transmits a flow to multiple Dante devices within a network. By default, flows are unicast. It is not necessary to change this usually. Each unicast flow can contain up to four audio channels.

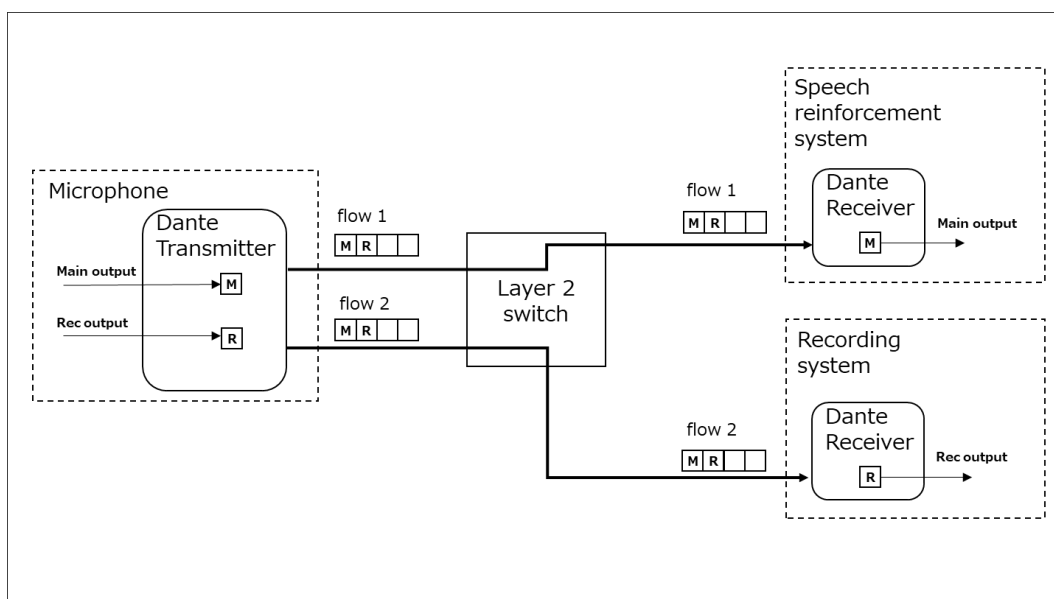
The following diagram shows an example of one unicast flow, where the MAS-A100 sends the Main output and the Rec output to the Dante receiver of the DSP mixer. Then, the Main output passes through the speech reinforcement system, and the Rec output passes through the recording system, respectively.



The following diagram below shows an example of two unicast flows, where the MAS-A100 sends the Main output to the Dante receiver of the speech reinforcement system, and sends the Rec output to the Dante receiver of the recording system.

The maximum number of flows of each MAS-A100 is two. Therefore, it is necessary to use multicast flows instead of unicast flows if there is a need to send audio to three or more Dante devices.

For details about Dante audio transmission using a multicast flow, refer to the Dante Controller user guide.



Compatible third-party products

The following products have been confirmed to be compatible.

Category	Manufacturer and model name	Version
Dante devices (Receivers)	Audinate ADP-DAO-AU-0X2	Dante firmware 4.1.5.2 Hardware 4.1.4.2
	Audinate ADP-DAO-AU-0X1	Dante firmware 4.1.5.2 Hardware 4.1.4.2
	Audinate ADP-USB-AU-2X2	Dante firmware 4.1.9.1 Hardware 4.1.8.1
	Audinate Virtual Soundcard	Windows 4.1.0.4
	Symetrix Prism 4x4	4.0.7
	Shure P300	2.1.6
	BOSE EX-1280C	2.290

Note that the power supply of Symetrix Prism 4x4 and Shure P300 is PoE+.

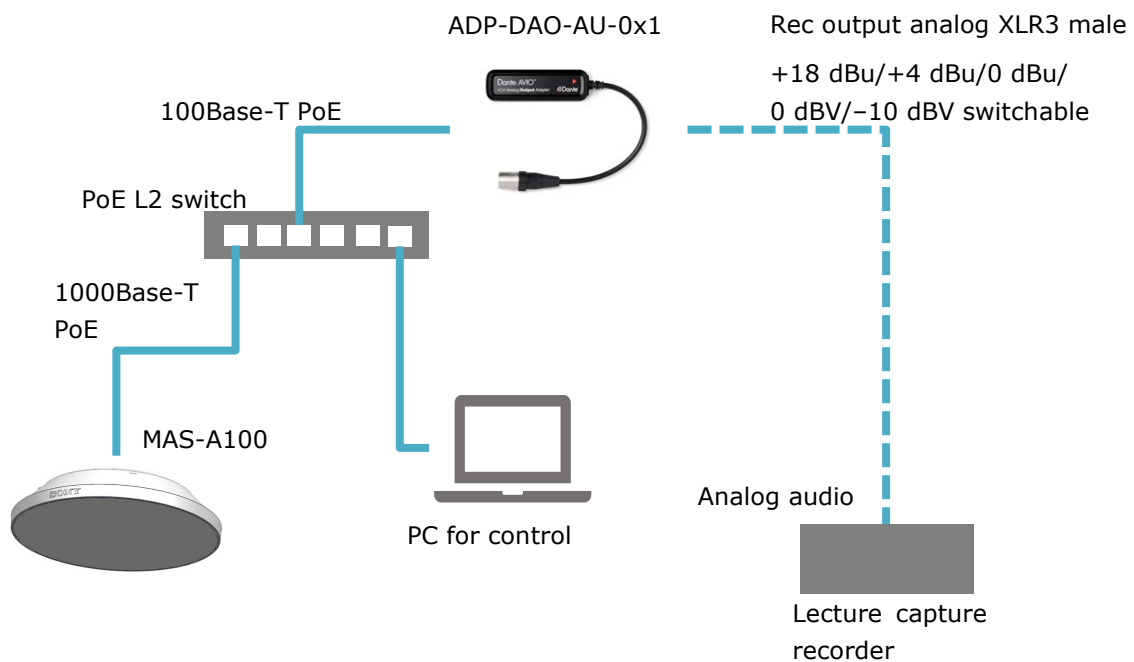
Small-scale audio capture system design

This sections describes a small-scale system that uses one MAS-A100 unit with the following configuration.

Analog connection

See “3.1 Recording a lecture in a small room” (page 8) for a configuration example. The Dante audio output of the MAS-A100 is used as the analog audio signal for recording (Rec) by using an Audinate ADP-DAO-AU-0x1.

Audio is recorded by connecting the analog audio signal to the input of an existing lecture capture recorder.

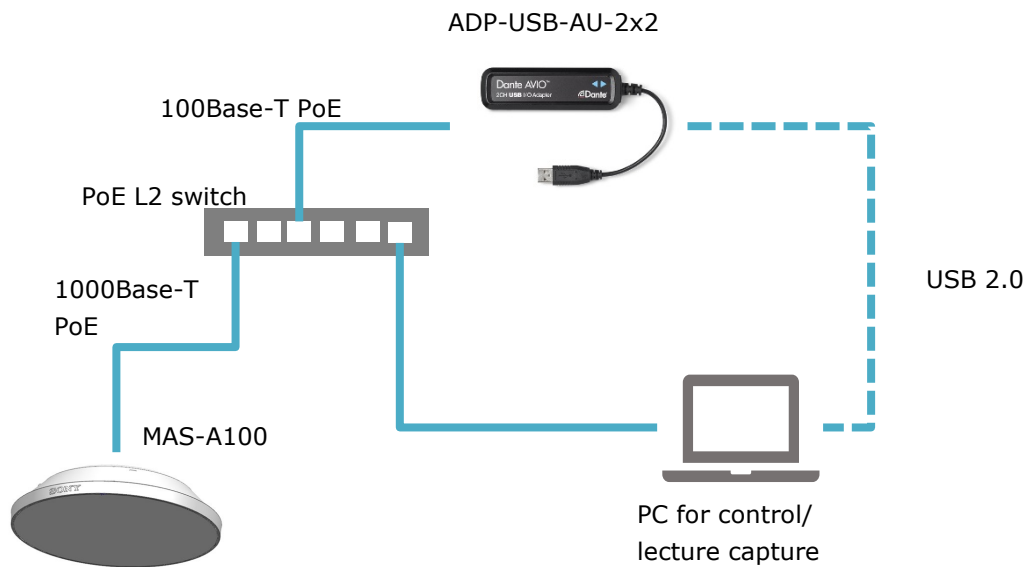


USB connection

The Dante audio output of the MAS-A100 is used as the audio signal for recording (Rec) by using an Audinate ADP-USB-AU-2x2.

When using a Windows-based lecture capture recorder, the ADP-USB-AU-2x2 is recognized as a sound card without the need to install additional drivers. This allows you to easily connect to PC-based telecommunications software (such as Skype, Microsoft Teams, Zoom, Webex).

Since the ADP-USB-AU-2x2 has two channels of USB output, the Rec output of two MAS-A100 units can be imported into the computer independently.

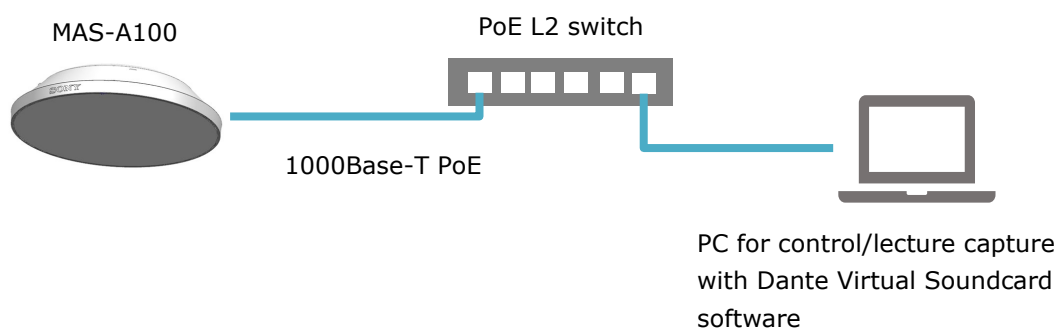


Software connection

The Dante audio output of the MAS-A100 is used as the audio signal for recording (Rec) by using an Audinate Dante Virtual Soundcard.

Dante Virtual Soundcard software installed for a Windows/macOS-based lecture capture recorder is recognized as a sound card.

Since Dante Virtual Soundcard has up to 64-channel input, the Rec output of 64 MAS-A100 units can be imported into the computer independently (the maximum number of channels is dependent on the computer performance).



Speech reinforcement system design

This section describes the design of the hands-free speech reinforcement system using the MAS-A100.

Since the MAS-A100 has a Dante audio output, D/A conversion is required to connect to an existing analog speech reinforcement audio system. In addition, in environments where a Dante-compatible DSP is installed, the Dante audio and the audio system for speech reinforcement can be connected directly using digital audio signals.

Speech reinforcement level adjustment

During installation, use the MAS-A100 test signal to adjust the audio system settings so that the speech can be reinforced with optimum volume, and then calibrate the audio system with those fixed settings.

Design the audio system so that volume adjustment of the audio system for speech reinforcement (channel volume, master volume) for the audio signal from the MAS-A100 is not possible during normal operation. To make volume adjustments during normal operation, use the MASM-1, the web interface of each MAS-A100 unit, or the external control API.

Signal processing

Insertion of non-linear processing, dynamic processing, user-controlled volume adjustment, and audio quality adjustment functions between the MAS-A100 and the speech reinforcement loudspeakers is not recommended.

If connecting to the microphone input of a speech reinforcement system, the system may not function normally due to non-linear processing of devices, such as a feedback canceler, automixer, noise gate, etc.

If you cannot exclude these functions, check the sound quality carefully.

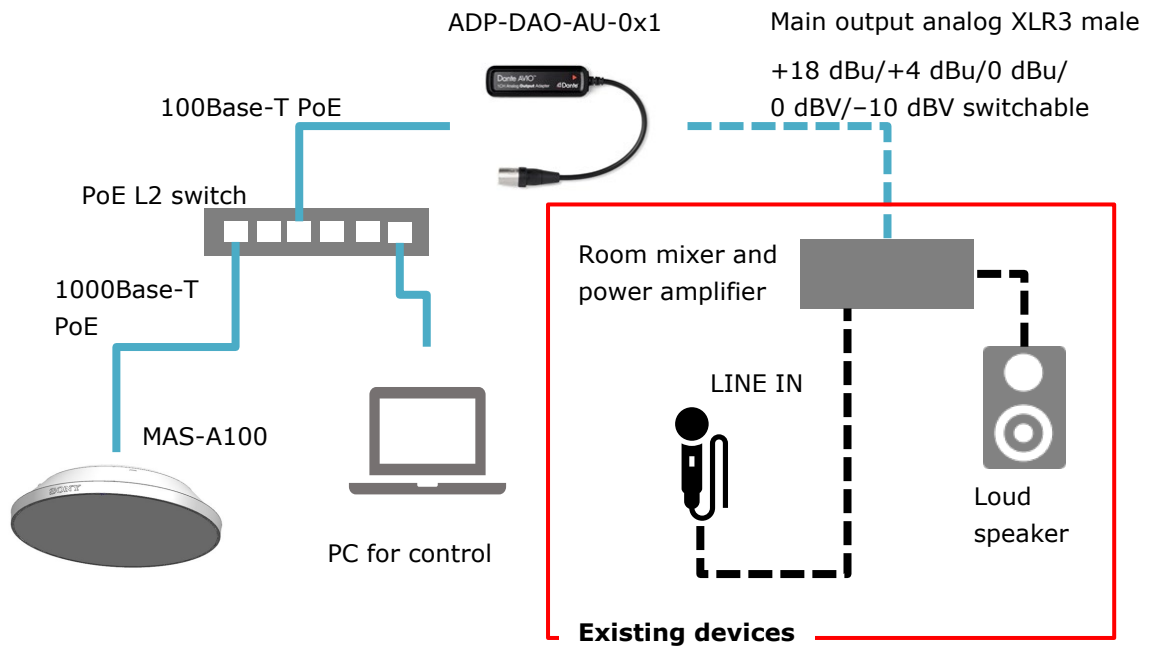
Examples of non-recommended signal processing

Function name	Reason	Remarks
Automixer, AGC	Dynamic processing	
Compressor, limiter	Dynamic processing, non-linear processing	
Expander	Non-linear processing	
Distortion and other effects	Non-linear processing	
Feedback canceler	Dynamic processing, non-linear processing	
Feedback fighter	Dynamic processing, non-linear processing	

Function name	Reason	Remarks
Feedback reducer	Dynamic processing, non-linear processing	
Volume and audio quality adjustments	User operated	Muting is supported
SPL processor	Dynamic processing, non-linear processing	
Echo canceler	Dynamic processing, non-linear processing	
Noise gate	Dynamic processing	
Chorus, flanger	Dynamic processing	
Transient suppressor	Dynamic processing	
Ducker	Non-linear processing	
Dynamics processor	Non-linear processing	
Subharmonics	Non-linear processing	
De-esser	Non-linear processing	
Delay, echo	Other	Long duration effects

Analog connection with existing audio system for speech reinforcement using 1 MAS-A100 unit

The Dante audio output of the MAS-A100 is used as the Main analog audio signal by using an Audinate ADP-DAO-AU-0x1. Speech reinforcement is achieved by connecting the analog audio signal to the line input of an existing audio system.

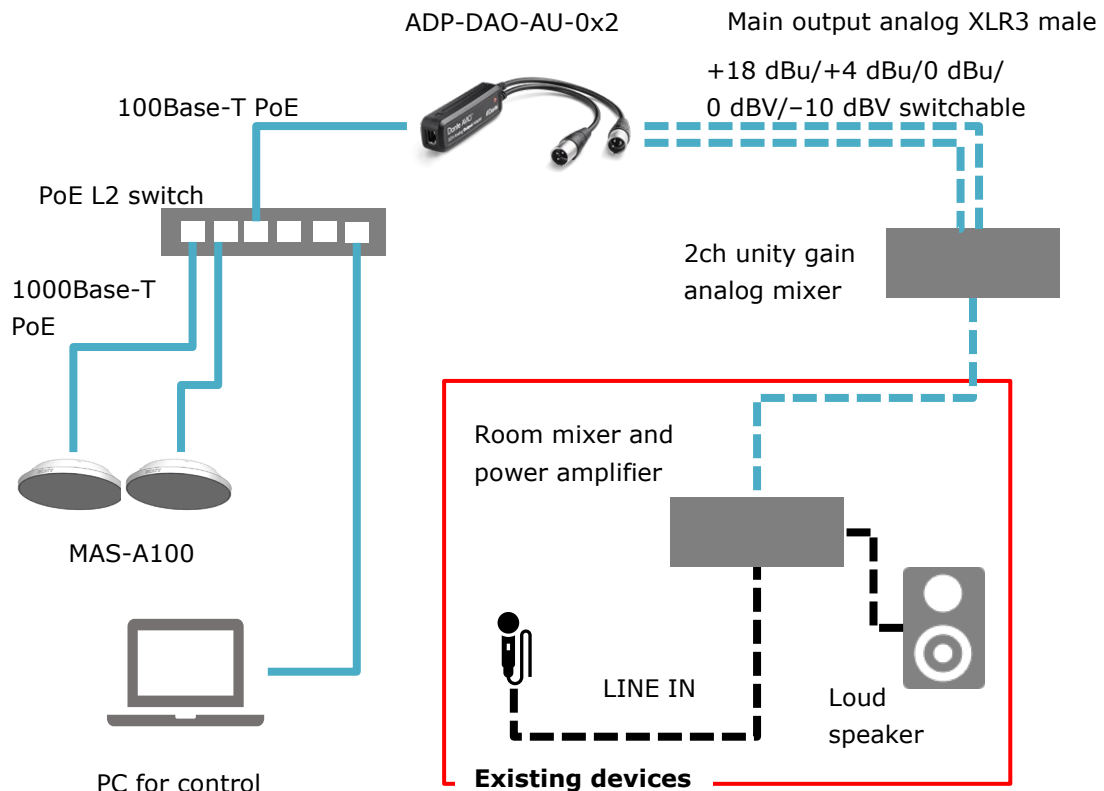


Analog connection with existing audio system for speech reinforcement using 2 MAS-A100 units

The Dante audio output of the MAS-A100 is used as the Main analog audio signal by using an Audinate ADP-DAO-AU-0x2. The signals are mixed 1:1 using a fixed-gain mixer, unity gain mixer, a microphone combiner, or splitter cable.

Speech reinforcement is achieved by connecting the mixer output to the line input of an existing audio system.

To connect three or more MAS-A100 units, use one ADP-DAO-AU-0x2 unit for every two MAS-A100 units, increase the number of input channels of the analog mixer by the number of MAS-A100 units, and mix the signals without increasing the gain.



Connection with Dante-compatible DSP using three or more MAS-A100 units

A system with a degree of freedom can be constructed by using a Dante-compatible DSP in the speech reinforcement audio system. This section describes a system using the Symetrix Prism 4x4.

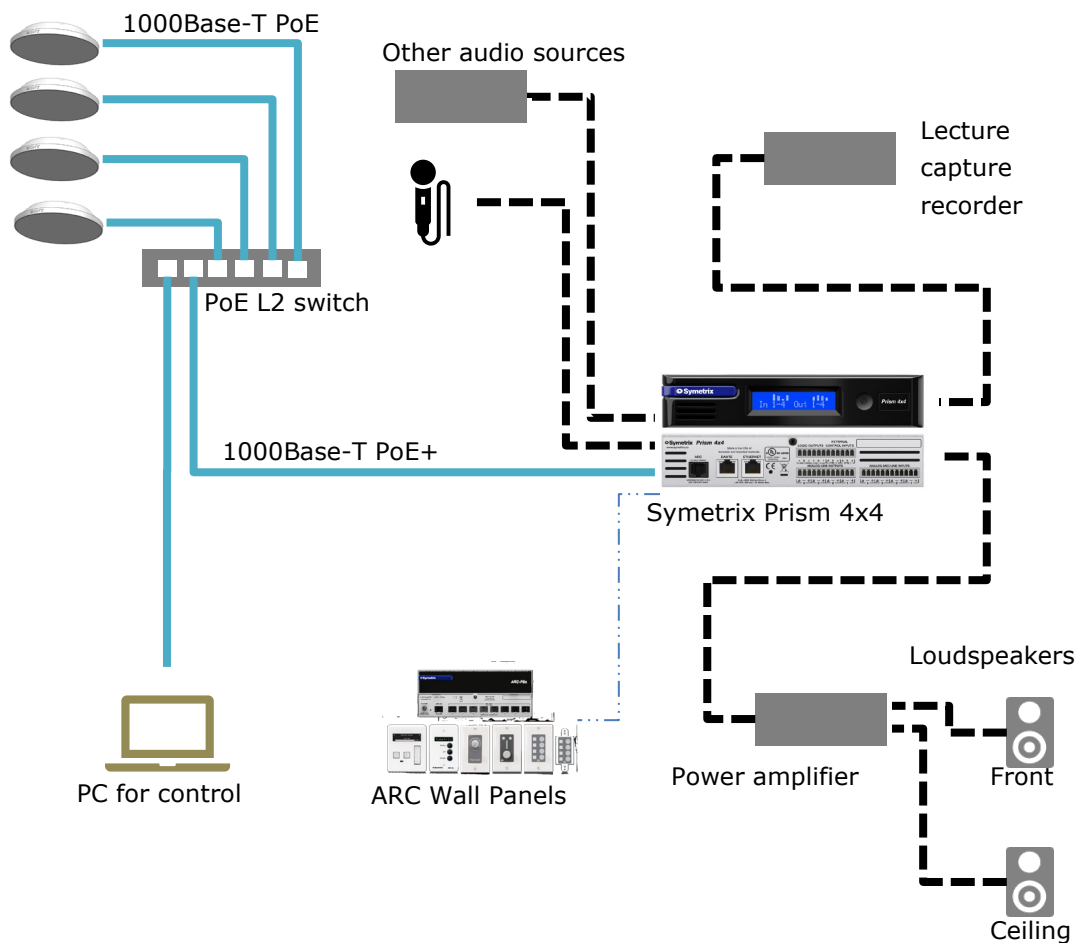
This assumes the MAS-A100 and loudspeaker installation conditions described in "3.2 Reinforcing/capturing a lecture in a medium-sized room" (page 9).

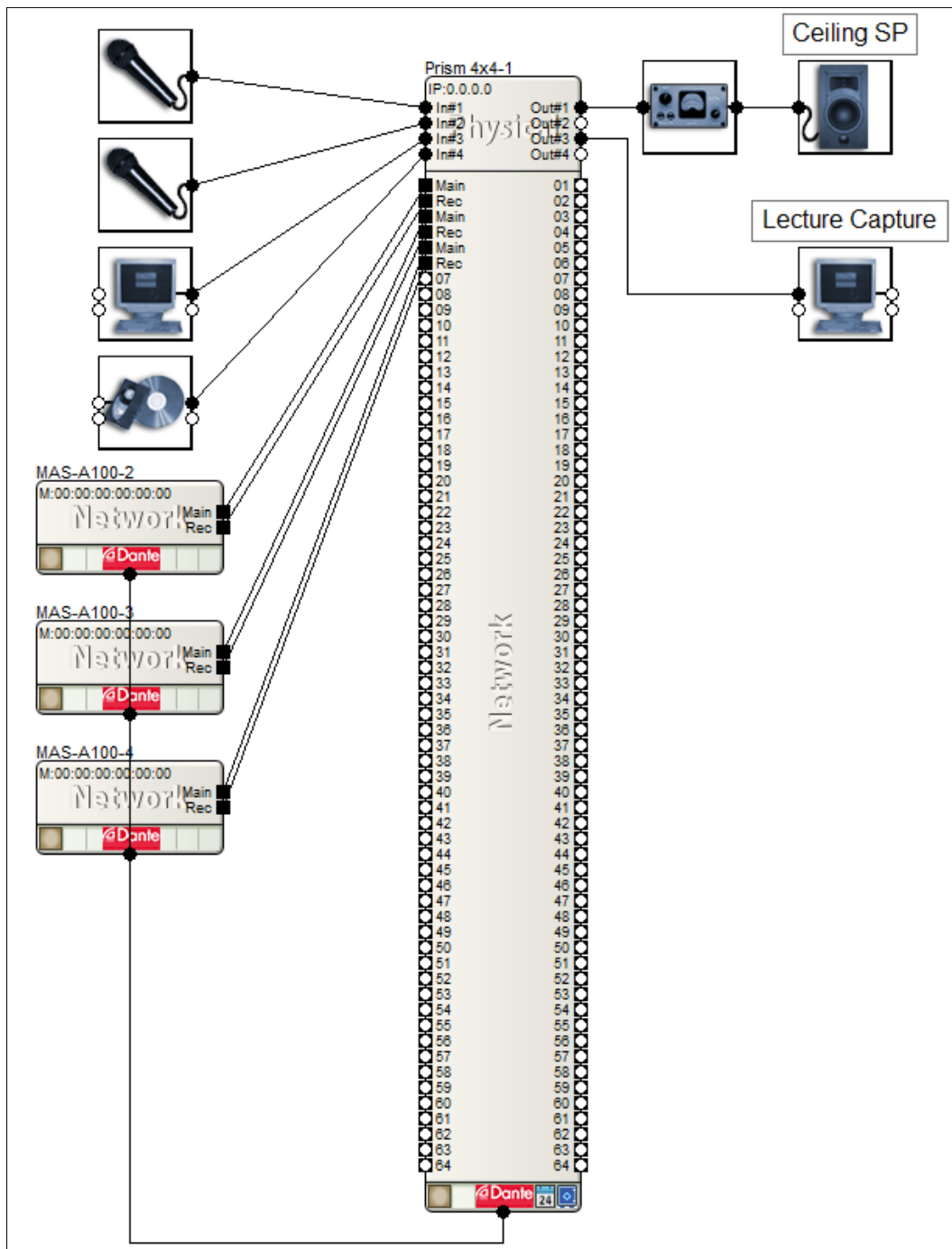
The following diagram shows an example with Prism 4x4 configured using Symetrix Composer.

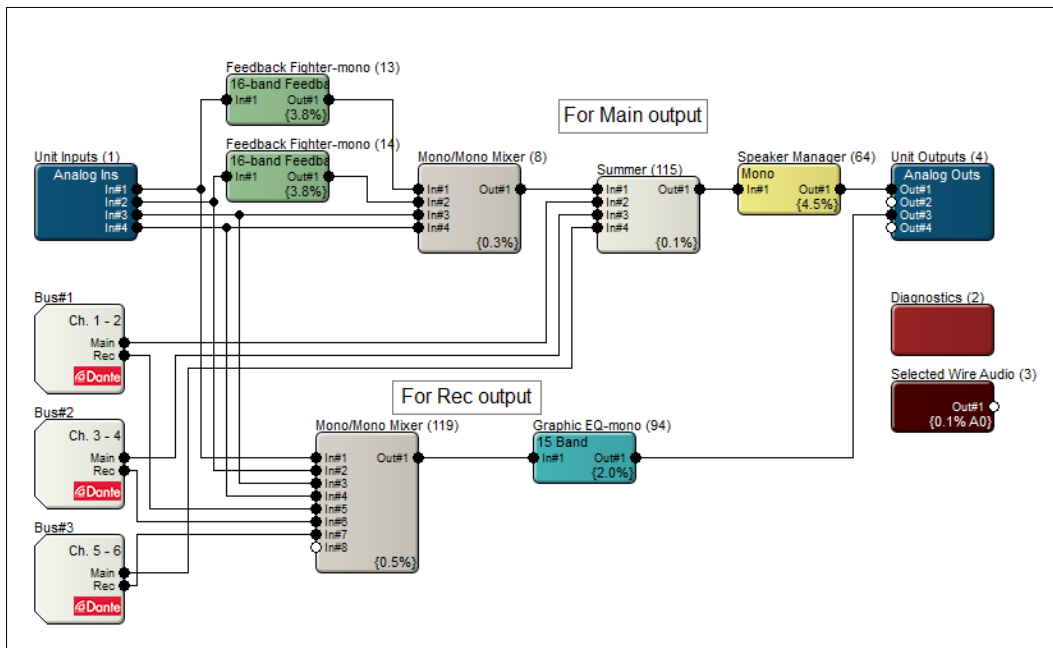
The Main output of the MAS-A100 units are routed to the Dante inputs of the Prism 4x4 (up to 64 channels). The Main outputs from the MAS-A100 units are mixed using a summer and the volume is trimmed during installation. The volume and sound quality of other audio sources (such as handheld microphones, PC audio) are adjusted on each channel of a separate bus.

The power amplifier mixes the MAS-A100 mixer output from the summer with the other audio sources.

Design the audio system so that manual volume adjustment by the mixer and volume adjustment of the audio system for speech reinforcement (channel volume, master volume) for the Main output from the MAS-A100 is not possible during normal operation. To make volume adjustments during normal operation, use the MASM-1, the web interface of each MAS-A100 unit, or the external control API. For a sample program for control using the external API, see “B.6 Sample program” (page 116).







4.5. Remote lecture system design

This section describes an example of a remote lecture system design.

The Rec output for audio capture of the MAS-A100 is used for sending audio to a remote destination. Sending the Main output for speech reinforcement to a remote destination is not recommended.

For details about audio capture design and important points, see "Small-scale audio capture system design" (page 32). For details about speech reinforcement design and important points, see "Speech reinforcement system design" (page 34).

A remote conferencing system includes non-linear and non steady-state processing, such as echo cancelers and codecs, that can cause audio interruptions with respect to the MAS-A100 signal processing, adversely affecting audio quality. Before use, initiate a conference call and check that the sound quality is sufficient.

Remote lecture using analog connection of two MAS-A100 units

This section shows an example connection of an existing analog speech reinforcement device with an existing remote communication device.

The Main outputs from two MAS-A100 units are converted to two analog signals by D/A conversion using the ADP-DAO-AU-0x2. The signals are mixed 1:1 using a fixed-gain mixer, unity gain mixer, a microphone combiner, or splitter cable. The mixer

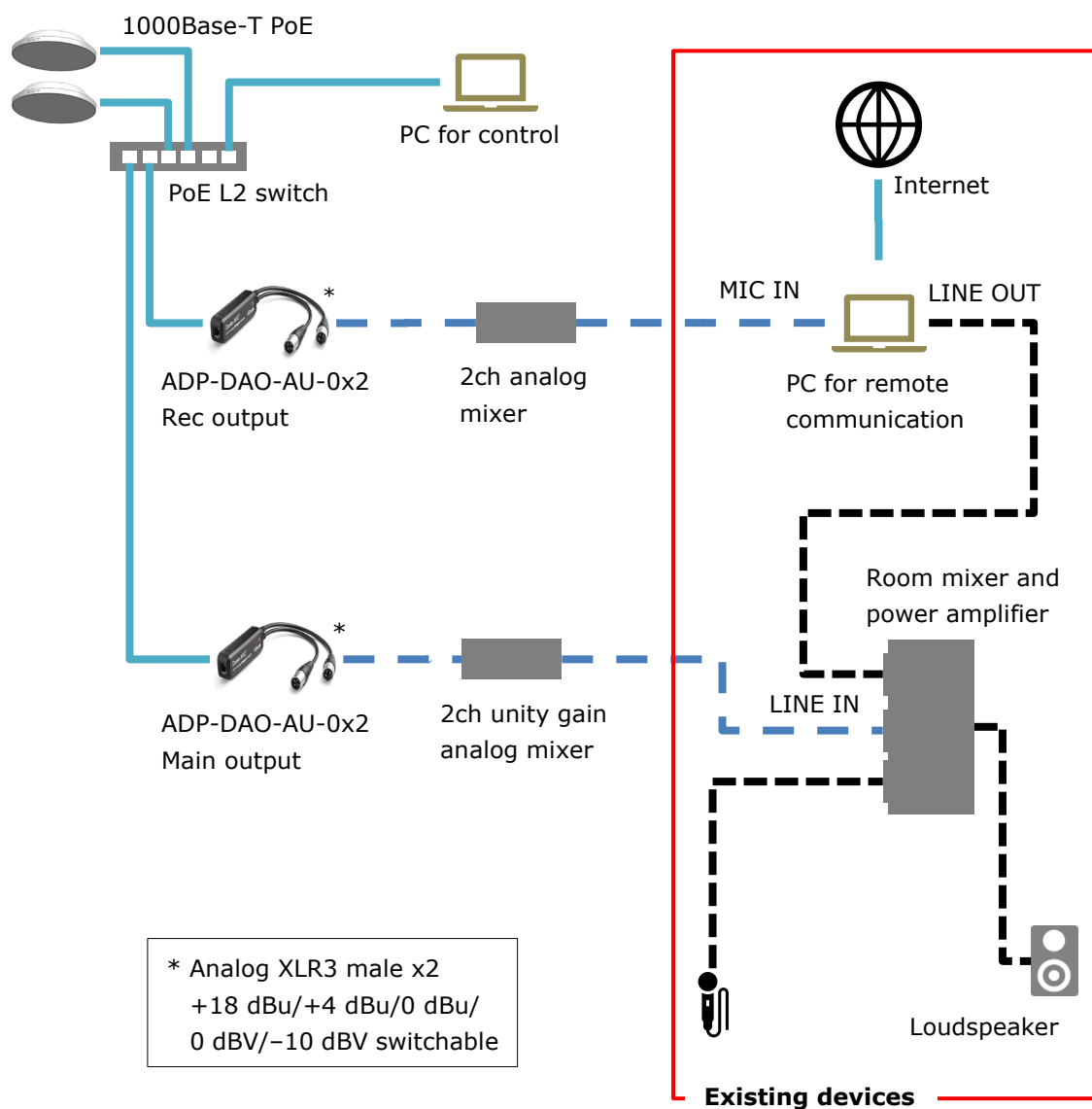
output is connected to the line input of an existing mixer and then output from the loudspeakers. The volume and sound quality are adjusted by the gain control and equalizer within the MAS-A100 units, so do not adjust when mixing the two signals. If connecting to the microphone input of a speech reinforcement system, the system may not function normally due to non-linear processing of devices, such as a feedback canceler, automixer, noise gate, etc.

Design the audio system so that volume adjustment by the mixer and volume adjustment of the audio system for speech reinforcement (channel volume, master volume) for the audio signal from the MAS-A100 is not possible during normal operation. To make volume adjustments during normal operation, use the MASM-1, the web interface of each MAS-A100 unit, or the external control API.

The Rec outputs are converted to analog by D/A conversion by the second ADP-DAO-AU-0x2, and then input to a mixer for volume and audio quality adjustments before input to the microphone input of the communications device.

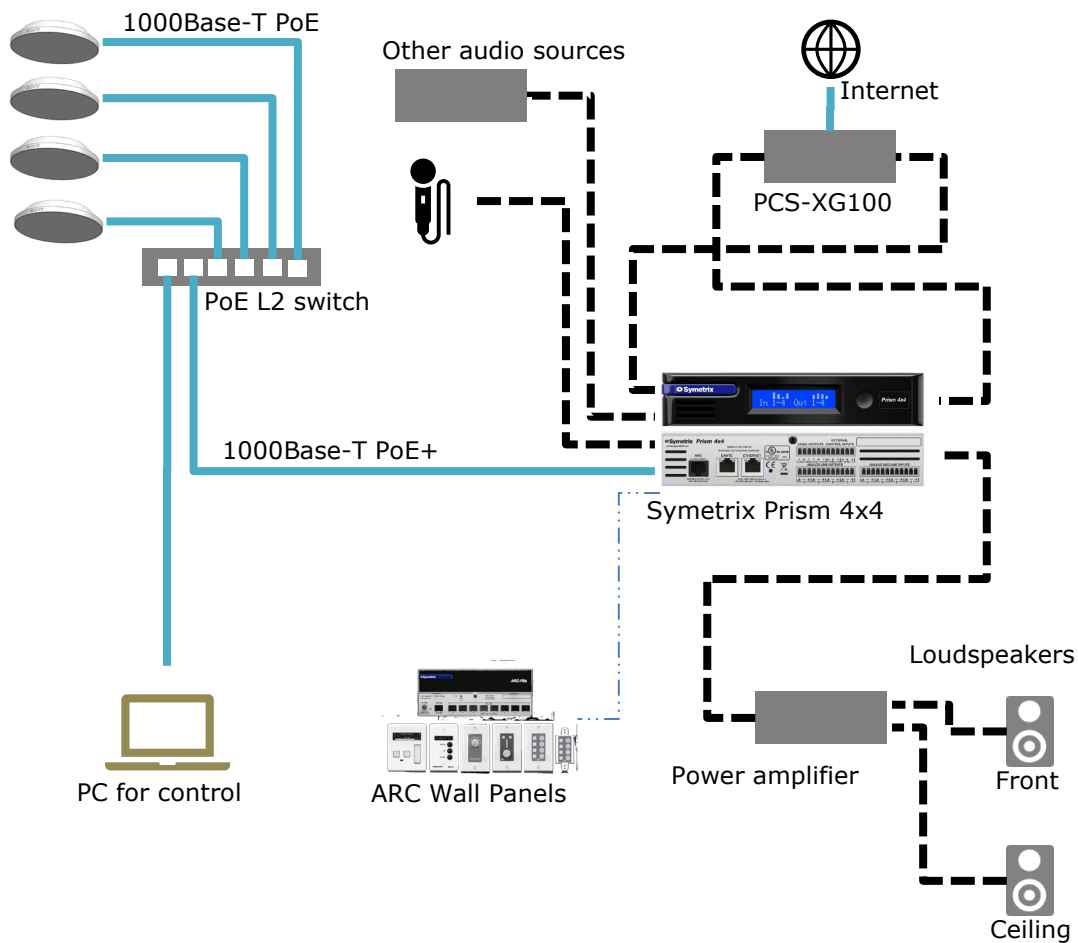
With this configuration, the audio of existing microphones cannot be sent to the communications device. The use of the output (MAS-A100 Main output) from the speech reinforcement audio mixer is not recommended. To send the sound of an existing microphone to a communication device, mix the Rec output of the MAS-A100 and the post-fader output of the existing microphone on a bus separate from the speech reinforcement system and use the audio output from the mixer. If Main and Rec are both input to the same mixer, use a mixer with input crosstalk between the Main output and Rec output of -80 dB or less. Crosstalk may be large if you use an inexpensive stereo mixer for L/R separation between the Main output and Rec output. If there is mixing of the Main output and Rec output due to crosstalk, the system may not calibrate properly and will cause unexpected feedback.

In this connection method, there are many additional devices and adjustment points that should not be operated, and stable operation may not be possible due to incorrect operation or settings. Even if you have existing equipment, it is recommended that you design a system using a Dante-compatible DSP as described in the next section.



Connection with Dante-compatible DSP using three or more MAS-A100 units for remote lecture

This assumes the MAS-A100 and loudspeaker installation conditions described in "3.3 Remote lecture" (page 11).



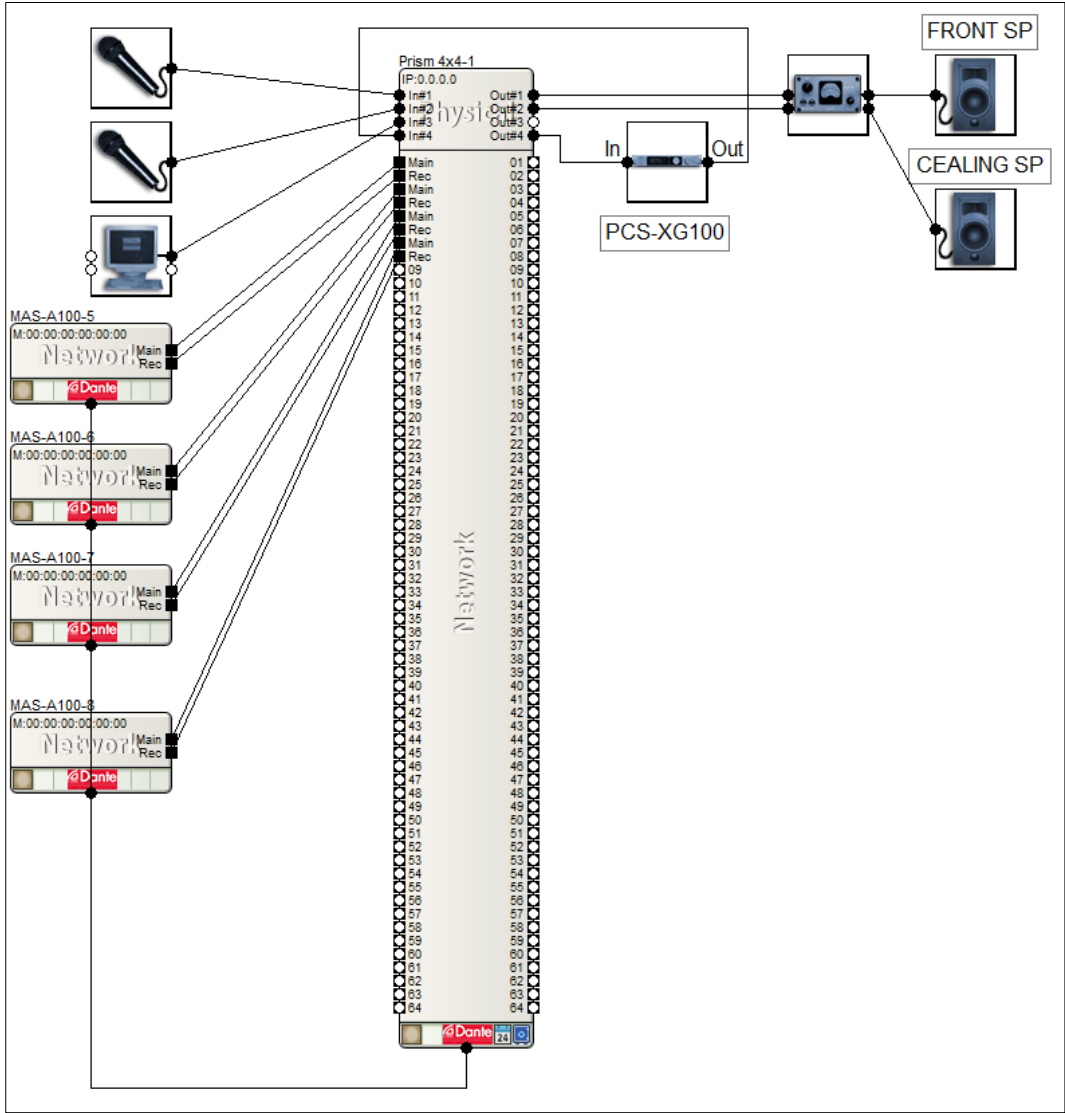
The following diagram shows an example with Prism 4x4 configured using Symetrix Composer.

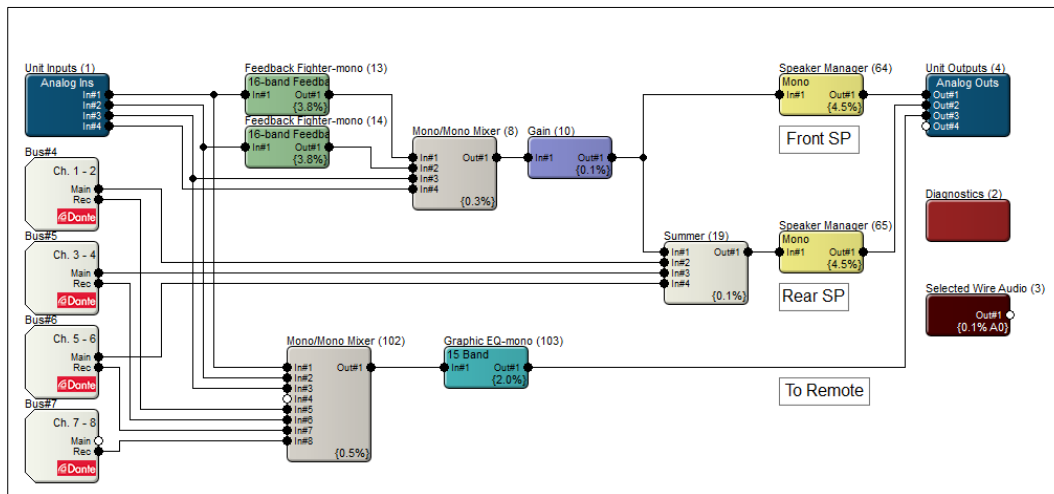
Design the audio system so that manual volume adjustment by the mixer and volume adjustment of the audio system for speech reinforcement (channel volume, master volume) for the Main output from the MAS-A100 is not possible during normal operation. To make volume adjustments during normal operation, use the MASM-1, the web interface of each MAS-A100 unit, or the external control API.

For a sample program for control using the external API, see "B.6 Sample program" (page 116).

This example uses the echo canceler function of the PCS-XG100. Since the echo canceler includes non-linear processing, it may affect speech reinforcement and audio capture. In addition, the microphone input volume of the PCS-XG100 and other communication devices that have an echo canceler function greatly affects the effectiveness of the echo canceler.

After installation, always perform a communication test and check that the echo canceler is operating normally at the communication destination and that sufficient quality is maintained.





5. Setup

5.1. MASM-1 preparation

Download MASM-1 and install it on a Windows PC.

Download

The software can be downloaded from the following website.

<https://pro.sony/product-resources/software-firmware/masm-1>

Operating environment

OS: Microsoft Windows 10 (64-bit version)

Processor: 2 GHz or higher

Memory: 4 GB or higher

Storage free capacity: 300 MB or higher

Display resolution: 1280×720 or higher

Browser: Google Chrome (version 70 or higher, installed in the default location)

Other: .NET framework (4.7.2 or higher)

Usage

For details about using MASM-1, refer to the following help guide.

<https://helpguide.sony.net/MAS/MASM-1/v1/en/index.html>

During setup, observe the following points.

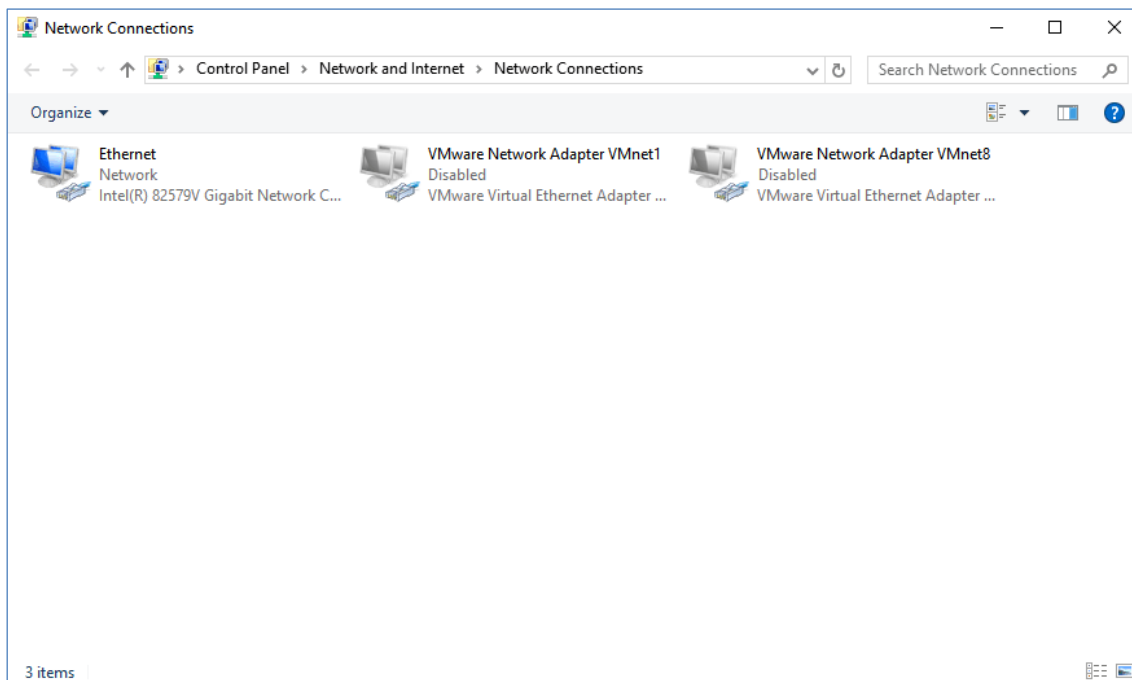
Firewall

MASM-1 discovers MAS-A100 units by communication via UDP port 42380.

Configure the firewall and security software of the PC to allow this communication to discover MAS-A100 units.

Multi interface

If multiple network adapters are operating, MAS-A100 units may not be discoverable. Disable all network adapters other than the one to which the MAS-A100 units are connected.



HTTP proxy

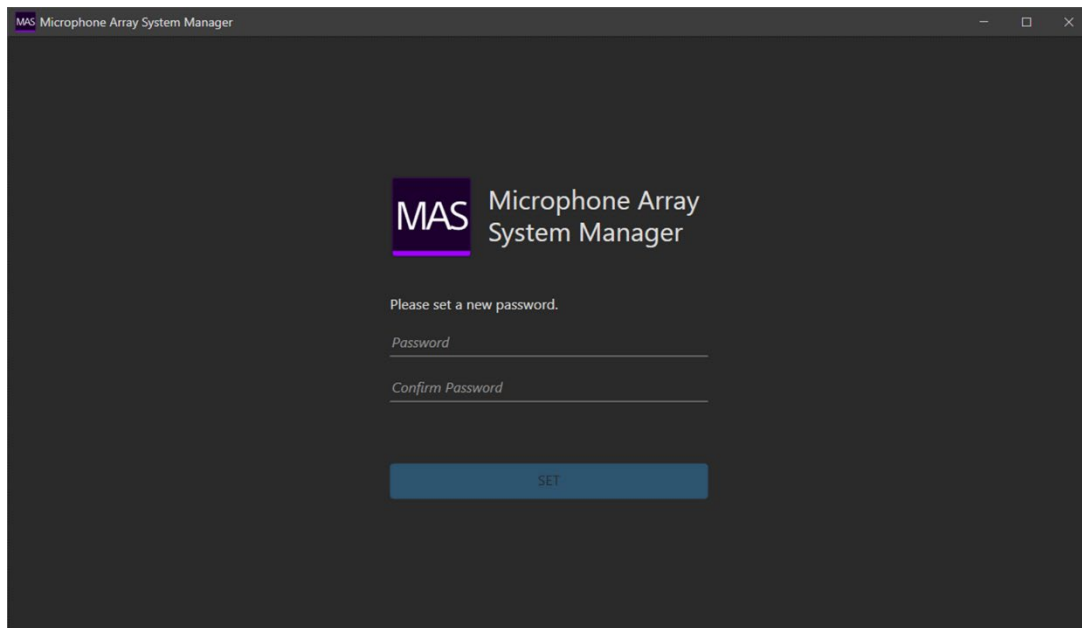
MASM-1 performs setup and operation of MAS-A100 units (except discovery) by communication via HTTP. Configure the HTTP proxy of the PC to allow HTTP access to the MAS-A100 units.

Default password setting

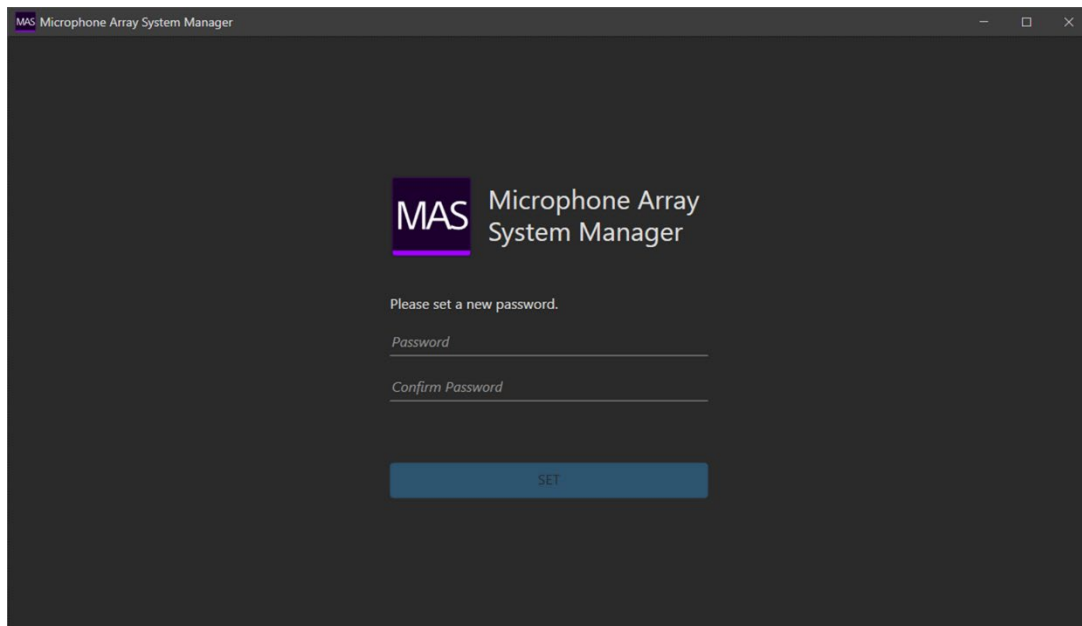
Set a password the first time the software is started after installation. The password must be at least eight characters and no more than 64 characters, and contains at least one lowercase letter, one uppercase letter, and one number. The password can also contain symbol characters.

! \$ % ' () = - ~ ^ | @ ` [{ }] _ / ? < > + * .

Setting the same password as the MAS-A100 allows for easy MAS-A100 registration.



When the software is subsequently started, enter the password that was configured when starting the application for the first time.



5.2. MAS-A100 initial setup using MASM-1

The initial setup is performed using MASM-1 on a PC connected to the same segment (same layer 2 network) as the MAS-A100 units in a room where the MAS-A100 units are installed.

MASM-1 is a free software application that performs setup and controls operation of one or more MAS-A100 units remotely via a network.

Do not use multiple instances of MASM-1, or use the web UI of the MAS-A100 concurrently with MASM-1 to perform setup and operation of the same MAS-A100 unit. Unexpected results that prevent setup and operation of the MAS-A100 according to this document may occur.

MAS-A100 discovery and IP address setting

Discover and set up the MAS-A100 units connected to the network and initialize them.

1. Open the [Register] tab of MASM-1, and press the [Scan] button.

Dante Controller discovers MAS-A100 units on the same segment (same layer 2 network).

Notes

- MAS-A100 units beyond the router or layer 3 switch cannot be discovered. When performing remote management across a router, connect the remote management PC to the same segment, register the unit, and then reconnect the computer to the management segment.
- If no MAS-A100 is discovered after pressing the [Scan] button, check the network connection and firewall settings (see "4.3 Network design" (page 23) section).
- If multiple network adapters are operating, MAS-A100 units may not be discoverable.

2. From the list of discovered MAS-A100 units, place a check mark beside the MAS-A100 units to register in MASM-1.

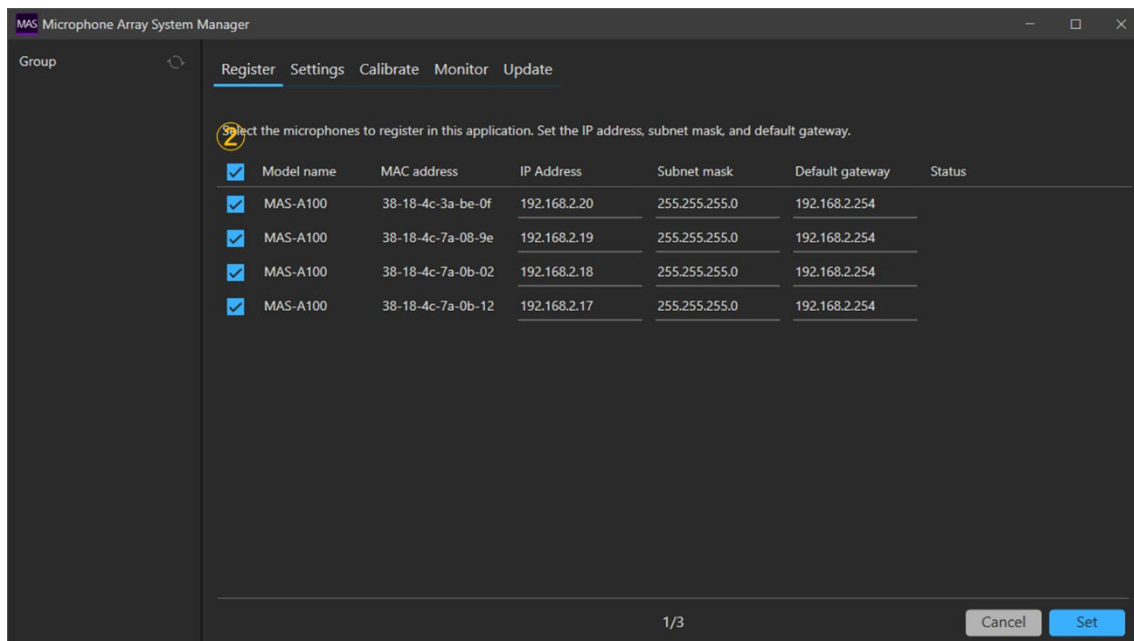
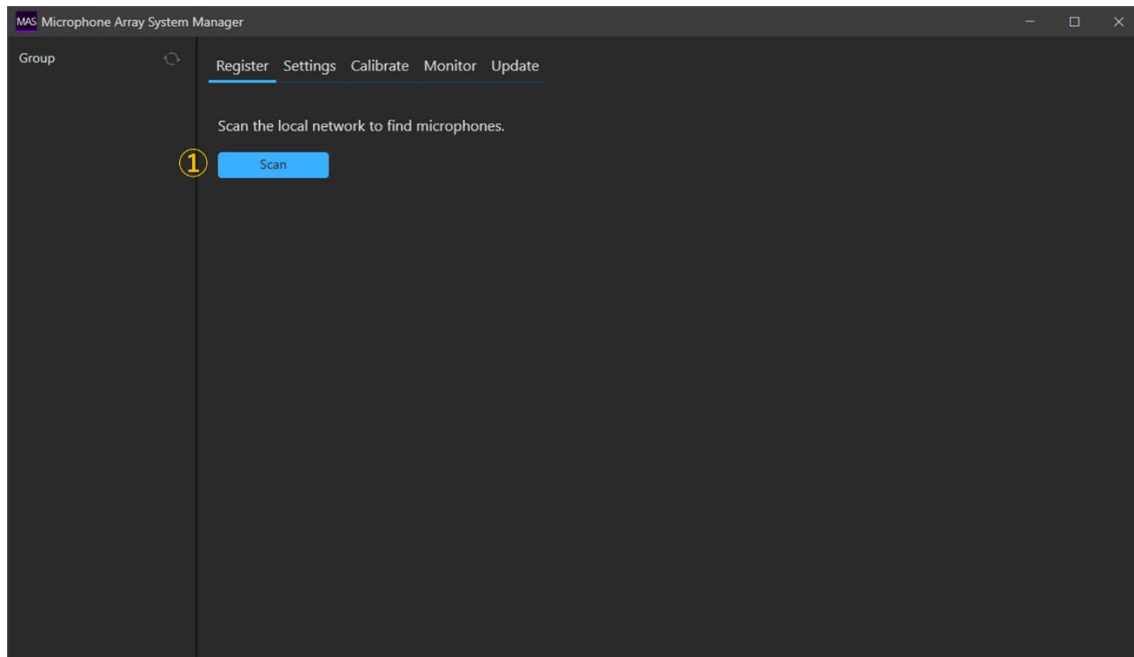
Change the IP address, subnet mask, and default gateway of the MAS-A100 units so that the MAS-A100 units are on the same subnet as the PC and the IP address can be accessed from the PC.

Do not configure an IP address already configured on another device.

If already on the same subnet as the PC, there is no need to change settings.

3. Press the [Set] button.

The IP address, subnet mask, and default gateway of the selected MAS-A100 units are applied. Proceed to the next step.



Initialization and user name/password entry

1. Place a check mark beside the MAS-A100 units to register in MASM-1.
2. Check the serial numbers.
3. To allow access to the MAS-A100 units from MASM-1, enter the user name and the password of the administrator for each MAS-A100 unit in [Username] and [password], respectively.

Placing a check mark in [admin] automatically sets [Username] to "admin".
Placing a check mark in [Same as this app.] configures the same password as the MASM-1 login password in [password].

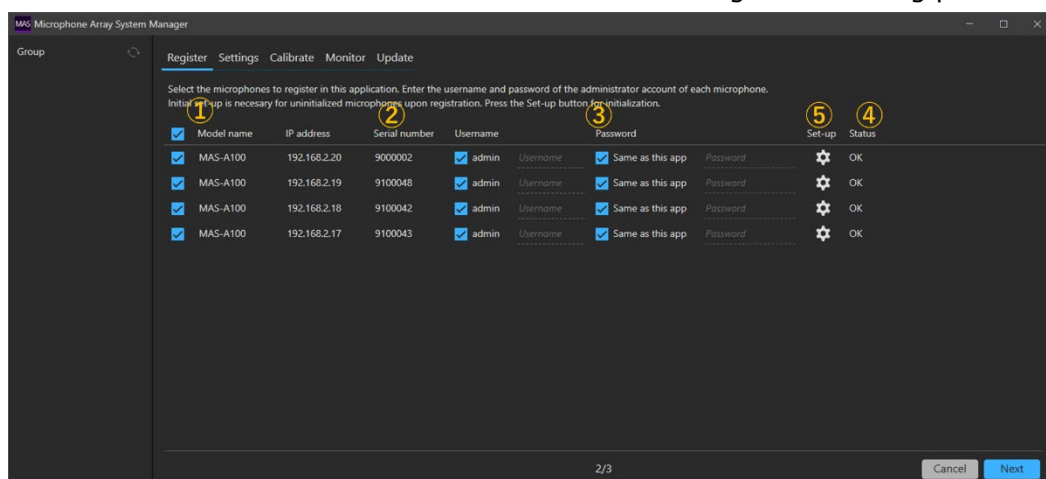
4. Check the [Status] display.

Only MAS-A100 units with [Status] of [OK] can be registered. For MAS-A100 units with other status, modify the settings as described below, or deselect the checkbox to skip registration.

[Status]	Description
[OK]	Ready to register. When all MAS-A100 units you want to register are in this state, proceed to step 16.
[Uninitialized]	MAS-A100 unit is in the factory default state. Perform the procedure from step 5 to initialize the MAS-A100.
[IP address is not accessible]	The IP address of the MAS-A100 is not accessible. Check the network connection, IP address setting of the MAS-A100, and HTTP proxy setting of the PC. To change the IP address of the MAS-A100, press [Cancel] to scan for MAS-A100 units again.
[Password is not correct]	User name or password is incorrect.

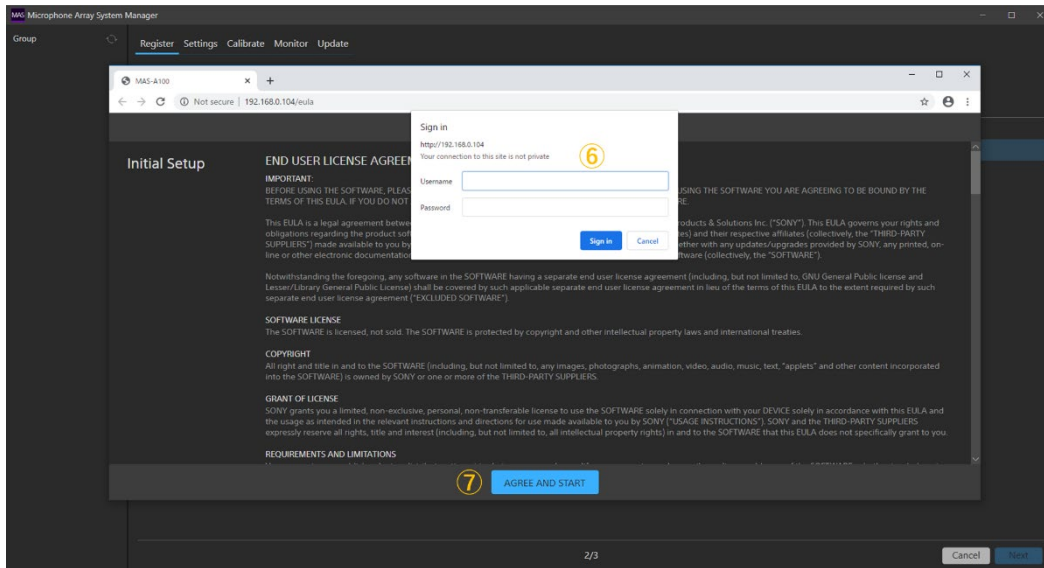
5. If the MAS-A100 unit is in factory default state, press the [Set-up] icon to start initialization of the MAS-A100.

Pressing the [Set-up] icon launches Google Chrome and displays the web UI of the MAS-A100. Perform initialization of each unit using the following procedure.



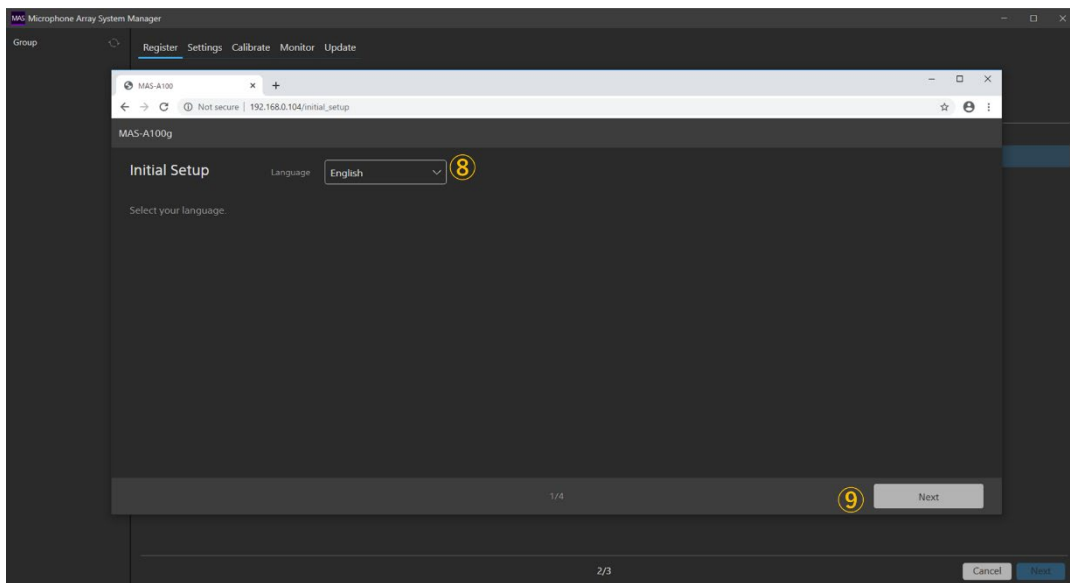
6. Log in using the default password (user name: admin, password: Admin_1234).

7. Press the [AGREE AND START] button.



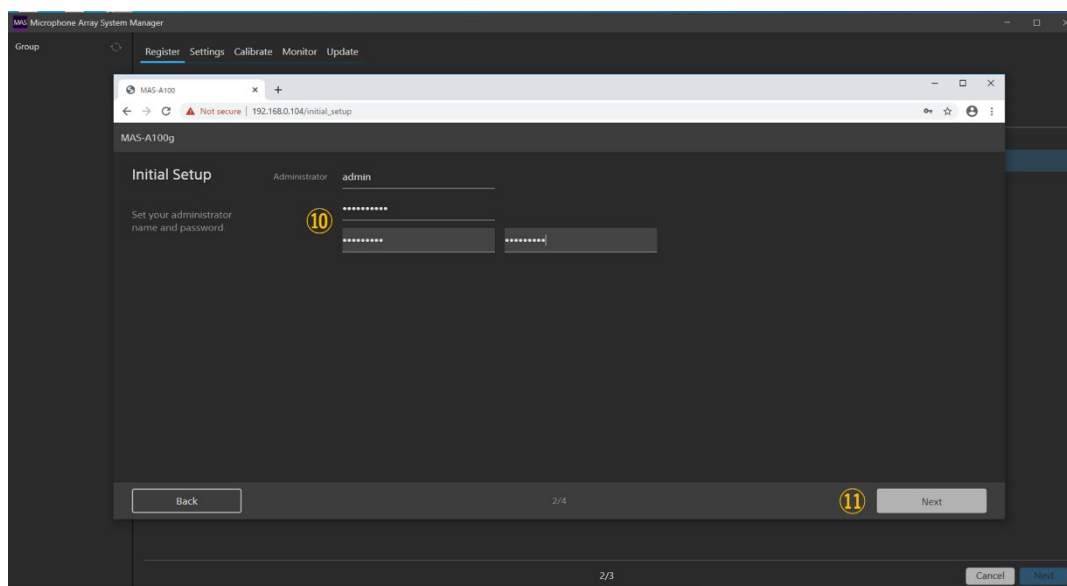
8. Select the display language.
9. Press the [Next] button.

The following screen appears.

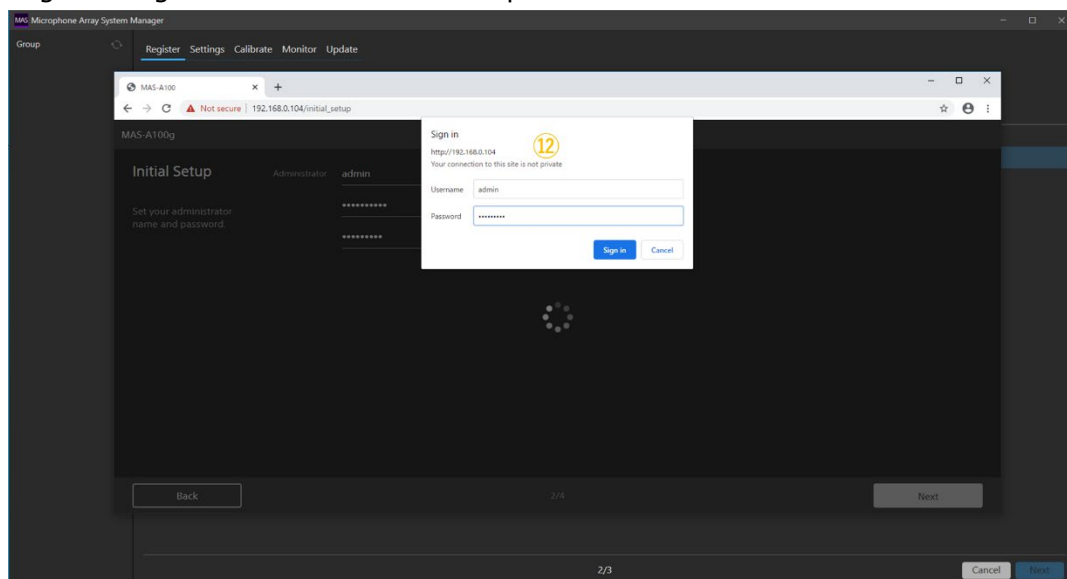


10. Change the user name and password of the administrator.
Set a user name of 5 to 16 characters, and a password of 8 to 64 characters.
The user name and password must contain at least one uppercase letter, one lowercase letter, and one numeral. Symbol characters can also be used.
! \$ % ` ( ) = - ~ ^ | @ ` [{ }] _ / ? < > + * .
Setting the user name to "admin" and the same password as the MAS-A100 allows for easy MAS-A100 registration.

11. Press the [Next] button.
The following screen appears.

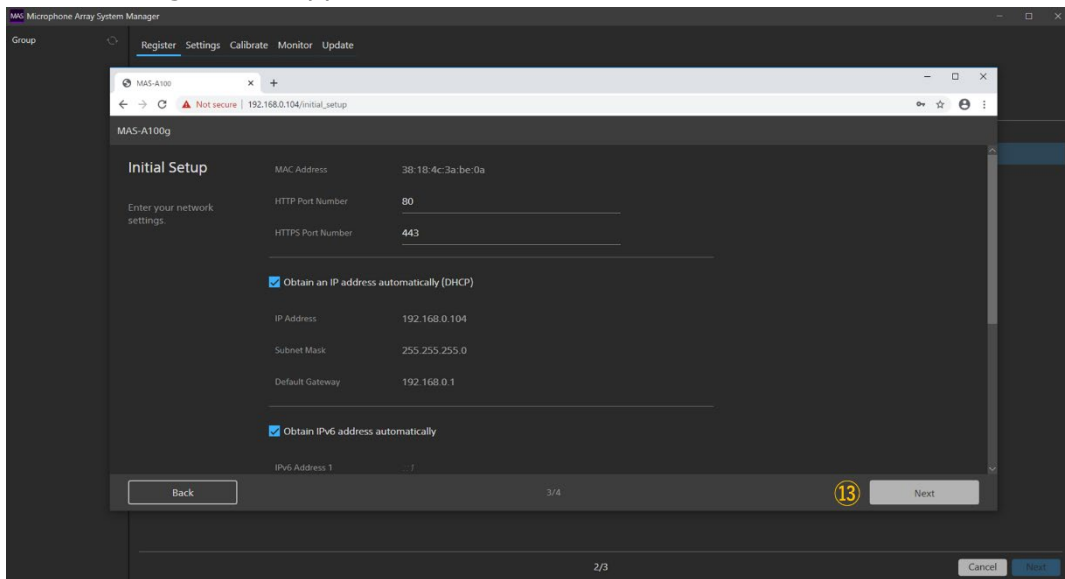


12. Log in using the new user name and password.



13. Change the network settings, and press the [Next] button.

The following screen appears.



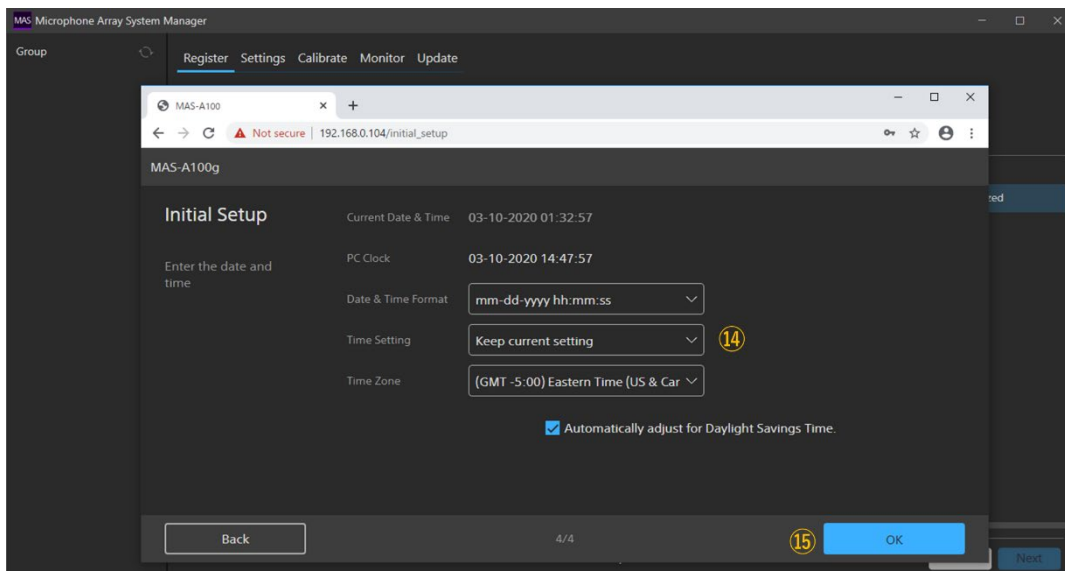
14. Set the date and time.

15. Press the [OK] button.

Initial setup is completed.

Note

Initial setup is completed regardless of whether or not you exit Google Chrome.



16. In MASM-1, press the [Next] button to proceed to the next step if all the MAS-A100 units with check marks display [OK] for [Status].

Registering MAS-A100 units in MASM-1

1. Place a check mark beside the MAS-A100 units to register in MASM-1.
2. Press the [Flash] icon to start the LED lamp of the MAS-A100 flashing to confirm its location, if necessary.
3. Set optional character strings in [Path 1], [Path 2], [Path 3], [Name], and [Description].

For example, enter a building name, a floor number, a room name, a name for identification, and a description.

See "Setting the name" (page 67) for the valid characters and maximum character string length.

[Path 2] must not be left blank if [Path 3] is specified.

Notes

- MASM-1 groups MAS-A100 units with the same entries for [Path 1], [Path 2], and [Path 3] on the [Group] pane, displaying a list of the MAS-A100 units in the same group. Accordingly, MAS-A100 units in the same room should have the same entries for [Path 1], [Path 2], and [Path 3] under normal circumstances.
 - It is recommended that the number of MAS-A100 units in a group be limited to 20 or lower to avoid slow response in MASM-1.
4. Press the [Done] button.
- MAS-A100 registration is completed.

Select the microphones to register in this application. Set Path 1, Path 2, Path 3, Name, and Description of each microphone.
Press Done button to complete registration.

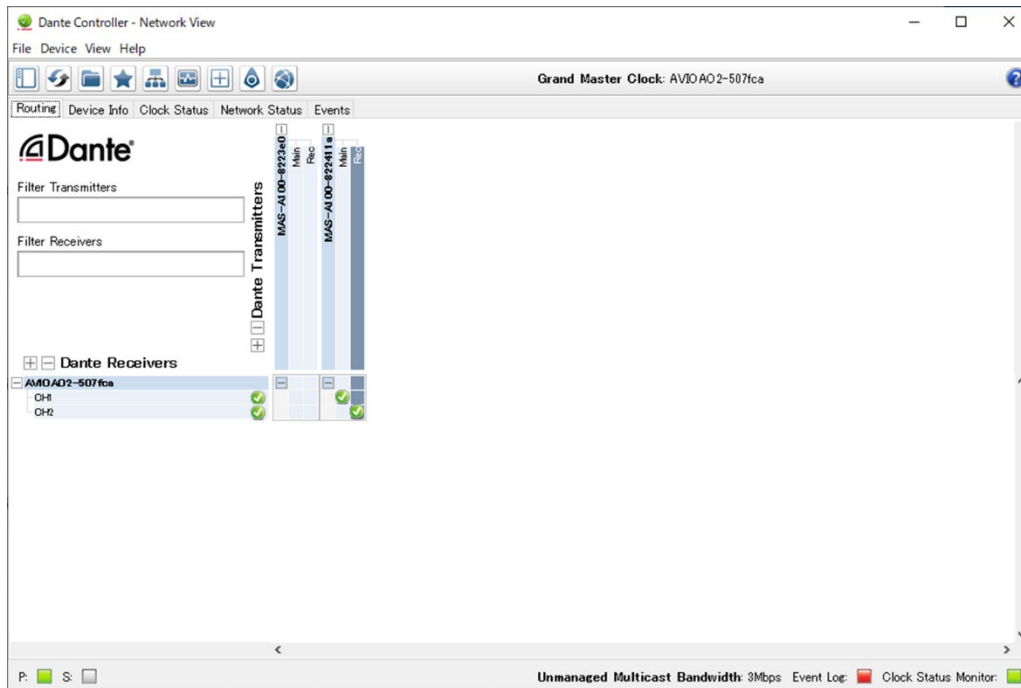
Model name	IP address	Flash	Path 1	Path 2	Path 3	Name	Description	Status
☒ MAS-A100	192.168.0.32		103G	6F		MAS-A100	Left	
☒ MAS-A100	192.168.0.8		Path1			MAS-A100		
☒ MAS-A100	192.168.0.43		Path1			MAS-A100		
☒ MAS-A100	192.168.0.42		103G	6F		MAS-A100	Right	
☒ MAS-A100	192.168.0.2		Path1			MAS-A100		
☒ MAS-A100	192.168.0.5		Path1			MAS-A100		
☒ MAS-A100	192.168.0.15		Path1			MAS-A100		
☒ MAS-A100	192.168.0.34		Path1			MAS-A100		
☒ MAS-A100	192.168.0.78		Path1			MAS-A100		

3/3

Cancel Back Done

5.3. Dante Controller preparation

Dante Controller is a free software application from Audinate for routing audio and configuring devices on a Dante network.



Download

The software can be downloaded from the following website.

<https://www.audinate.com/products/software/dante-controller>

Operating environment

Windows and macOS are supported.

Usage

For details about using Dante Controller, refer to the Dante Controller user guide.

During setup, observe the following points.

Firewall

Dante Controller automatically configures the required firewall settings of Windows and macOS during installation for Dante Controller operation.

However, if a third-party firewall application such as McAfee, Trendmicro, or Norton is used, it is necessary to temporarily disable or configure the firewall application manually to allow communication by Dante Controller.

IP address

Dante Controller discovers Dante devices on the same segment (same layer 2 network) and shows the list of the discovered devices on the [Device Info] tab.

Device Name	Model Name	Product Version	Dante Version	Device Lock	Primary Address	Primary Link Speed	Secondary Address	Secondary Link Speed
AVIO-AAAAAAAAAAAAAAAAAAAAA	AVIO-USB	1.1.0	4.1.5.3	☐	192.168.0.29	100Mbps	N/A	N/A
aviohub-507C03m	AVIO-USB	1.1.0	4.1.5.3	☐	192.168.0.45	100Mbps	N/A	N/A
AVIOUSB-507d03	AVIO-USB	1.1.0	4.1.5.3	☐	192.168.0.14	100Mbps	N/A	N/A
AVIOUSB-507d0d	AVIO-USB	1.1.0	4.1.5.3	☐	192.168.0.22	100Mbps	N/A	N/A
AVIOUSB-SANO	AVIO-USB	1.1.0	4.1.5.3	☐	192.168.0.19	100Mbps	N/A	N/A
MAS-A100-819343	MAS-A100	0.0.3	4.1.5.7	☐	192.168.0.30	1Gbps	N/A	N/A
MAS-A100-822311	MAS-A100	0.0.3	4.1.5.7	☐	192.168.0.13	1Gbps	N/A	N/A
MAS-A100-822405	MAS-A100	0.0.3	4.1.5.7	☐	192.168.0.48	1Gbps	N/A	N/A
MAS-A100-82240b	MAS-A100	0.0.3	4.1.5.7	☐	192.168.0.47	1Gbps	N/A	N/A
MAS-A100-822410	MAS-A100	0.0.3	4.1.5.7	☐	192.168.0.33	1Gbps	N/A	N/A
MAS-A100-822411	MAS-A100	0.0.3	4.1.5.7	☐	192.168.0.18	1Gbps	N/A	N/A
MAS-A100-822413	MAS-A100	0.0.3	4.1.5.7	☐	192.168.0.1	1Gbps	N/A	N/A
MXA910-f9aeds	MXA910		3.9.7.2	N/A	192.168.0.6	1Gbps	N/A	N/A
UXT-AERO-81933e	MAS-A100	MAS-A100	4.1.4.2	☐	192.168.0.23	1Gbps	N/A	N/A
UXT-AERO-819345	MAS-A100	MAS-A100	4.1.4.2	☐	192.168.0.4	1Gbps	N/A	N/A
UXT-AERO-819352	MAS-A100	MAS-A100	4.1.4.2	☐	192.168.0.24	1Gbps	N/A	N/A
UXT-AERO-GENRQ39	MAS-A100	MAS-A100	4.1.4.2	☐	192.168.0.39	1Gbps	N/A	N/A
AVIO-A02-507fca	AVIO-DAQ2	1.1.0	4.1.5.2	☐	192.168.1.100	100Mbps	N/A	N/A

Dante Controller configures only Dante devices on the same subnet. Dante devices on the same segment (same layer 2 network) but different subnet cannot be configured. The name of Dante devices that cannot be configured is displayed in red under [Device Name] on the [Device Info] tab. They are not displayed on the [Routing] tab.

Set the IP address of the Dante devices and the PC, where Dante Controller is running, so that they are on the same subnet. For Dante devices, you can change the IP address acquisition method in [Device View] – [Network Config] of Dante Controller.

It can be configured using the following three methods according to the installation environment.

Addresses	Connected network environment	IP address setting
Obtain an IP Address Automatically (default)	Using DHCP	DHCP automatically assigns the IP address of Dante devices. Set the PC running Dante Controller to obtain IP address from DHCP automatically.
	Without DHCP	The IP addresses of Dante devices are automatically assigned in the range of 169.254.0.0/16 by Auto IP. Set the IP address of the PC running Dante Controller in the range of 169.254.0.0/16 manually.
Manually configure an IP Address		You can set the IP address of a Dante device manually using [Device View] – [Network Config] tab of Dante Controller. After configuration, change the setting of the IP address of the PC running Dante Controller so that it is also on the same subnet.

5.4. Routing of Dante audio

Configure the routing of the audio signals between the Dante devices (register the logical connection to the Dante devices).

This section describes routing the output of the MAS-A100 to a Dante receiver device (digital mixer or Dante/Analog converter).

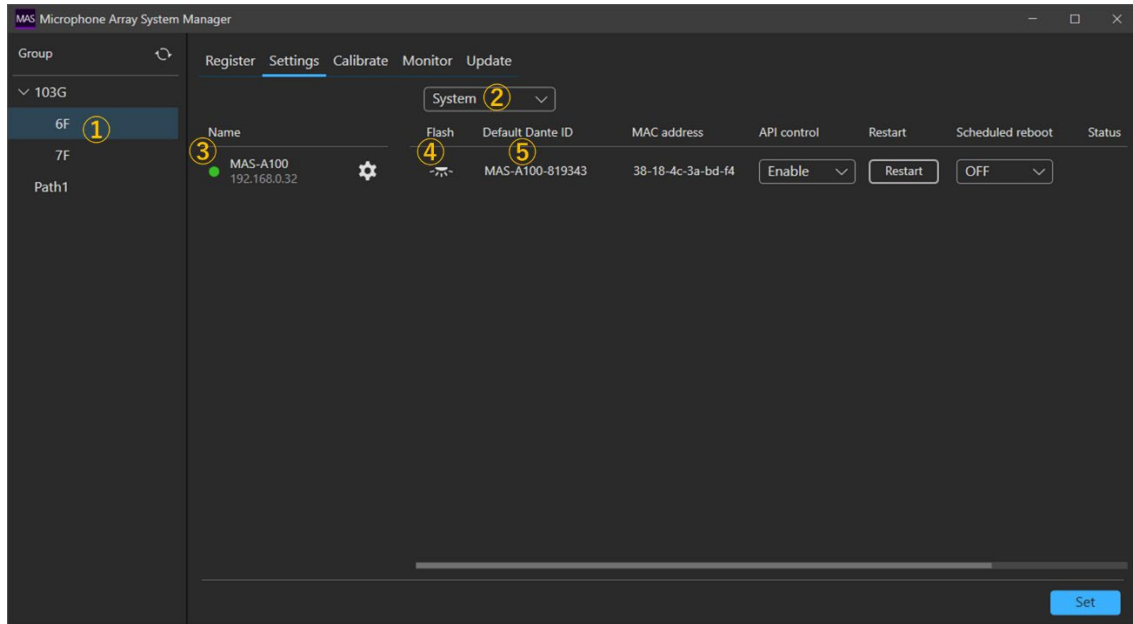
Routing is performed using Dante Controller installed on a PC connected to the same segment (same layer 2 network) as the MAS-A100 units in a room where the MAS-A100 units are installed.

[Default Dante ID] check

1. Open the [Settings] tab of MASM-1, and select any group in the left pane.
2. Select [System] from the drop-down menu in the right pane.
3. Check the MAS-A100 status using the status lamp.

If a lamp is gray, the MAS-A100 is offline and the [Default Dante ID] is not displayed.

4. Press the [Flash] icon to start the LED lamp of the MAS-A100 flashing to confirm its location, if necessary.
5. Check the [Default Dante ID] of the MAS-A100.



[Default Dante ID]

[Default Dante ID] is the Dante device name of the MAS-A100 in factory default state, and consists of the characters "MAS-A100-xxxxxxx" (where xxxxxx is a unique alphanumeric 6-character string).

Note

Even if you change the Dante device name of the MAS-A100 using Dante Controller, the [Default Dante ID] displayed in the web UI of the MAS-A100 and in MASM-1 does not change.

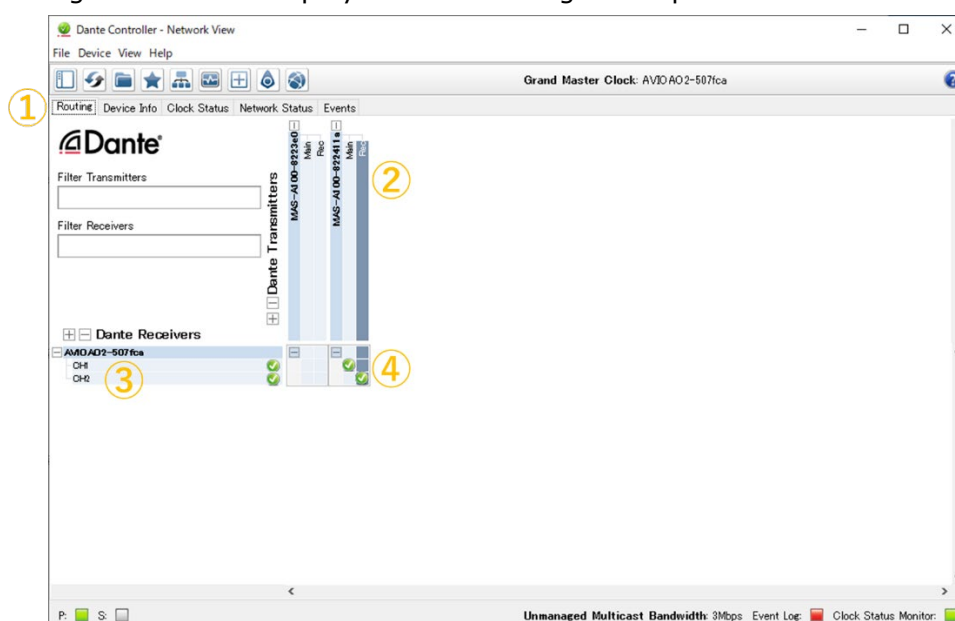
It is recommended that you do not change the Dante device name of the MAS-A100, as doing so will mean it is not possible to configure the MAS-A100 according to this document. Specifically, changing the unique alphanumeric 6-character string suffix of the Dante device name will make it difficult to identify the MAS-A100 units on the network.

Routing

1. Open the [Routing] tab of Dante Controller.
2. Find the MAS-A100 unit with the matching Default Dante ID noted in "[Default Dante ID] check" in the list of Dante devices displayed in [Dante Transmitters], and check that it has two channels (Main and Rec).
3. Find the Dante receiver device (e.g., digital mixer with Dante inputs, or Dante/Analog converter) to be connected in the list of Dante devices displayed in [Dante Receivers].
4. Place check marks in the output/input matrix to route the Main output of the MAS-A100 to the speech reinforcement system, and the Rec output of the MAS-A100 to the recording system.

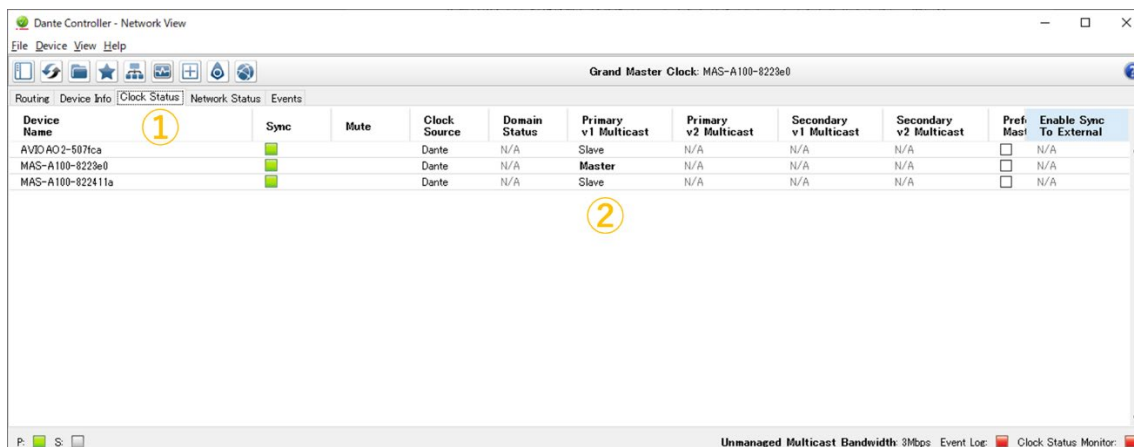
The following image shows an example where CH1 and CH2 of a Dante/Analog converter are routed to the speech reinforcement system and the recording system, respectively.

A green mark is displayed when routing is completed.



Clock check

1. Open the [Clock Status] tab of Dante Controller.
2. Check that only one Dante device displays "Master" in [Primary V1 Multicast].
If more than one Dante device displays "Master", there is a problem with the clock of the Dante network, and Dante audio cannot be transmitted correctly.



5.5. Calibration

Calibration is necessary before using the Main output of the MAS-A100 for speech reinforcement. Calibration simulates the positional relationship of MAS-A100 units with the loudspeakers and the acoustic characteristics of a room to optimize the audio for the Main output. Calibration is required when the MAS-A100 units are installed, when the loudspeaker or MAS-A100 position is changed, or the settings for the audio system is changed.

If the volume and audio quality settings (master volume, microphone input gain, equalizer settings) of the audio system in which the MAS-A100 units are connected are changed, the units will be out of calibration and may not operate normally.

If the installation location or audio system changes significantly or the intended use changes, reset the MAS-A100 units to the factory default state (see "6.3 Factory reset" (page 81)), initialize the units again, and then perform calibration.

Tips on calibration

Remain quiet during calibration. Random noises may affect the optimization. Calibration where the MAS-A100 unit is located may not be successful in a noisy environment (such as exhibition hall, noisy classroom, air conditioning, outdoor construction work) where the noise level exceeds 40 dB (A weighted).

Set the audio system for speech reinforcement not to use any non-linear signal processing (such as noise reduction or automatic gain control where the gain is automatically reduced in response to input noise) for the input from the MAS-A100. If such a process is activated, calibration using noise cannot be performed normally.

During the calibration of an MAS-A100, other MAS-A100 units for speech reinforcement in the same audio system should be muted. MASM-1 automatically switches calibration and muting of MAS-A100 units one at a time when calibrating a selection of multiple MAS-A100 units. All microphones other than the MAS-A100 units should be muted (low volume or switched off) during calibration.

Recalibration is required in the following circumstances.

- Change in speech reinforcement loudspeakers
- Change in position or direction of speech reinforcement loudspeakers
- Significant changes in room acoustic characteristics due to size, shape, or interior materials of the room
- Changes in volume and audio quality settings (master volume, microphone input gain, equalizer settings) of the audio system in which the MAS-A100 units are connected

Re-calibration is not required in the following circumstances.

- Movement of the audience in and out
- Changes to furniture (desks, chairs)
- Other changes that do not significantly affect the acoustic characteristics of a room
- Changes to audio settings of the MAS-A100 (see "5.7 Audio settings").

Calibration preparation

Set the master volume of the audio system for speech reinforcement, to which the Main output of the MAS-A100 is connected, to the default position. The default position is the level used for normal operation with other audio input devices, such as lavalier microphones, hand microphones, video conference systems, DVD players, and PCs.

Set the audio system for speech reinforcement not to use any non-linear signal processing, such as noise reduction or automatic gain control, for the input from the MAS-A100.

1. Open the [Calibrate] tab of MASM-1, and select any group in the left pane.
2. Check the MAS-A100 status using the status lamp.

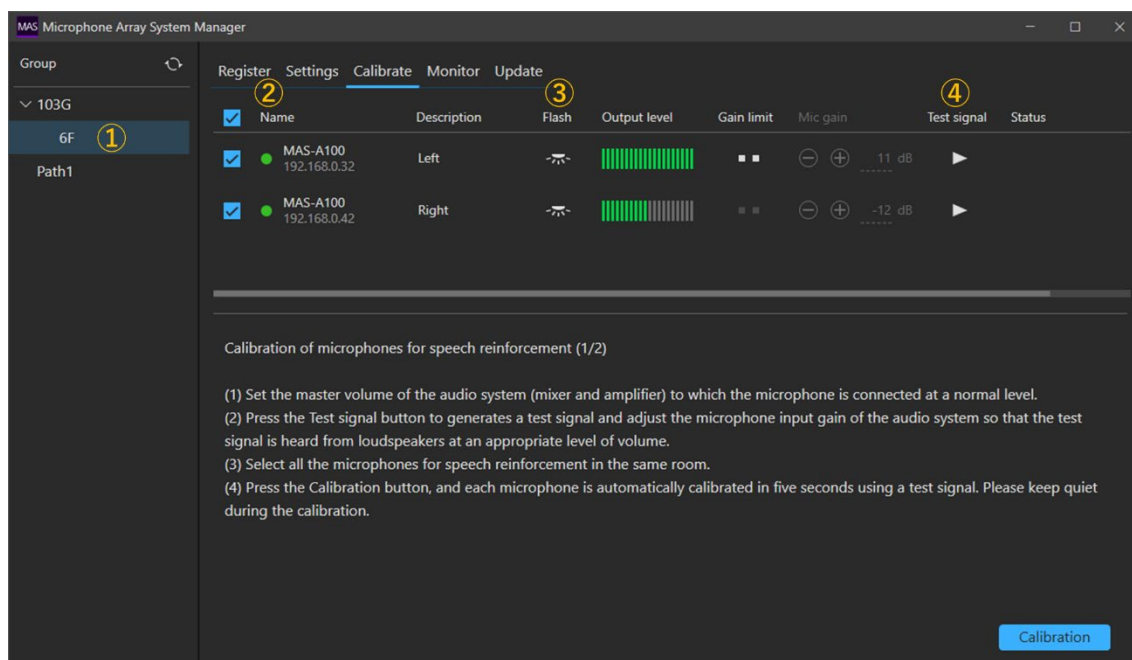
MAS-A100 units that display a gray lamp are offline and cannot be calibrated.

MAS-A100 units that display a green lamp (Active), amber lamp (Mute), or red lamp (Standby) can be calibrated. After calibration is completed successfully, the MAS-A100 switches to Active state.

3. Press the [Flash] icon to start the LED lamp of the MAS-A100 flashing to confirm its location, if necessary.
4. Press the [Test signal] icon of an MAS-A100 used for speech reinforcement to output a test signal.

The test signal is a voice saying "Test, one, two, three" at an output level (−16 dBFS) equivalent to speaking at a volume of 63 dB SPL (A) from a distance of 1 m in an environment where no feedback occurs.

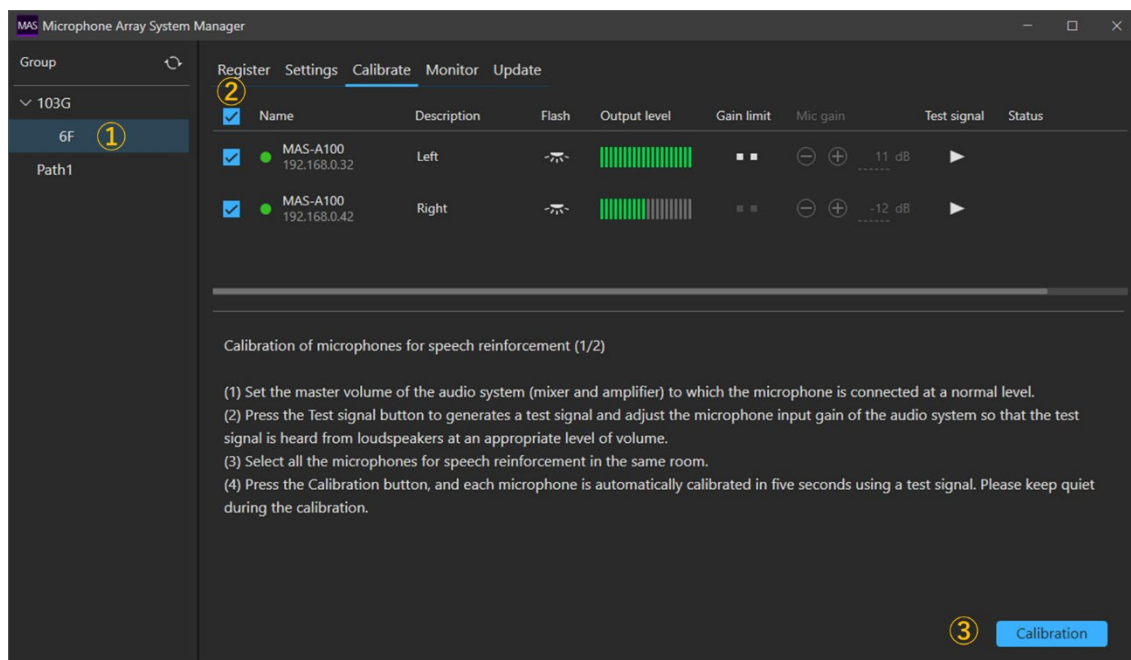
Press the [Test signal] icon again to stop the test signal. Play the test signal through the loudspeakers of the audio system and adjust the MAS-A100 input gain of the audio system so that the signal can be heard with the appropriate level of volume.



Calibrating

1. Open the [Calibrate] tab of MASM-1, and select any group in the left pane.
2. Place a check mark beside the MAS-A100 units to use for speech reinforcement. If there are multiple MAS-A100 units in the same room to be used for speech reinforcement that are connected to the same audio system, select those units and calibrate all of them together.
3. Press the [Calibrate] button.
Calibration starts, and a noise signal is output from the MAS-A100 for about 5 seconds.

The MAS-A100 captures the noise signal played from the loudspeakers, and this is used to simulate the positional relationship with loudspeakers and the acoustic characteristics of the room to optimize the audio processing for feedback reduction. For multiple MAS-A100 units, each microphone outputs the noise signal in sequence.



4. Check the calibration results.

When calibration is completed, the results are displayed in the [Status] field. A 5-level rating ([Excellent (5)], [Good (4)], [Average (3)], [Fair (2)], [Poor (1)]) is displayed. The higher the rating value, the higher the limit for speech reinforcement of the audio.

If the rating is [Fair (2)] or [Poor (1)], review the installation environment of the MAS-A100 (see "4.2 Installation" (page 16)) and the audio system settings ("4.4 Audio design" (page 30)), and then perform calibration again.

If the MAS-A100 audio is not emitted from the speech reinforcement loudspeakers or the Rec output is connected by mistake, the calibration sound will not be heard and the result will be [Excellent (5)] even though the calibration has not been completed successfully. Check the connections and perform the calibration again.

5. Adjust the microphone gain.

After calibration, the microphone gain is set to -12 dB. Speak normally, within the capture range of the MAS-A100, and increase the microphone gain for the appropriate speech reinforcement level.

6. Check the gain limit.

The gain limit lamps may be lit as the microphone gain increases.

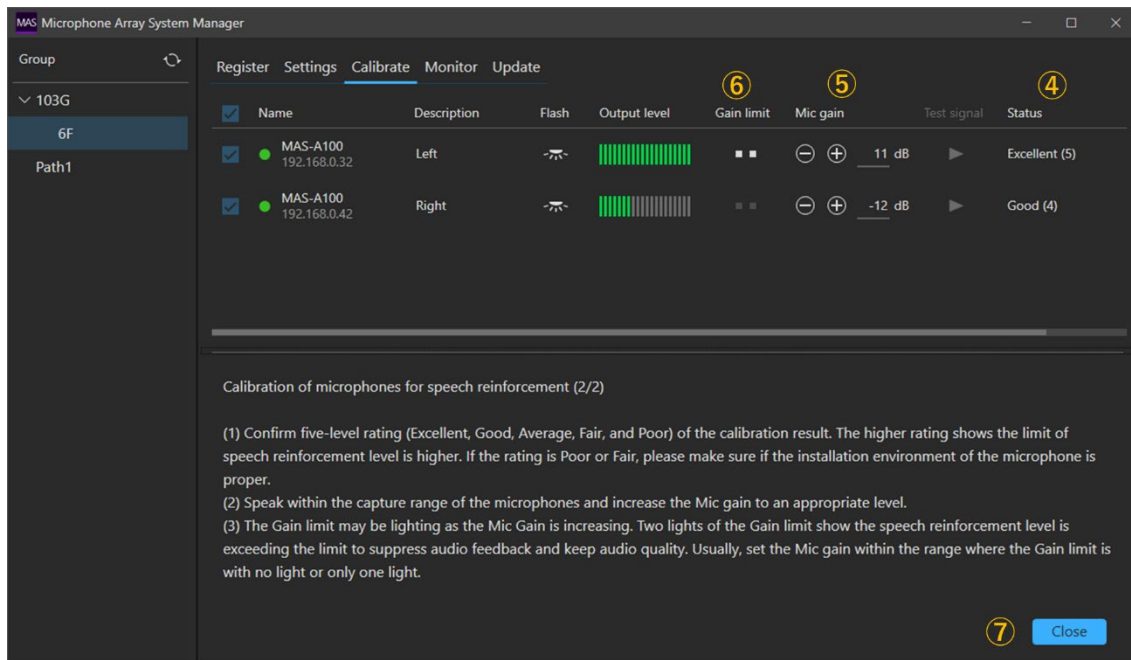
To avoid deterioration in audio quality, set the microphone gain in the range so that the gain limit lamps are not lit or that only one lamp is lit.

A single gain limit lamp indicates that the speech reinforcement level is reaching the limit to suppress audio feedback while maintaining audio quality. Two gain limit lamps indicate that the speech reinforcement level has exceeded the limit and audio quality has deteriorated.

If a larger speech reinforcement level is needed, review the MAS-A100 installation environment and calibrate again.

7. Press [Close].

The microphone gain is set as the default, and calibration ends.



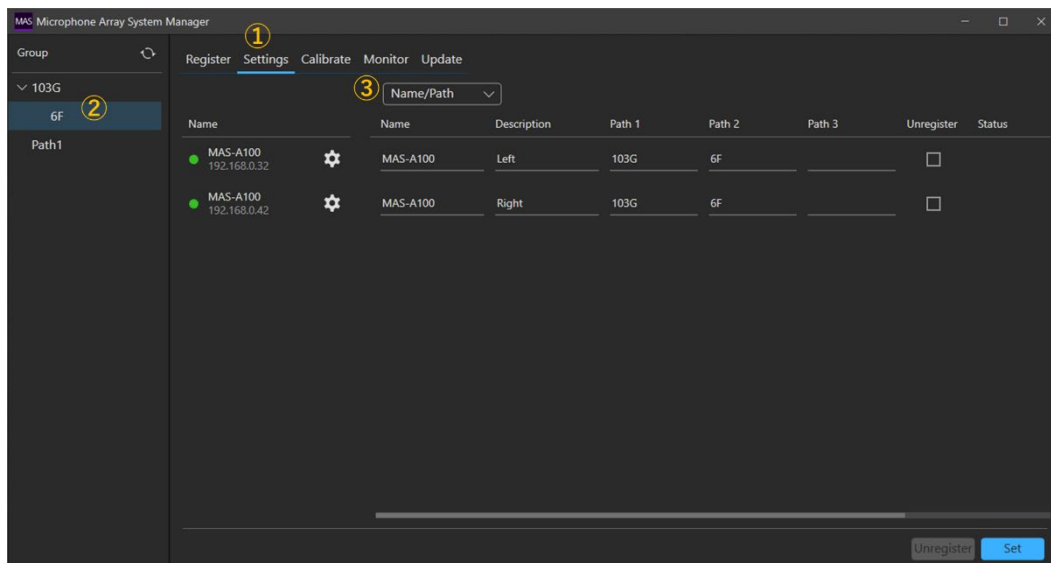
5.6. General settings

Use MASM-1 to set the name and installation location of MAS-A100 units, cancel registration, and set up MAS-A100 access for the web UI.

Displaying the settings screen

1. Open the [Settings] tab of MASM-1.

2. Select any group in the left pane.
3. Select [Name/Path] from the pull-down menu.



Checking the MAS-A100 status

The MAS-A100 status is indicated by lamp color.

MAS-A100 units that display a gray lamp are offline and cannot be configured.

MAS-A100 units that display a green lamp (Active), amber lamp (Mute), or red lamp (Standby) can be configured.

Configuring MAS-A100 settings via the web UI

Press the gear wheel icon to launch Google Chrome with the web UI of the MAS-A100.

Log in using the MAS-A100 user name and password, and then configure settings.

Setting the name

Check or edit the contents of [Name], [Description], [Path 1], [Path 2], and [Path 3], as required.

Set [Path 1] and [Name] to a character string of 4 to 16 characters.

Set [Path 2], [Path 3], and [Description] to an empty string or a character string of up to 16 characters.

Valid characters are alphabetic, numeric, and symbol characters.

! " ` () * + - . _ < >

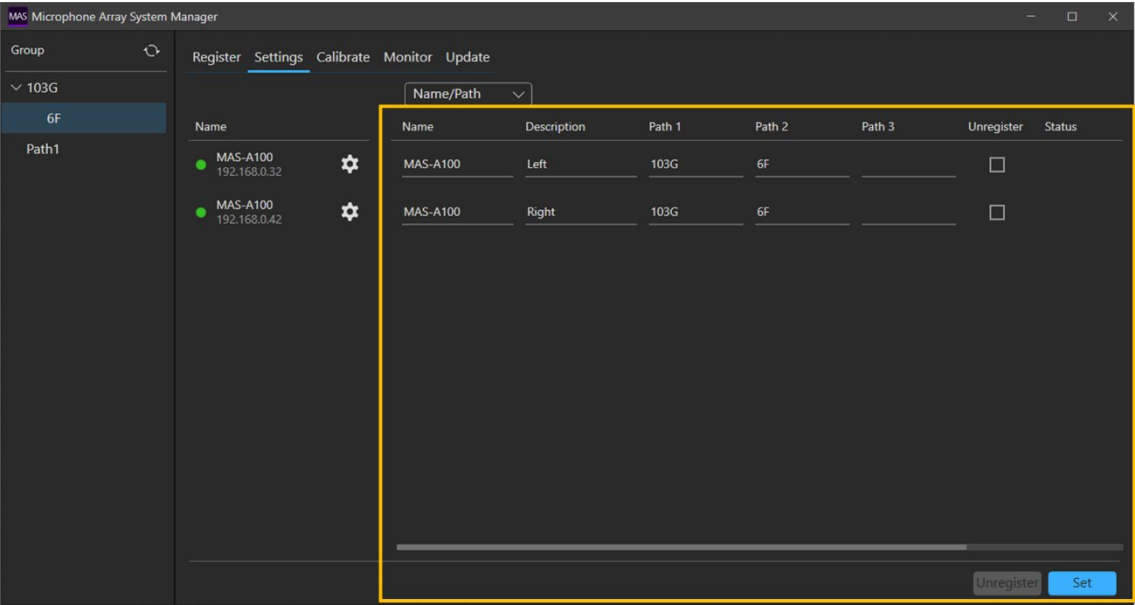
[Path 2] must not be left blank if [Path 3] is specified.

Press the [Set] button to save the changes.

Note

MASM-1 groups MAS-A100 units with the same entries for [Path 1], [Path 2], and [Path 3] on the [Group] pane, displaying a list of the MAS-A100 units in the same group. Accordingly, MAS-A100 units in the same room should have the same entries for [Path 1], [Path 2], and [Path 3] under normal circumstances.

It is recommended that the number of MAS-A100 units in a group be limited to 20 or lower to avoid slow response in MASM-1.



Canceling registration

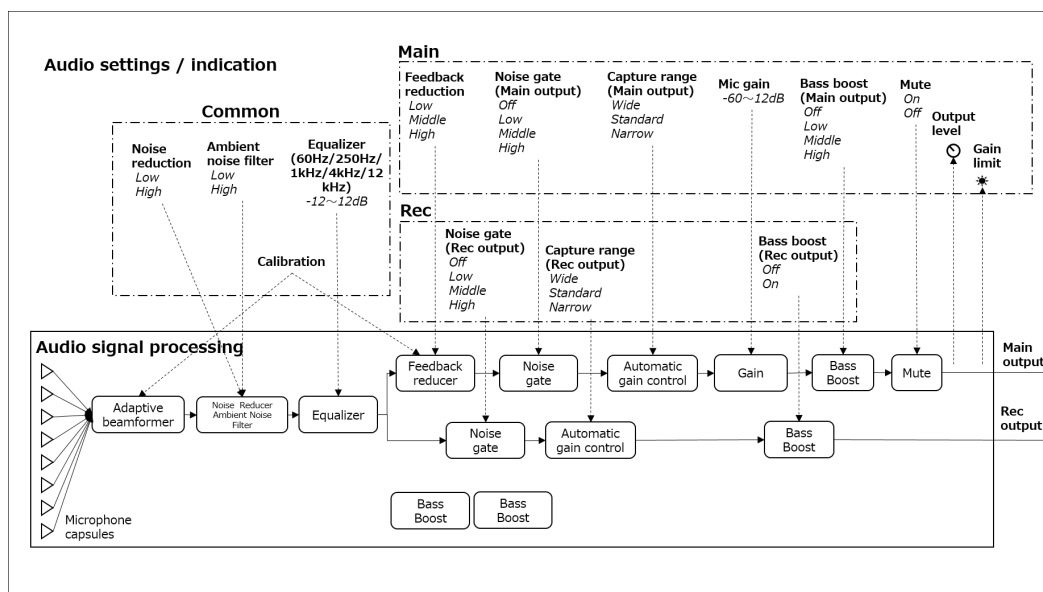
Place a check mark beside MAS-A100 units that you want to unregister, and press the [Unregister] button to unregister the selected MAS-A100 units.

5.7. Audio settings

This section describes the audio settings and display indications.

Audio settings and signal processing functions

The relationship between the settings and display of the audio setting screen for the signal processing function of the MAS-A100 is shown below.



The software that can be used for configuration and changes is indicated by a symbol.

Symbol	
MASM	Display can be set on the MASM-1 settings screen
Web	Display can be set on the web UI screen of the MAS-A100 using a web browser
API	Settings can be accessed by a device that supports the MAS-A100 external control API

Noise reduction

Name: Noise reduction

Target: Common for both Main output and Rec output

Settings: Low, Middle, High

Applicable software: **MASM** **Web**

Function description

Reduces steady-state noise in a classroom or a meeting room (e.g., fan noise of air conditioners and projectors).

The effect is weakest when set to [Low], and strongest when set to [High].

The stronger the effect, the greater the noise reduction, but voices may become distorted or artificial noise called musical noise may be heard.

Ambient noise filter

Name: Ambient Noise Filter

Target: Common for both Main output and Rec output

Settings: Off, Low, High

Applicable software: [Web](#)

Function description

The ambient noise filter is a function that reduces sporadic noise (such as keyboard-type noise and paper-rustling noise) that cannot be reduced by normal noise reduction.

When set to [High], the noise reduction is stronger, but it may adversely affect the audio quality.

Equalizer

Name: Equalizer

Target: Common for both Main output and Rec output

Settings: 60 Hz, 250 Hz, 1 kHz, 4 kHz, and 12 kHz bands with -12 dB to +12 dB range adjustable in 1 dB increments

Applicable software: [MASM](#) [Web](#)

Function description

5-band graphic equalizer. Use to adjust the sound quality.

Wind noise and vibration noise are reduced by lowering the [60 Hz] gain.

Vocal thickening and warmth are increased by raising the [250 Hz] gain.

Boost the sound so that the voice stands out by raising the [1 kHz] gain. Surrounding noise may also increase.

Suppress noise and soften the sound for easier listening by lowering the [4 kHz] and [12 kHz] gain.

Adjusting may cause side effects, so make adjustments while checking the sound quality.

Calibration

Name: Calibration

Target: Common for both Main output and Rec output

Settings: None (starts calibration)

Applicable software: **MASM** **Web**

Function description

Runs beamforming and feedback canceler calibration.

For details of usage, see "5.5 Calibration" (page 62).

Feedback reduction

Name: Feedback reduction

Target: Main output

Settings: Low, Middle, High

Applicable software: **MASM** **Web**

Function description

Enables/disables the feedback reduction function and controls the level of feedback reduction.

When feedback occurs due to close proximity of the MAS-A100 and loudspeakers or due to calibration not performed normally, the frequency of the feedback is suppressed by lowering the gain to stop the feedback.

Setting to [High] achieves greater feedback reduction, but decreases the audio quality.

To ensure better speech reinforcement sound quality, it is recommended that the MAS-A100 and loudspeakers be installed so that feedback does not occur when using the [Middle] and [Low] settings.

Bass boost (for Main)

Name: Bass Boost

Target: Main output

Settings: Off, Low, Middle, High

Applicable software: **Web**

Function description

Bass boost is a function that enhances the low frequency range of the captured audio. When set to [High], the low frequency range of the audio is strongly enhanced.

Use this when you feel that the loudspeaker volume is insufficient.

Depending on the usage environment, the audio clarity may be reduced due to the enhancement of the low range.

Noise gate (for Main)

Name: Noise gate

Target: Main output

Settings: Off, Low, Middle, High

Applicable software: MASM Web

Function description

Noise gate stops the audio output when the input level is low.

Even low-level input audio is output when set to [Low], but only high-level input audio is output when set to [High]. Usually, set this to [Off].

If the level of noise reduction achieved using [Noise reduction] is insufficient, set this to [Low], [Middle], or [High]. For example, use when there is random, low volume noise (footsteps in the hallway, whispering voices, etc.).

Note that the beginning of speech may be lost due to the slight delay in detecting sound input and deactivating the noise gate.

Installation in locations that require this feature to be used at all times is not recommended. Take measures to reduce the noise, such as moving the MAS-A100 away from the noise source and lowering the noise level in the room.

Capture range (for Main)

Name: Capture range

Target: Main output

Settings: Narrow, Standard, Wide

Applicable software: MASM Web

Function description

Adjusts the capture range (so that even low voices are captured). The capture range for general speech is about 2 m to 3 m radius when set to [Standard].

Setting to [Narrow] makes the capture range narrow, but the upper limit of speech reinforcement becomes higher.

Setting to [Wide] makes the capture range wider, but the upper limit of speech reinforcement becomes lower.

For high-quality speech reinforcement, set to [Standard] or [Narrow] and ask the speaker to speak with a loud voice or have the speaker stand close to the MAS-A100. When using [Wide], maintain a large distance between the MAS-A100 and the loudspeakers, and reduce the sound pressure entering the MAS-A100 from the loudspeakers as much as possible.

Microphone gain

Name: Mic gain

Target: Main output

Settings: -60 dB to 12 dB

Applicable software: **MASM** **Web** **API**

Function description

Controls the microphone gain (output level). This feature allows you to adjust the output level without affecting the calibration.

The MAS-A100 is equipped with an AGC function that maintains the volume of the captured audio at a constant level. The microphone gain sets the AGC target level.

By raising the output level, the sound pressure output from the loudspeakers will increase, but feedback is more susceptible to occur and the sound quality also deteriorates.

Muting

Name: Mute

Target: Main output

Settings: Off, On

Applicable software: **MASM** **Web** **API**

Function description

When set to [On], the Main output is stopped.

When set to [Off], the Main output audio is output.

Output level display

Name: Output level

Target: Main output

Settings: None (display only)

Applicable software: **MASM** **Web**

Function description

Displays the output level of the Main output.

Gain limit display

Name: Gain limit

Target: Main output

Settings: None (display only)

Applicable software: **MASM** **Web**

Function description

Displays the control state of the Main output.

One gain limit lamp indicates that the speech reinforcement level is reaching the limit to suppress audio feedback while maintaining audio quality.

Two gain limit lamps indicate that the speech reinforcement level is exceeding the limit.

For details, see "5.5 Calibration" (page 62).

Bass boost (for Rec)

Name: Bass Boost

Target: Rec output

Settings: Off, On

Applicable software: **Web**

Function description

Bass boost is a function that enhances the low frequency range of the captured audio. When set to [On], the low frequency range of the audio is strongly enhanced.

Soft sound quality is achieved by enhancing the low range, but depending on the usage environment, the audio clarity may be reduced due to the enhancement of the low range.

Noise gate (for Rec)

Name: Noise gate

Target: Rec output

Settings: Off, Low, Middle, High

Applicable software: **Web**

Function description

Noise gate stops the audio output when the input level is low.

Even low-level input audio is output when set to [Low], but only high-level input audio is output when set to [High]. Usually, set this to [Off].

If the level of noise reduction achieved using [Noise reduction] is insufficient, set this to [Low], [Middle], or [High]. For example, use when there is random, low volume noise (footsteps in the hallway, whispering voices, etc.).

Note that the beginning of speech may be lost due to the slight delay in detecting sound input and deactivating the noise gate.

Installation in locations that require this feature to be used at all times is not recommended. Take measures to reduce the noise, such as moving the MAS-A100 away from the noise source and lowering the noise level in the room.

Capture range (for Rec)

Name: Capture range

Target: Rec output

Settings: Narrow, Standard, Wide

Applicable software: **Web**

Function description

Adjusts the capture range (so that even low voices are captured). The capture range for general speech is about 7 m radius when set to [Standard].

When set to [Narrow], it is difficult to capture low voices.

When set to [Wide], low voices will be captured, but noise may also be captured if present.

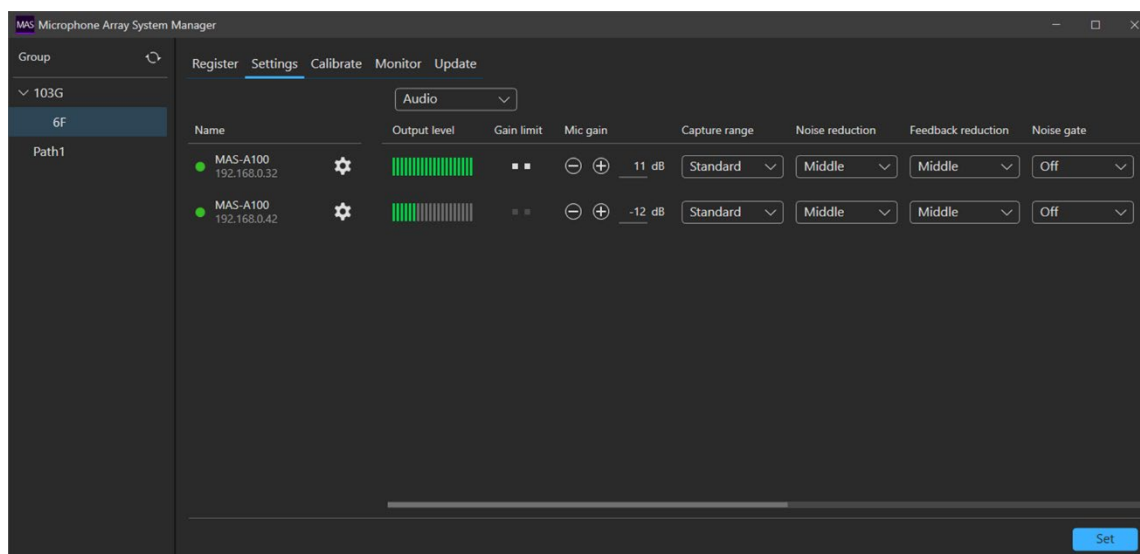
If there is a noise source within the audio capture range of the MAS-A100, set to [Narrow].

MASM-1 audio settings

1. Open the [Settings] tab of MASM-1.
2. Select any group in the left pane.
3. Select [Audio] from the pull-down menu.
4. Configure settings and then press the [Set] button.

The settings are saved.

Note that the microphone gain and equalizer settings are applied and saved immediately upon being changed.



5.8. System settings

1. Select [System] from the pull-down menu.
2. Press the [Flash] icon to start the LED lamp of the MAS-A100 flashing to confirm its location, if necessary.
3. Check the [Default Dante ID].
4. Check the MAC address.

This MAC address is used for MAS-A100 setup and operation functions. It is not the MAC address as a Dante device. The MAC address as a Dante device can be checked using Dante Controller.

5. Check the serial numbers.
6. Set whether to enable control from an external device.

For details, see "Appendix B. External Control" (page 112).

7. Press the [Restart] button, as required.

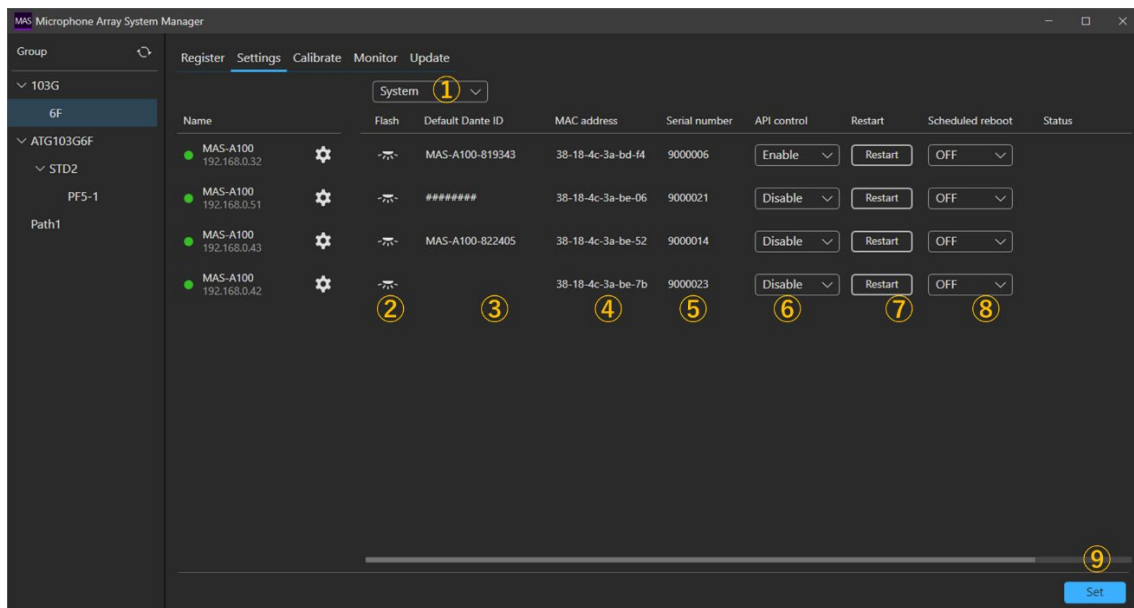
The MAS-A100 restarts. While restarting, the MAS-A100 is offline and the status lamp is gray.

8. Set a time for scheduled restart.

If the time is set in the range 0:00 to 23:00, the MAS-A100 automatically restarts at the specified time every day, based on the internal clock of the MAS-A100.

9. Press the [Set] button.

The changes for external control and scheduled restart are saved.

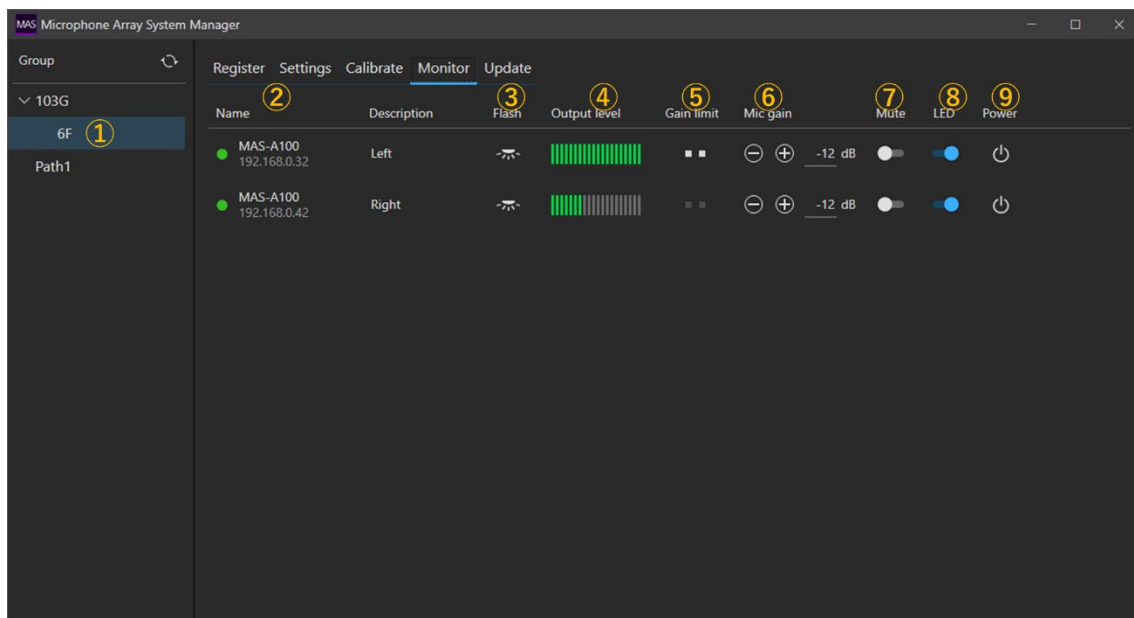


6. Operation and Maintenance

6.1. Monitor

You can browse the MAS-A100 status and temporarily change the audio settings.

1. Open the [Monitor] tab of MASM-1, and select any group in the left pane.
2. Check the status lamp, name, and IP address.
MAS-A100 units that display a gray lamp are offline and cannot be monitored.
3. Press the [Flash] icon to start the LED lamp of the MAS-A100 flashing to confirm its location, if necessary.
4. Check the output level.
5. Check the gain limit.
6. Check/adjust the microphone gain.
7. Switch the muting state.
8. Turn the LED lamp on/off.
9. Switch the MAS-A100 between Active state and Standby state.



Status lamps

The status lamp on the left side of the [Name] column in MASM-1 indicates the status of the MAS-A100.

The color of the lamp is the same as the color of the LED lamp on the MAS-A100 (provided the color setting of the LED is not changed from the factory default).

Green (Active): Audio for speech reinforcement is output from the Main output and Rec output of the MAS-A100.

Amber (Mute): Main output of the MAS-A100 is temporarily silent. The Rec output audio is output.

Red (Standby): Main output and Rec output of the microphone are silent, and MAS-A100 temporary settings are reset to defaults. See "5.7 Audio settings" (page 68).

Gray (offline): The MAS-A100 is not accessible from MASM-1. For details about resolving issues, see "5.1 MASM-1 preparation" (page 47).

Temporary settings

Temporary changes in the microphone gain, muting on/off, and LED lamp on/off on the [Monitor] tab view are reset to the defaults when the MAS-A100 switches from Active state to Standby state. See "5.7 Audio settings" (page 68).

6.2. Firmware update

You can update the firmware of MAS-A100 units on the [Update] tab of MASM-1.

Up to 10 MAS-A100 units can be updated at the same time. However, the maximum number may not be achievable if the network transfer speed between the PC and MAS-A100 units is low (for example, if local network is not Gigabit speed).

After updating, the LED lamp of the MAS-A100 flashes red and the unit may be reset to factory default state. For details about initial setup, see "5.2 MAS-A100 initial setup using MASM-1" (page 49).

Download

When new firmware is released, download updater files from the following website.

<https://pro.sony/mas-a100>

Update

1. Open the [Update] tab of MASM-1, and select any group in the left pane.
2. Place a check mark beside the MAS-A100 units to update.

Note

Up to ten units can be selected at any one time. When ten units become selected, the other checkboxes are grayed out.

3. Check the name, IP address, and status lamp.

MAS-A100 units that display a gray lamp are offline and cannot be updated.

4. Check the current firmware version of the MAS-A100.

5. Check whether the MAS-A100 firmware can be updated.

MAS-A100 units that display [OK] can be updated.

Only updating to a newer version is supported.

6. Press the [Choose file] button.

The Windows OS Explorer appears. Select a firmware file.

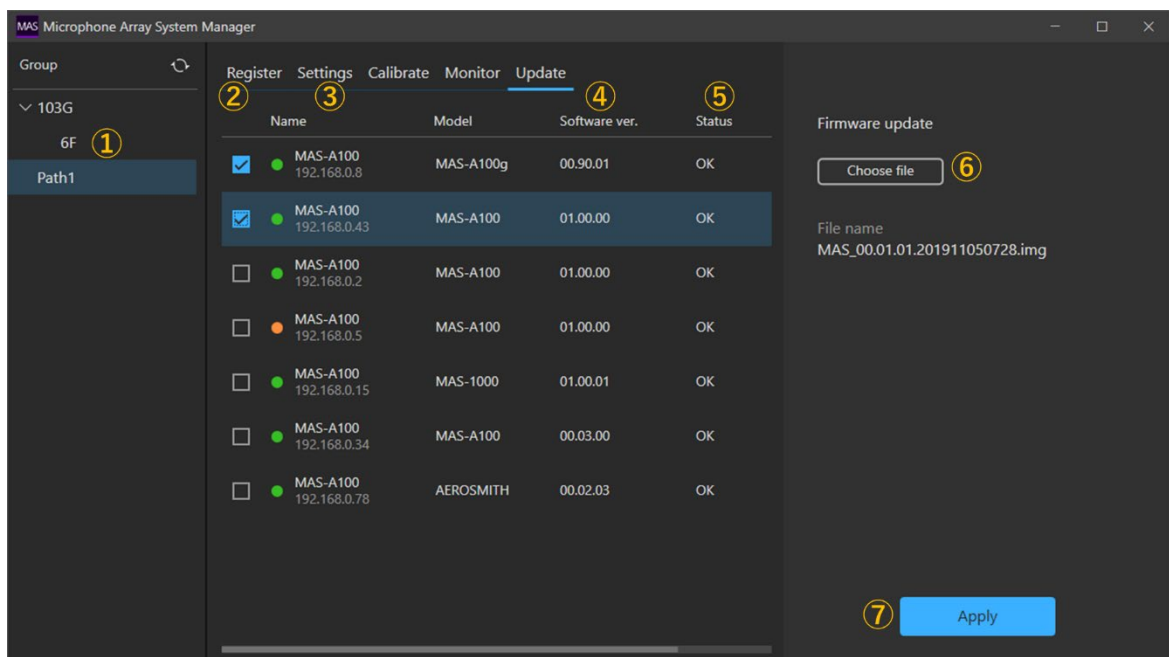
7. Press the [Apply] button.

The update is executed.

The update file is transferred over the network and installed in the MAS-A100.

Then, the MAS-A100 restarts, and the update is completed.

The update takes around four minutes, depending on the network bandwidth between the PC running MASM-1 and the MAS-A100.



6.3. Factory reset

Resets the settings of the unit to factory default state. Network settings, audio settings, and calibration state are all initialized.

In factory default state, the LED lamp of the MAS-A100 flashes green. For details about initial setup, see "5.2 MAS-A100 initial setup using MASM-1" (page 49).

If the installation location is changed or if the intended use is changed, such as from speech reinforcement to audio capture, perform a factory reset, initial setup, and calibration.

The reset operation is performed only to the network for setup. For details about resetting the Dante network interface to factory defaults, refer to the Dante Controller manual.

Web UI

Open the web UI of an MAS-A100 in a web browser, log in as the administrator, and reset the unit to factory default state. For details, refer to the MAS-A100 Operating Instructions.

Reset button

Locate the pinhole on the back of the MAS-A100, insert a pin, and depress the reset button at the base of the hole for 10 seconds while the MAS-A100 is operating. For details, refer to the MAS-A100 Operating Instructions.

6.4. MASM-1 log files

The log files of communications with the MAS-A100 units are stored in the C:\Users\[username]\AppData\Local\Sony\MasManager\Log folder.

The discovery and IP address setup (see "MAS-A100 discovery and IP address setting" (page 50)) communications logs are stored in SCAN.log, and all other communications logs are stored in CGI.log. The maximum size of log files is approximately 10 MB. When log files reach the maximum size, they are backed up with a number prefix added to the file name extension. Up to 10 files of each type of backup file can be created and used in rotation in the order SCAN.log.1, SCAN.log.2, ... SCAN.log.10 and CGI.log.1, CGI.log.2, ... CGI.log.10.

6.5. MASM-1 settings initialization

MASM-1 settings are stored in the following configuration file.

C:\Users\[username]\AppData\Local\Sony\MasManager\Config\MasManager.config

When the files are deleted, all of the following settings in MASM-1 are initialized.

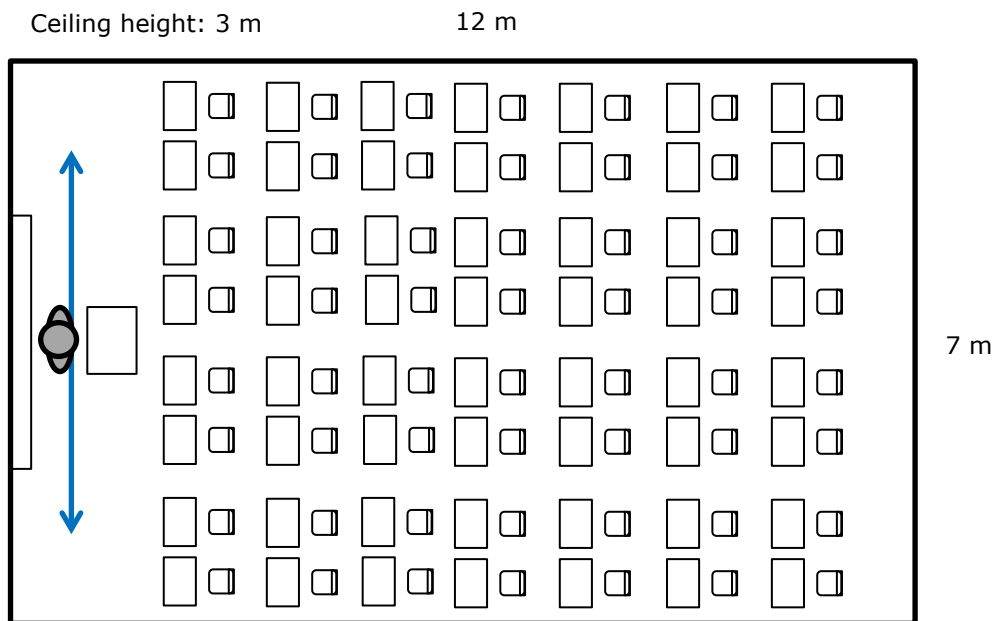
- MASM-1 login password information
- MAS-A100 registration information, Group

Note that configuration files are not deleted by uninstalling MASM-1.

7. Design, Installation, and Operation Example

This section describes the design, installation, and operation procedures for a specific installation.

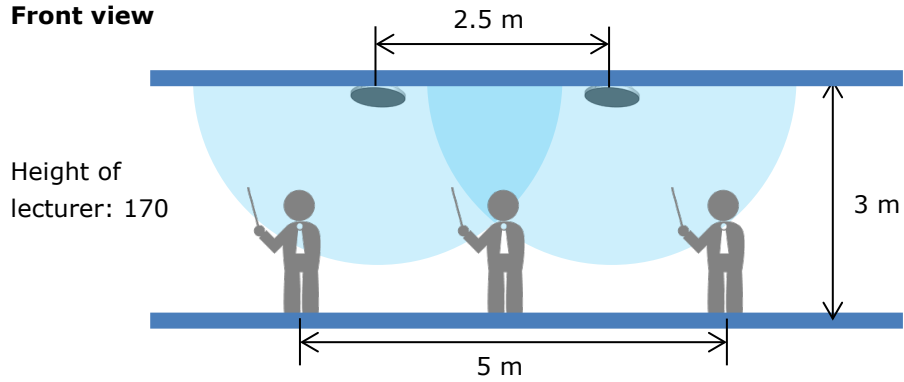
The target installation assumes speech reinforcement for lectures in the following lecture room, audio capture of lecture content, and remote lectures.



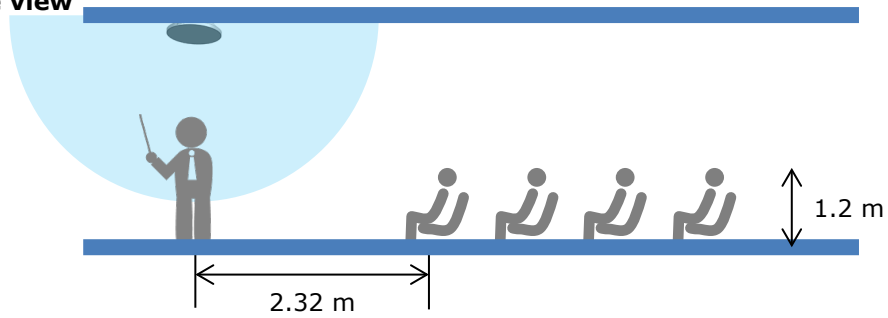
7.1. Installation design

Since this design requires the speech of the speaker to be reinforced, design the system as described in "Speech reinforcement system design" (page 34). Assuming a movement range of the speaker of 5 m, calculation using the online calculator gives the following results (see "Calculator for speech reinforcement" (page 21)).

Front view

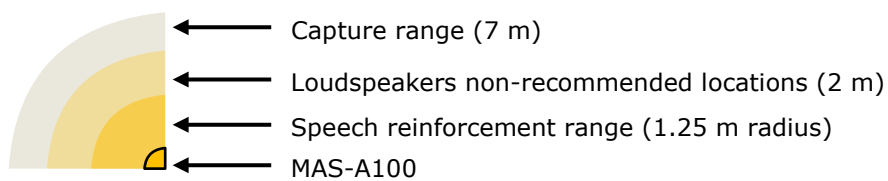
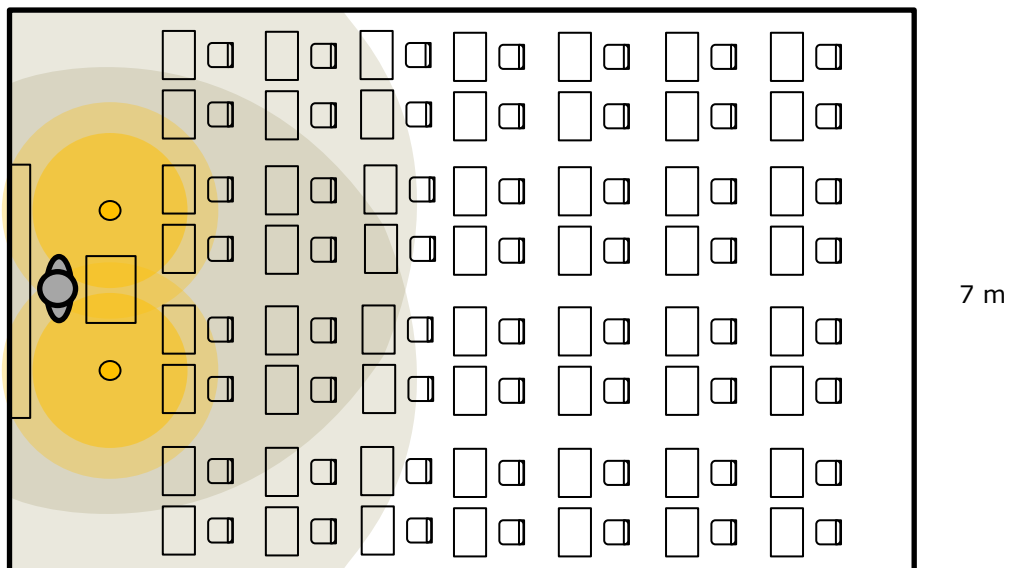


Side view



Within this distance, the sound of the audience may be captured.

12 m

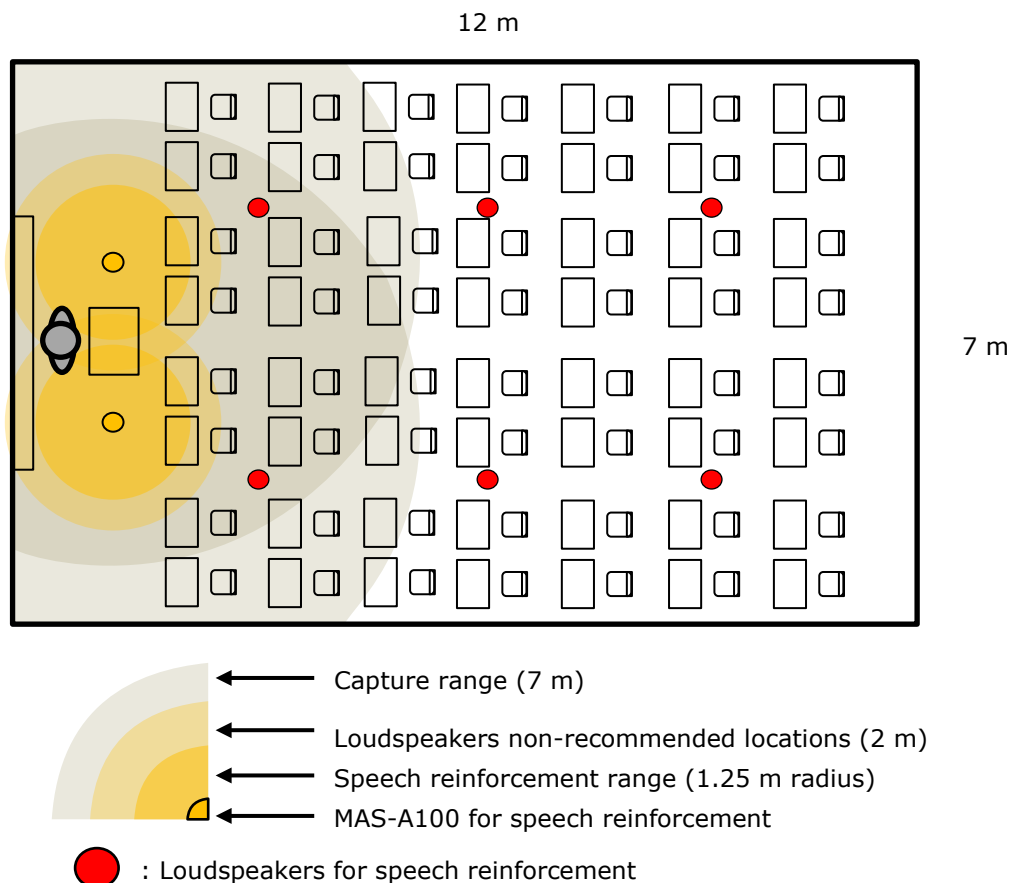


This result shows that sound from the audience in the rear seats will be difficult to capture. If necessary, consider installing additional MAS-A100 units to capture the sound from the audience.

Also, the results show the prohibited distance from the MAS-A100 within which loudspeakers should not be installed.

When determining the position of speech reinforcement loudspeakers, avoid installation within the prohibited area and install loudspeakers so that their directivity is not facing the MAS-A100. If there is an existing loudspeaker within the installation prohibited range, change the wiring so that the speech reinforced sound from the MAS-A100 is not emitted from the loudspeaker.

Based on the above results, 85 dB SPL speech reinforcement can be achieved throughout the lecture room by installing six loudspeakers (87 dB SPL/W/m 120 degrees) and avoiding the prohibited area.



7.2. Network design

Based on the results of the installation design, two MAS-A100 units are used with two outputs required, one for speech reinforcement and one for remote conferencing. Design a network with the devices and settings required for this configuration.

Network device list

No.	Name	Manufacturer	Remarks
1	MAS-A100	Sony	
2	MAS-A100	Sony	
3	PoE HUB GS108PEv3	NETGEAR	
4	Dante DAC ADP-DAO-AU-0X2	Audinate	
5	Dante DAC ADP-DAO-AU-0X2	Audinate	
6	Windows PC	Various companies	Windows 10

IP address assignment design

The network should be a closed network that does not connect to the Internet and which is not connected to other devices. IP addresses should be assigned from the 192.168.0.X private IP address range with a net mask of 255.255.255.0.

No.	Name	IP address	Remarks
1	MAS-A100	192.168.0.100	Dante
		192.168.0.110	Web UI/MASM-1
2	MAS-A100	192.168.0.101	Dante
		192.168.0.111	Web UI/MASM-1
3	PoE HUB GS108PEv3	192.168.0.239	
4	Dante DAC ADP-DAO-AU-0X2	192.168.0.102	
5	Dante DAC ADP-DAO-AU-0X2	192.168.0.103	
6	Windows PC	192.168.0.210	Windows 10

Note that the MAS-A100 requires two IP addresses.

Also, an IP address is required for the PoE hub.

PoE design

The GS108PEv3 has four PoE ports that can supply a total of 53 W.

No.	Name	Maximum power	Remarks
1	MAS-A100	13 W	GS108PEv3 Port 1
2	MAS-A100	13 W	GS108PEv3 Port 2
3	PoE HUB GS108PEv3		Supply source
4	Dante DAC ADP-DAO-AU-0X2	3 W	GS108PEv3 Port 3
5	Dante DAC ADP-DAO-AU-0X2	3 W	GS108PEv3 Port 4
6	Windows PC		GS108PEv3 Port 5

Wiring design

Design the length of the LAN cables (Cat 5e or higher) according to the installation location. The maximum length of Cat 5e cable is 100 m. Design the PoE hub installation location so that the distance is less than this length.

7.3. Audio design

Based on the installation design, the outputs of two MAS-A100 units will be connected to a speech reinforcement amplifier and a remote communication device.

Speech reinforcement system

A Dante DAC ADP-DAO-AU-0X2 (No. 4 in "Network device list" (page 86)) is used as the speech reinforcement system output.

Two analog outputs are connected to the line inputs of a speech reinforcement mixer (SRP-X500 in this example) using an adaptor cable. The audio input from the two MAS-A100 units can be connected to the L/R channels of an SRP-X500 mixer operating in SYSTEM TYPE 4 or SYSTEM TYPE 5 configuration, with output from the loudspeakers as monaural audio. If using a third-party amplifier that does not have a mixdown function from stereo to mono, use the following third-party adaptor cable.

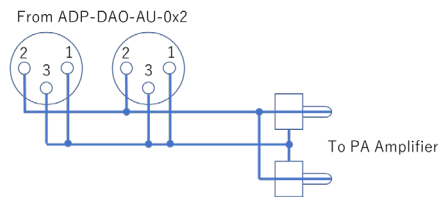
Speech reinforcement mixer
(e.g. SRP-X500)



Adaptor cable



Third-party adaptor cable



The SRP-X500 has a compressor inserted in the loudspeaker output system. Since this is a non-linear process, set the function to OFF or operate at a volume at which the compressor has no effect.

A feedback canceler and AGC are inserted in the microphone input of the SRP-X500. This is a non steady-state process, so connecting the MAS-A100 to the microphone input is not recommended.

When using the MAS-A100, do not adjust the master volume using the front panel of the SRP-X500.

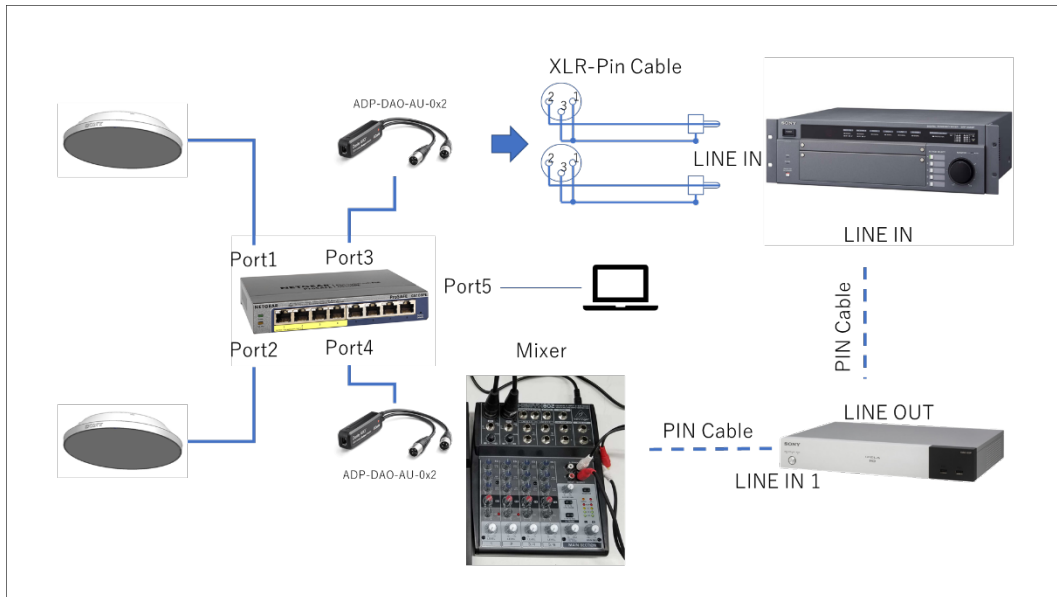
Remote lecture system

A Dante DAC ADP-DAO-AU-0X2 (No. 5 in "Network device list" (page 86)) is used as the remote lecture system output.

Mix the two analog outputs and adjust the volume and sound quality using a portable mixer. Connect the output of the mixer to the LINE 1 input of a video conferencing system (PCS-XG100 in this example). On the [Basic Setup1] page of the [Audio] setup menu of the PCS-XG100, set [Audio Input] to [MIC] and set [Input Select] to [AUDIO 1].

Note

If [Audio Input] is set to [AUX], the echo canceler is not applied.



For the volume of the MAS-A100, check the input volume meter of the video conferencing system and adjust the volume of the mixer to the appropriate volume. If the volume is too loud, abnormal noise will occur. If it is too low, the sound will cut in and out.

Connect the audio output of PCS-XG100 to the AV/RGB INPUT audio input E connector of the SRP-X500 to connect it to the speech reinforcement loudspeaker. Then select input E for the remote lecture.

In this design example, the audio from other microphones (handheld microphones, wireless microphones) connected to the speech reinforcement audio system are not sent directly to the remote location. The sound from the speech reinforcement loudspeakers may be captured by the MAS-A100 and then sent with poor volume and sound quality. To also capture other handheld microphones and wireless microphones with high audio quality, a different design approach is recommended.

Device list

No.	Name	Manufacturer	Remarks
1	MAS-A100	Sony	
2	MAS-A100	Sony	
3	PoE HUB GS108PEv3	NETGEAR	
4	Dante DAC ADP-DAO-AU-0X2	Audinate	
5	Dante DAC ADP-DAO-AU-0X2	Audinate	
6	Windows PC	Various companies	Windows 10

No.	Name	Manufacturer	Remarks
7	XLR-to-pin cable stereo or adaptor cable	Various companies	
8	Audio mixer (2ch or more) XLR in, Pin out	Various companies	
9	Pin cable	Various companies	

7.4. Network settings

Before connecting the system, set and check the IP address of each device in the following order.

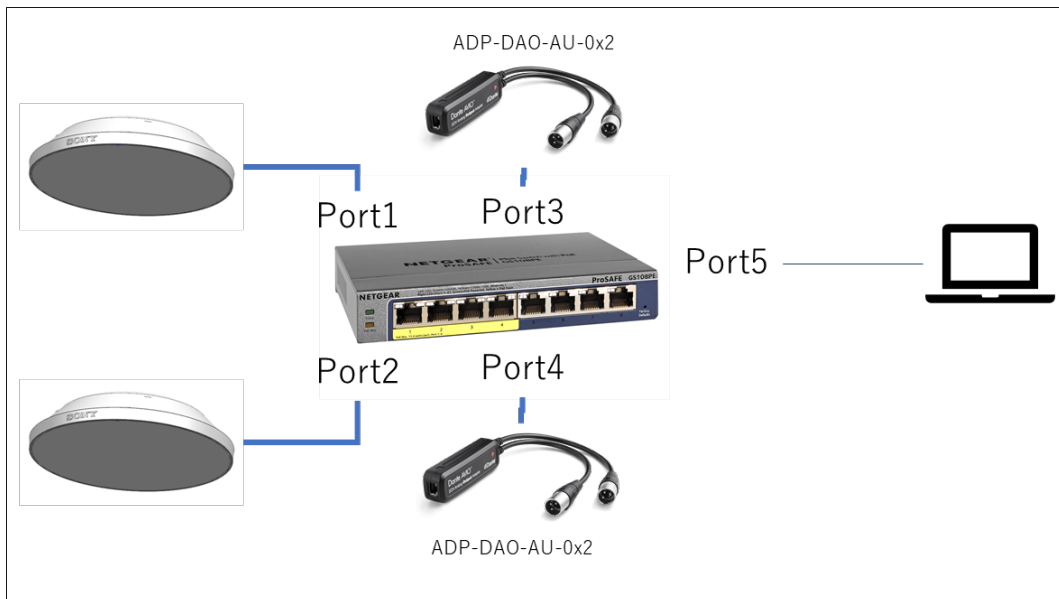
Windows PC

Download and install the following applications.

No.	Name	Remarks
1	MASM-1	See "5.2 MAS-A100 initial setup using MASM-1" (page 49)
2	Dante Controller	See "5.3 Dante Controller preparation" (page 57)
3	ProSAFE Plus Utility	See "NETGEAR GS108PEv3 setup" (page 28)
4	Web browser	Google Chrome

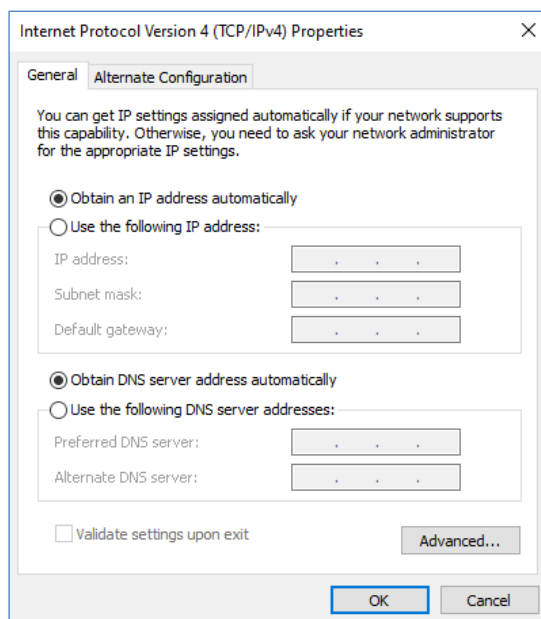
Device connection

Referring to the following diagram, connect the Windows PC, MAS-A100, and DAC ADP-DAO-AU-0X2 to a PoE hub, then turn on each device.



Dante device address settings

1. Configure the PC to obtain an IP address automatically.



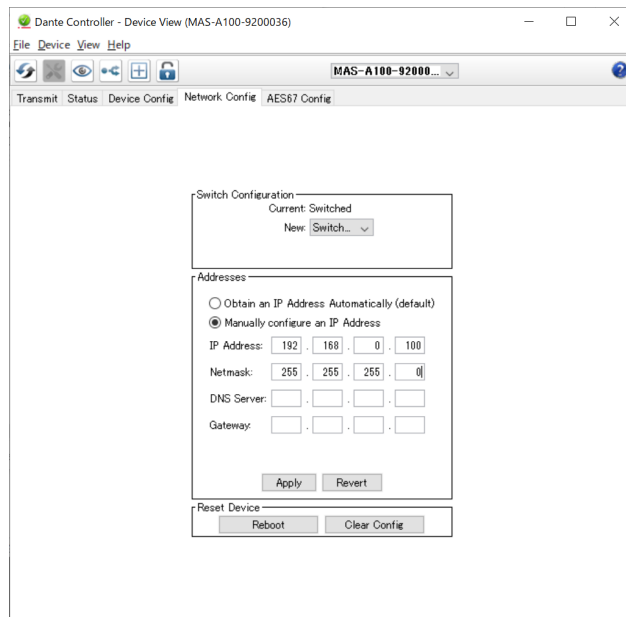
The PC operates with an address 169.254.x.x (x is any number).

To prevent incorrect setup, connect each Dante device to the hub and configure the device one at a time.

2. Launch Dante Controller.

The Dante devices connected to the hub are displayed on the [Device Info] tab.

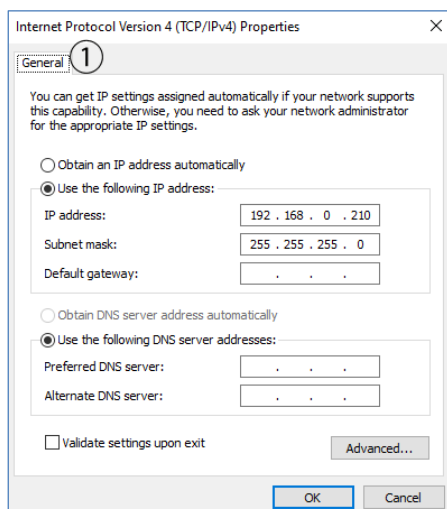
3. Double-click a device to display the [Device View] screen.
4. On the [Network Config] tab, set [IP Address] and [Netmask].
Enter the IP address from the design in “IP address assignment design” (page 86) in [IP Address], and set [Netmask] to “255.255.255.0”. [DNS Server] and [Gateway] settings are not required.



5. Press the [Reboot] button to reboot the device.

Hub (NETGEAR GS108PEv3)

1. Set the IP address of the PC to “192.168.0.210” and the subnet mask to “255.255.255.0”.



Refer to “NETGEAR GS108PEv3 setup” (page 28) and disable IGMP snooping.

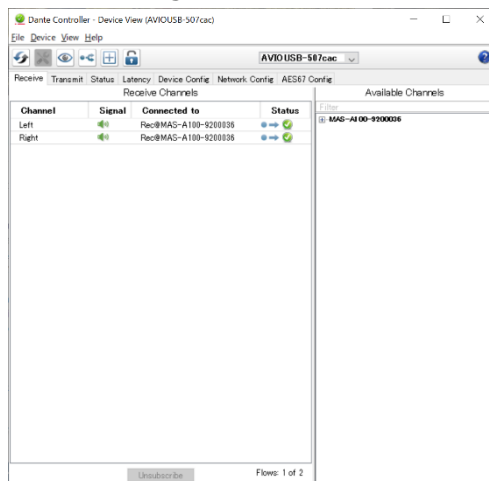
MAS-A100 initial setup

Refer to “5.2 MAS-A100 initial setup using MASM-1” (page 49) to configure the IP address according to the design, and then perform initial setup and registration.

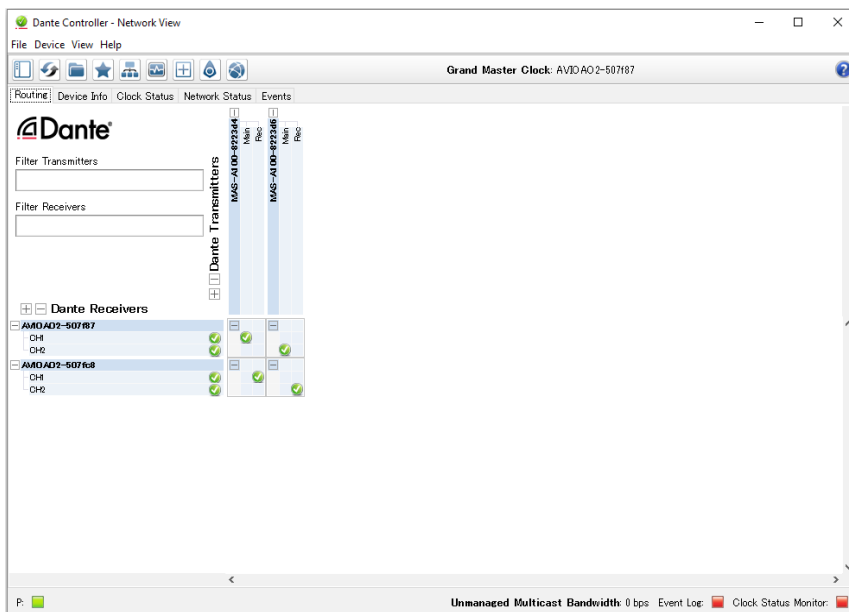
7.5. Audio settings

Dante routing

With all Dante devices connected, use Dante Controller to control Dante routing. To distinguish between the two ADP-DAO-AU-0X2 units, view the two LED lamps on the ADP-DAO-AU-0X2 Ethernet connector and press the [Identify] button on the [Dante View] tab of Dante Controller. The two LED lamps of the selected device will start flashing.



Route the Main outputs of the two MAS-A100 units to the ADP-DAO-AU-0X2 for Main, and the Rec outputs of the two MAS-A100 units to the ADP-DAO-AU-0X2 for Rec.



Connection check

If connected successfully, the sound captured by the MAS-A100 is now input to the speech reinforcement system and remote conference system.

Use the test signal generator function of the MASM-1, and check that the test signal is emitted from the loudspeakers with the appropriate volume.

Next, speak facing the MAS-A100 and check that there are no issues with the audio input using the input level meter of the remote conference device.

7.6. Installation

Install the MAS-A100 and other devices according to the design.

For details about installing, refer to the MAS-A100 Operating Instructions.

7.7. Calibration

Refer to "Calibration preparation" (page 63) and check that the test signal is emitted from the speech reinforcement loudspeakers with the appropriate volume. If required, adjust the volume of the mixer or amplifier.

Perform calibration as described in "Calibrating" (page 64).

For important points after calibration, see "Speech reinforcement system design" (page 34).

7.8. Operation

After calibration, you can adjust the volume and sound quality using MASM-1 (see "6.1 Monitor" (page 78)).

If you adjust the master volume of the amplifier or the input volume of the MAS-A100, perform the calibration again.

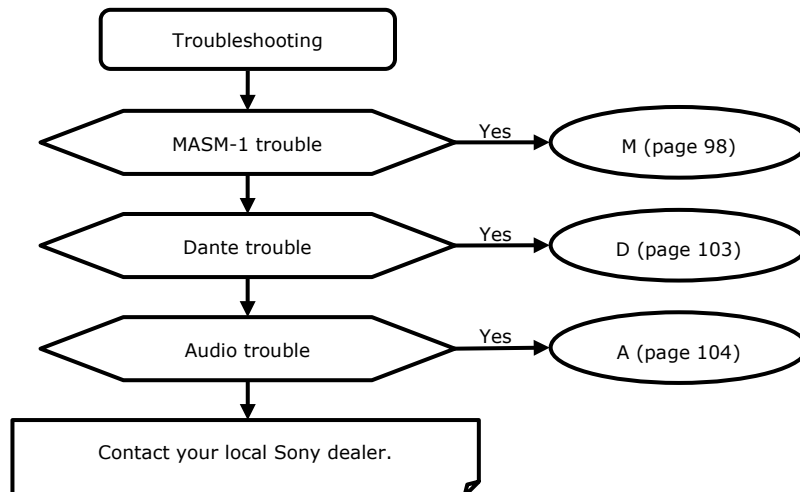
8. Usage Precautions

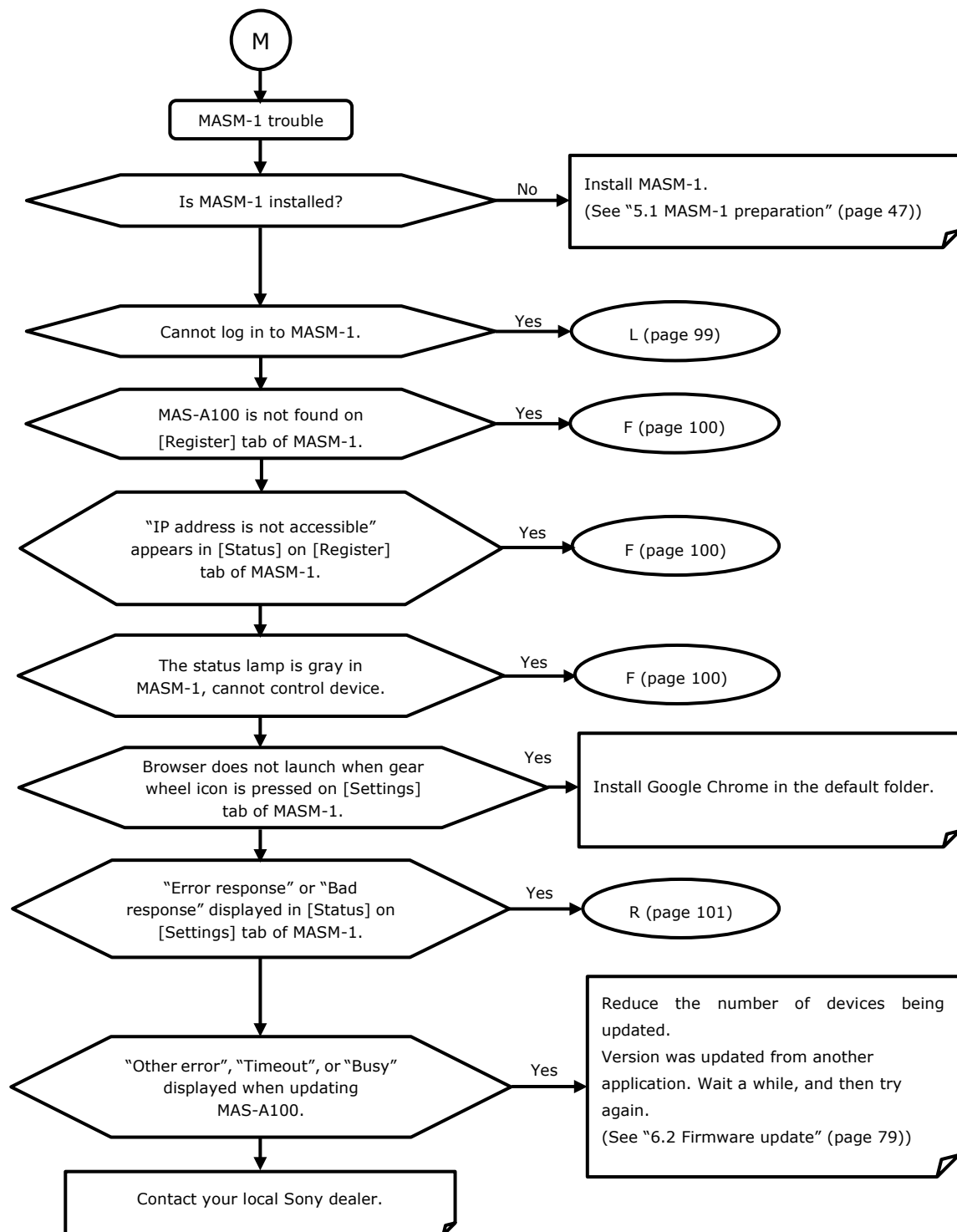
- MASM-1 does not have an SSL function. If the SSL function of the MAS-A100 is enabled, MASM-1 can only perform MAS-A100 discovery and IP address setting. Other operations are not available.
- In some MAS-A100 installation environments and audio systems, the level of speech reinforcement and audio quality may not be sufficient.
- Noise may not be reduced sufficiently in some MAS-A100 installation environments.

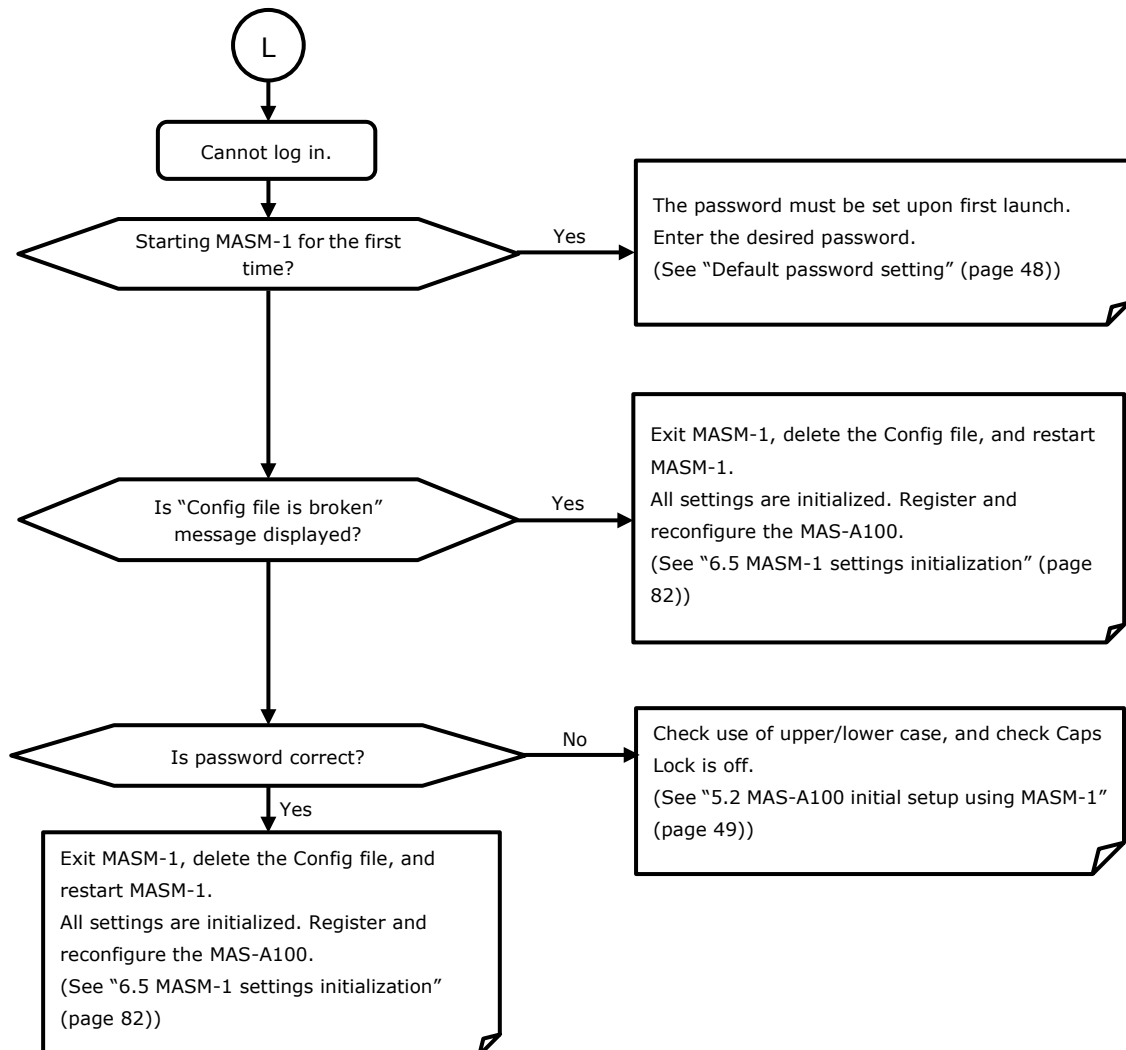
Network security

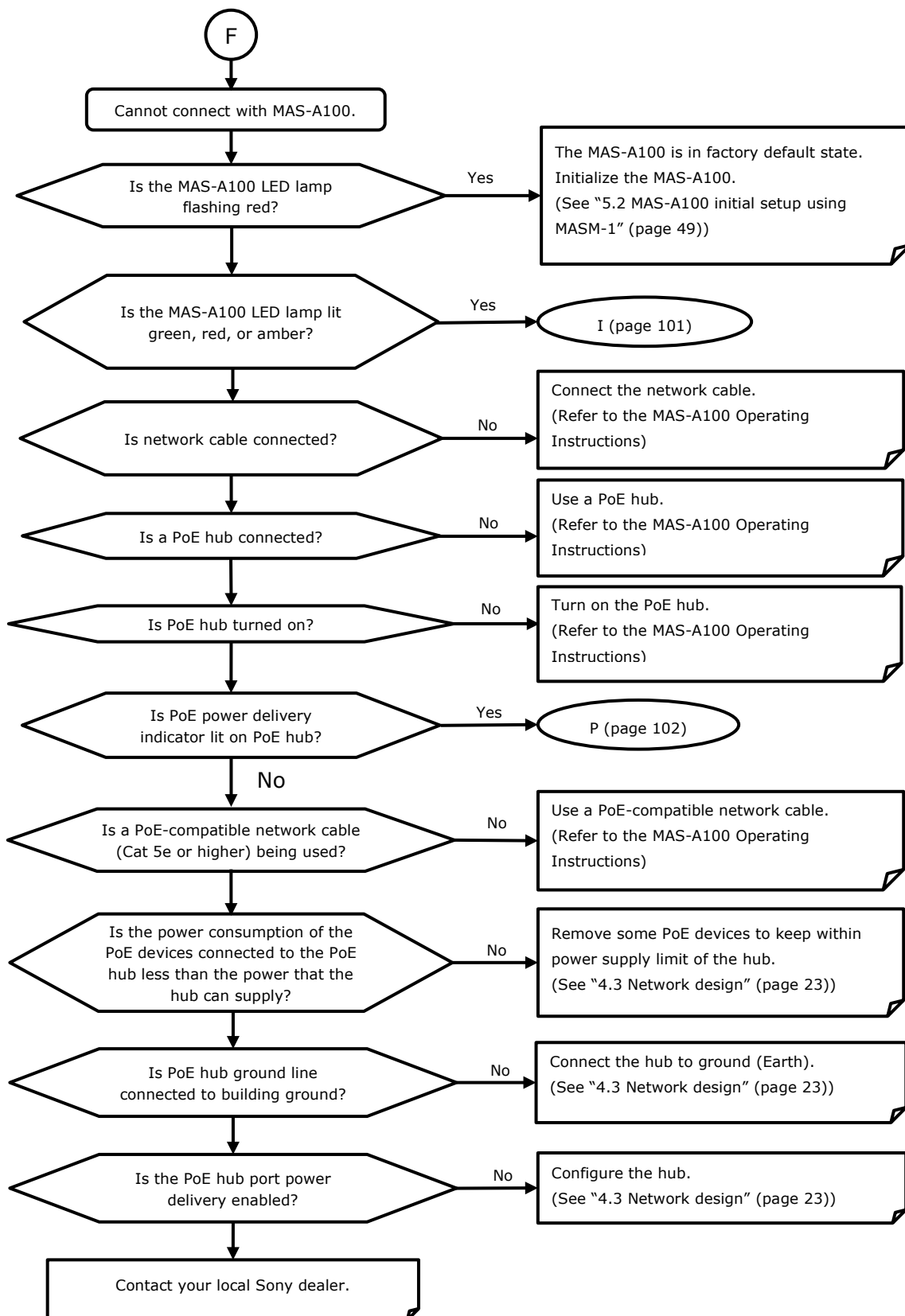
- SONY WILL NOT BE LIABLE FOR DAMAGES OF ANY KIND RESULTING FROM A FAILURE TO IMPLEMENT PROPER SECURITY MEASURES ON TRANSMISSION DEVICES, UNAVOIDABLE DATA LEAKS RESULTING FROM TRANSMISSION SPECIFICATIONS, OR SECURITY PROBLEMS OF ANY KIND.
- Depending on the operating environment, unauthorized third parties on the network may be able to access the unit. When connecting the unit to the network, be sure to confirm that the network is protected securely.
- MASM-1 does not have an SSL function. Communications between MASM-1 and MAS-A100 units is unencrypted. Use only on a secure network.

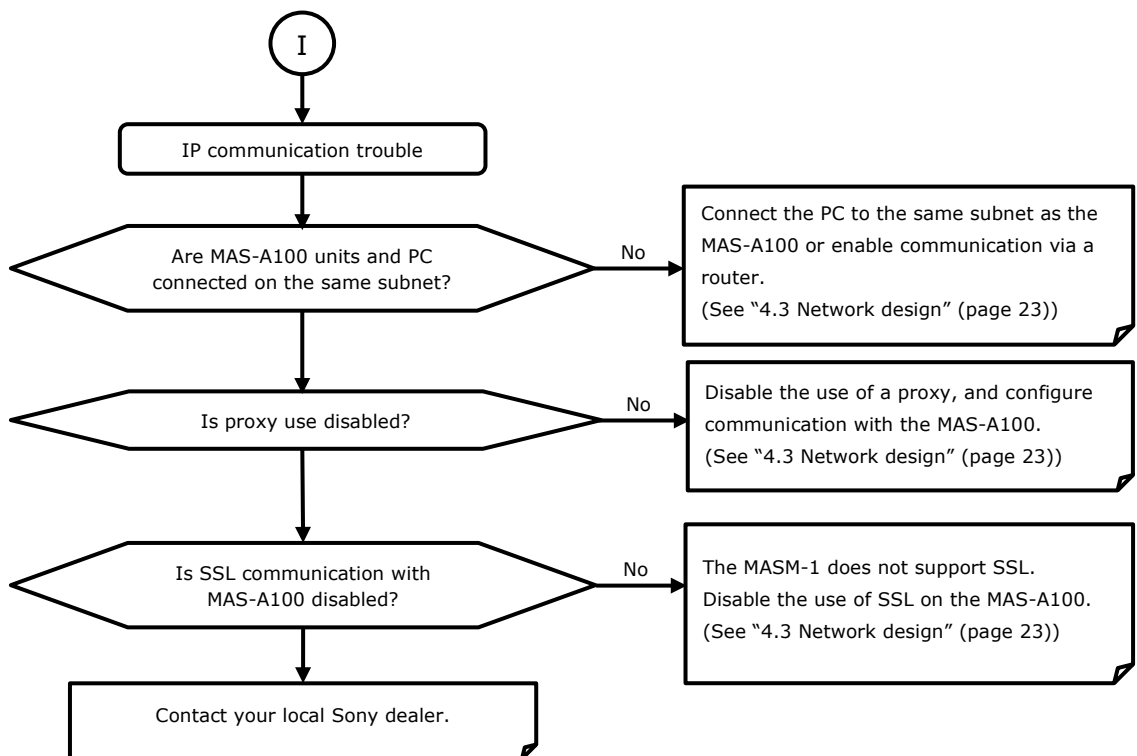
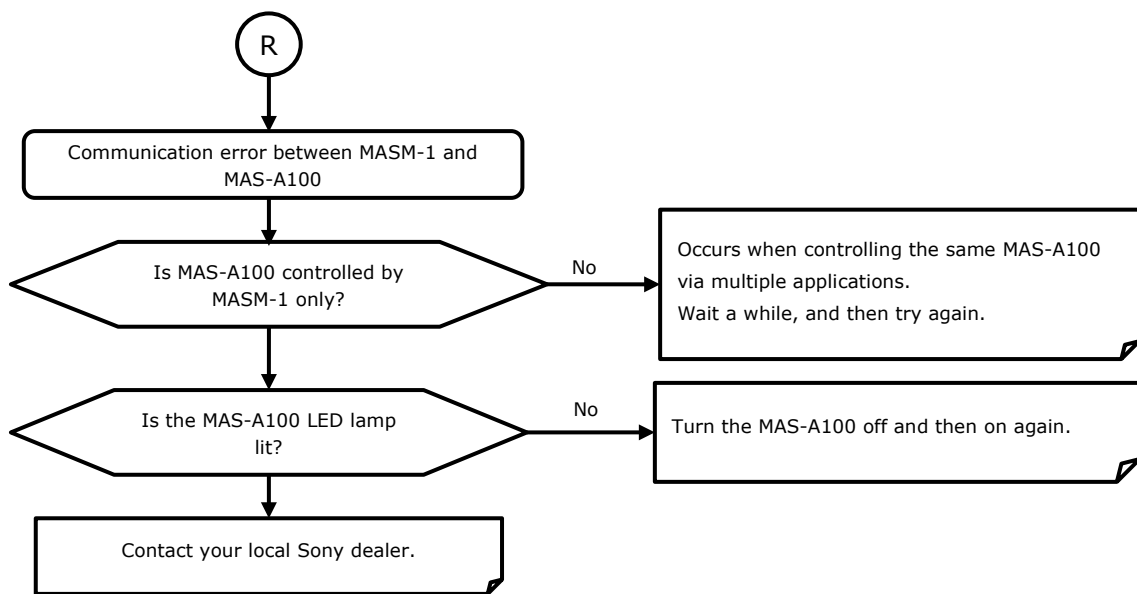
Appendix A. Troubleshooting

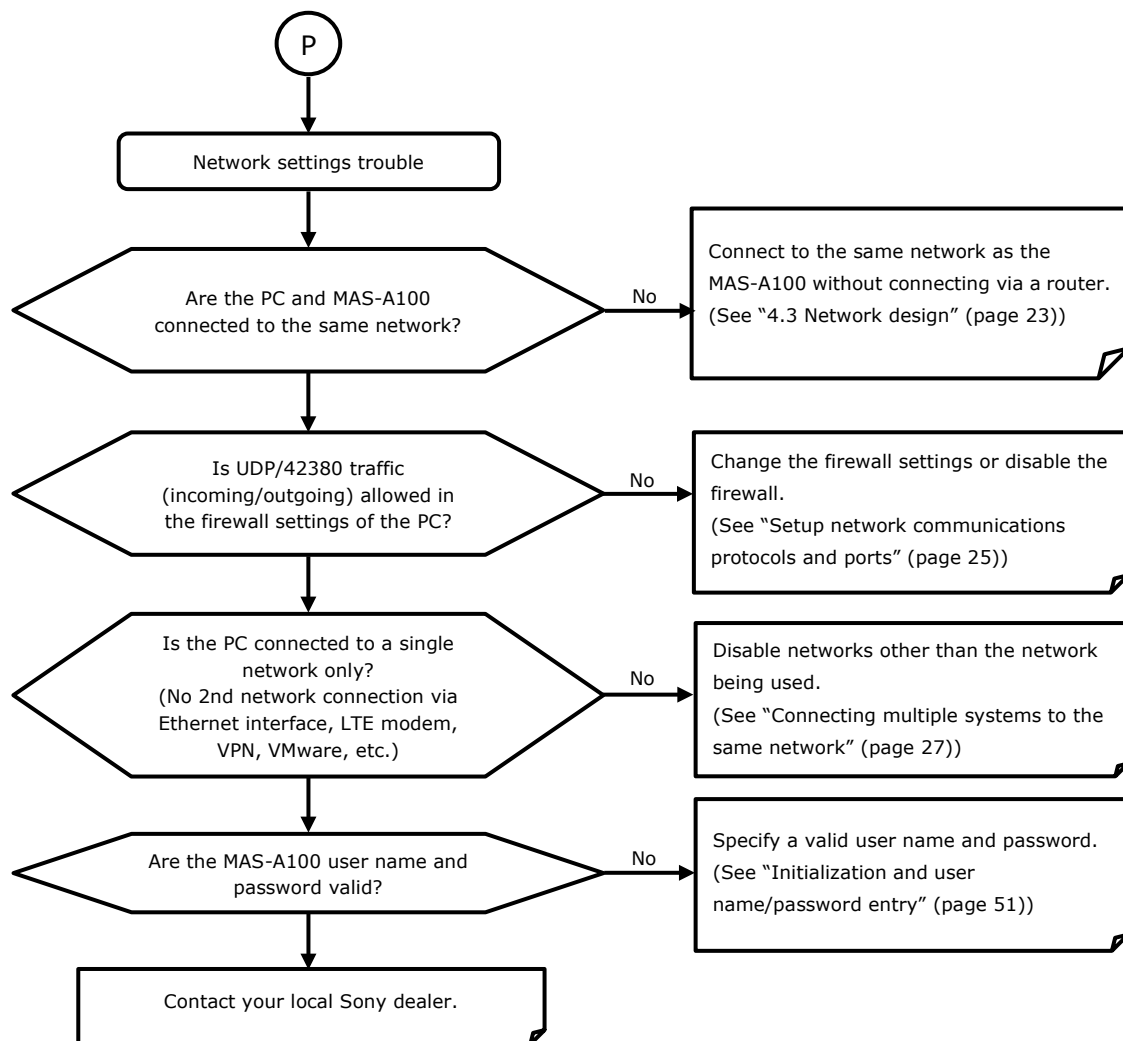


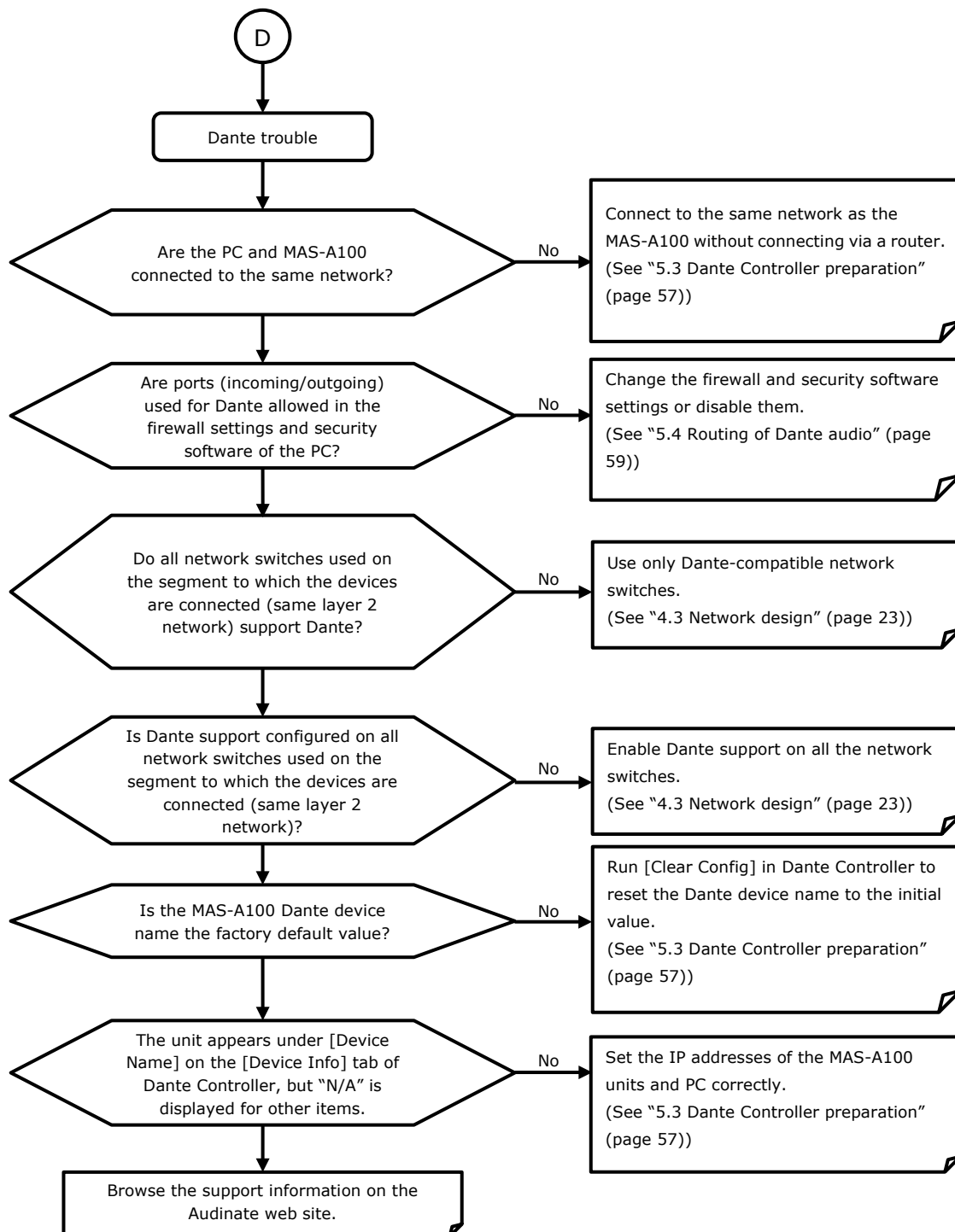


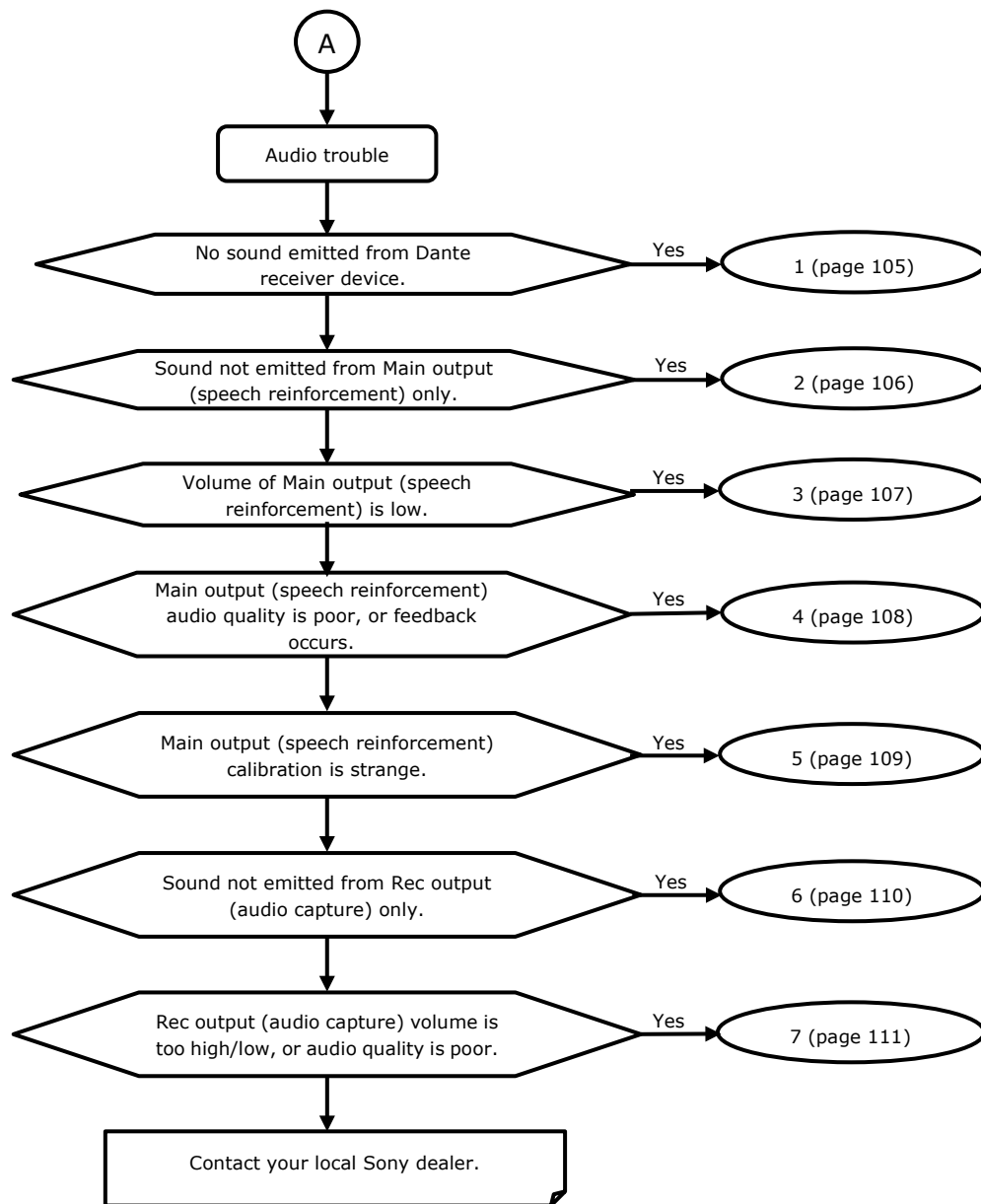


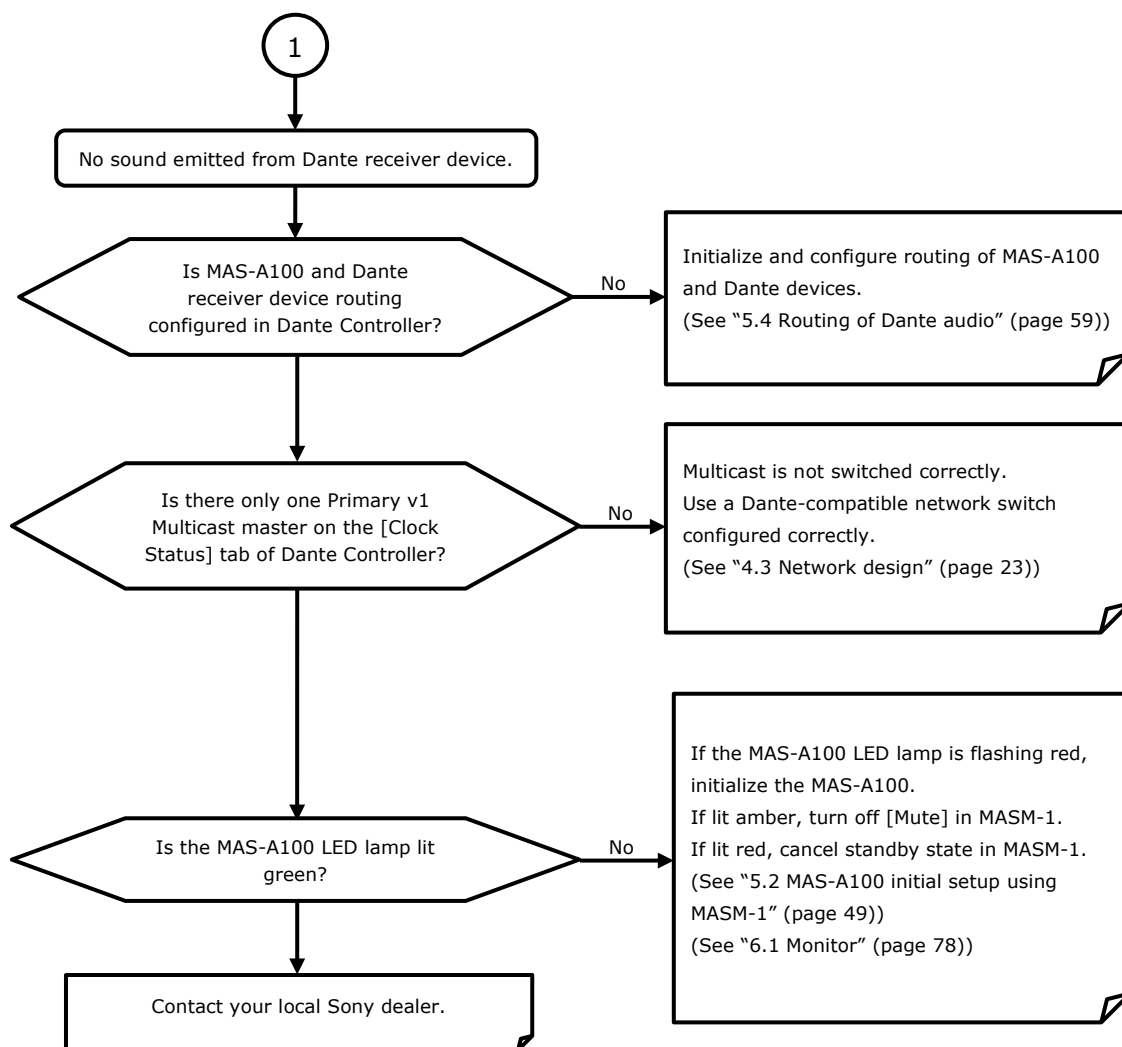


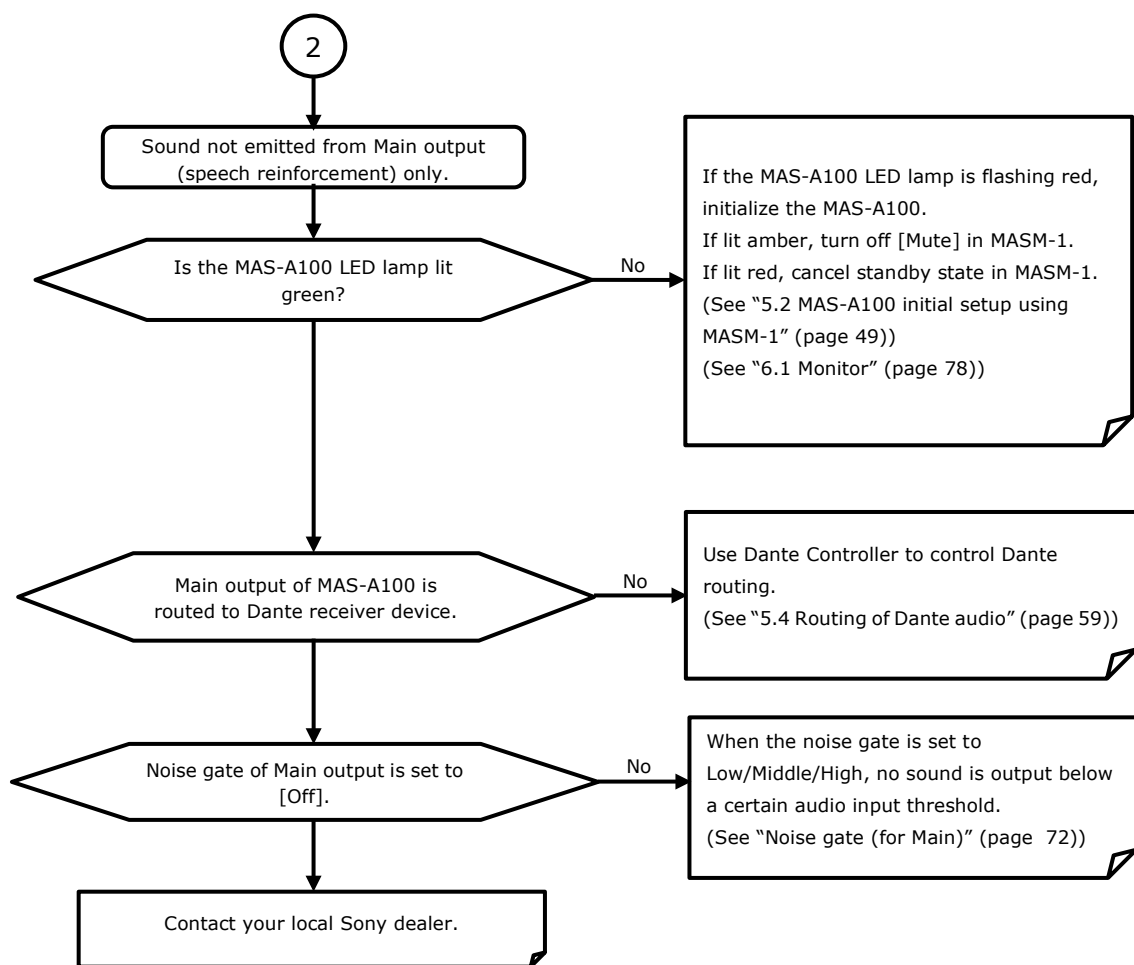


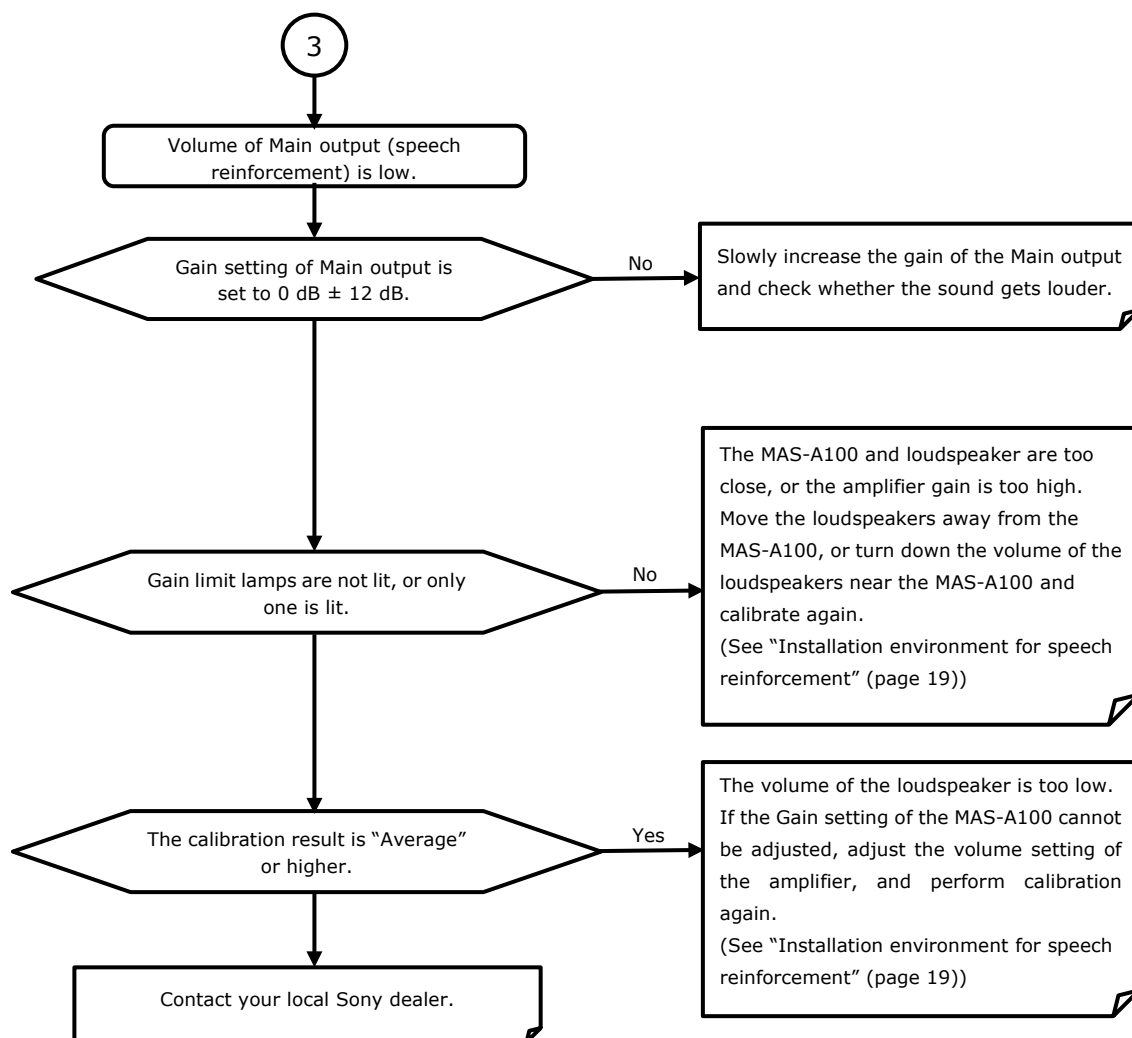


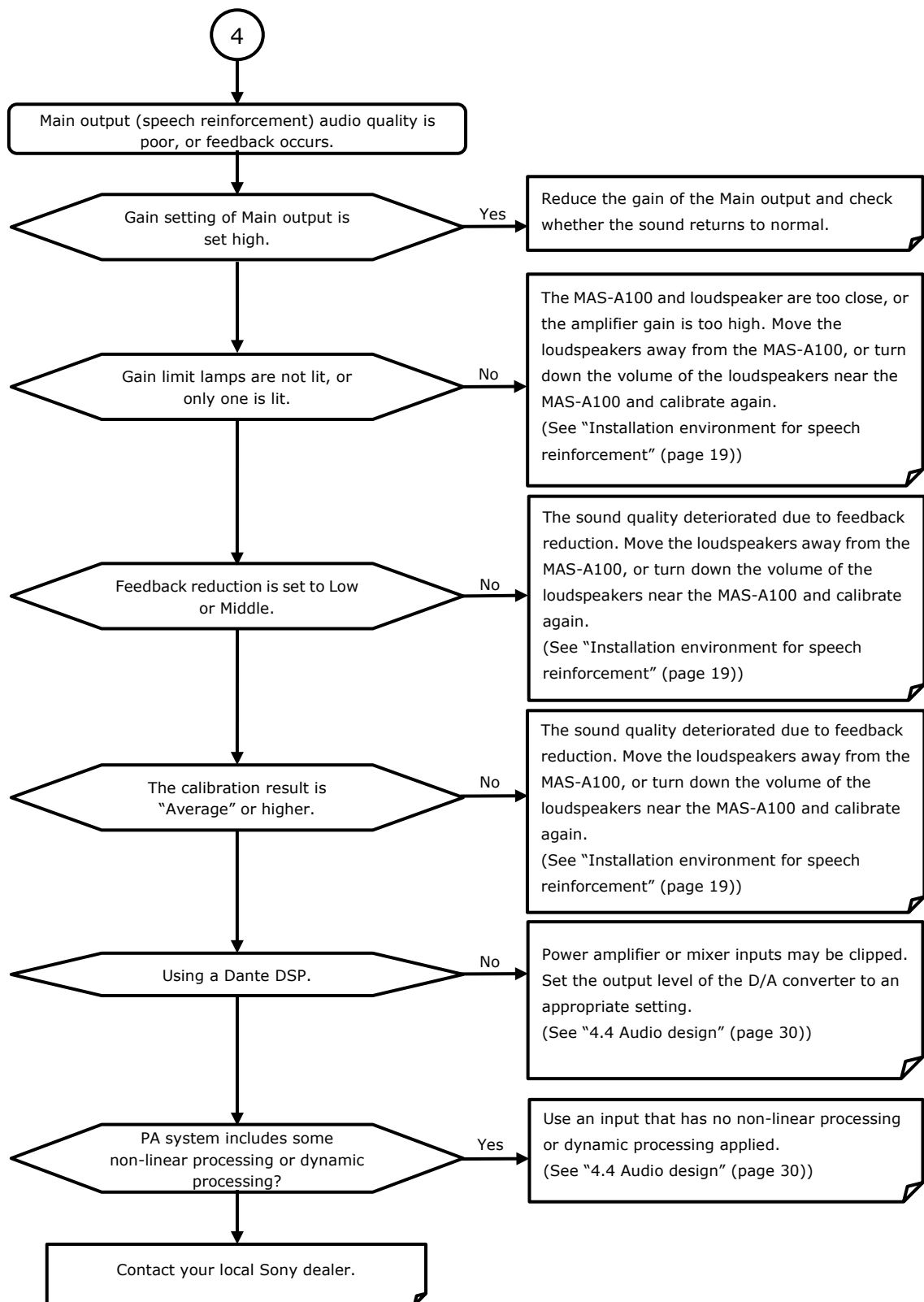


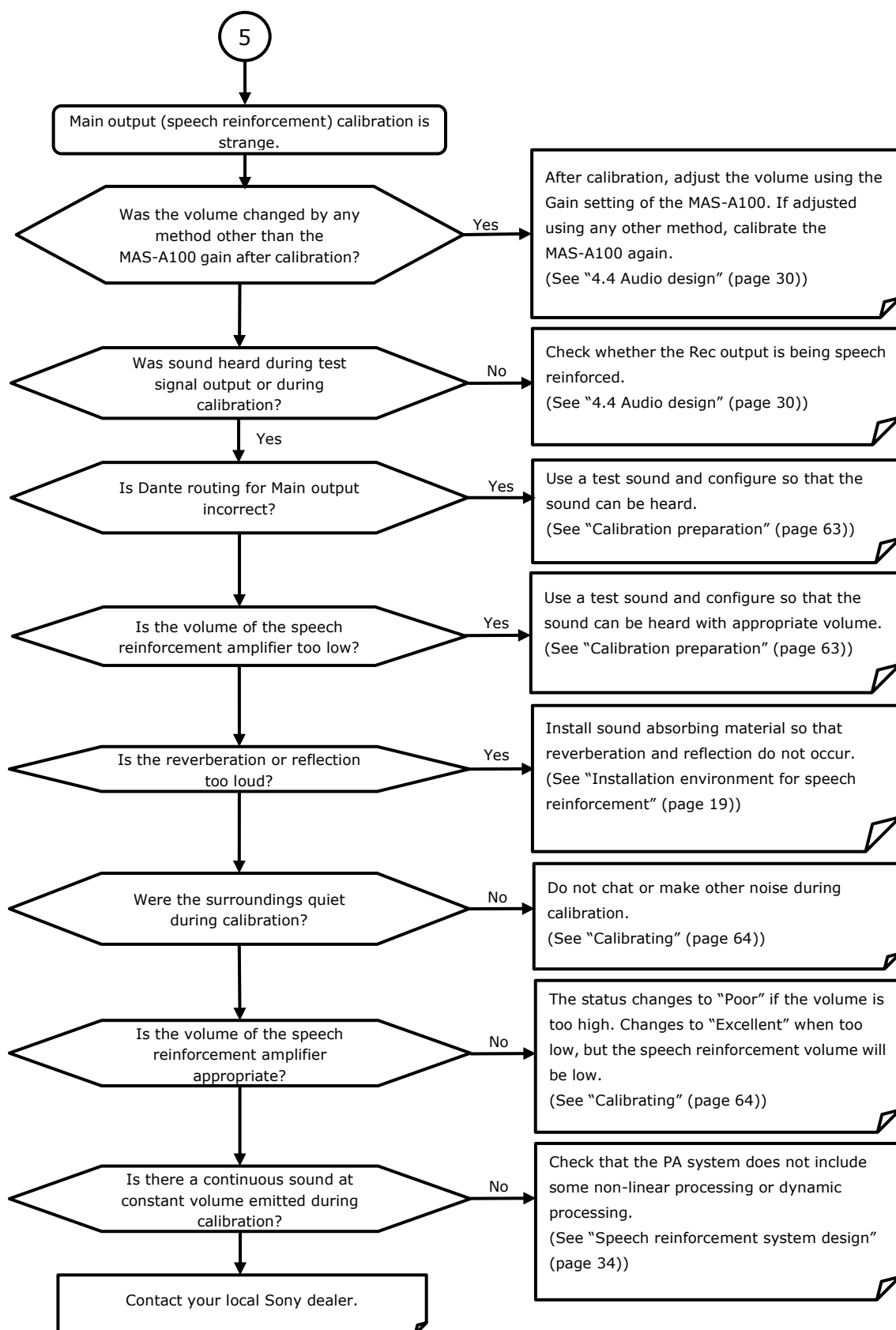


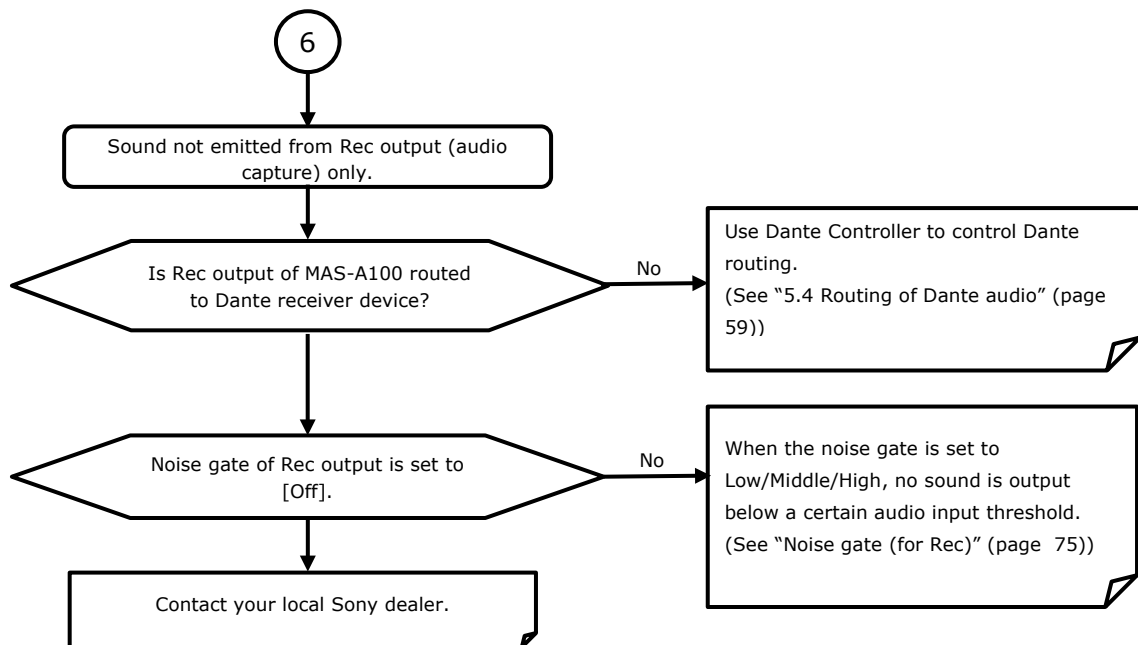


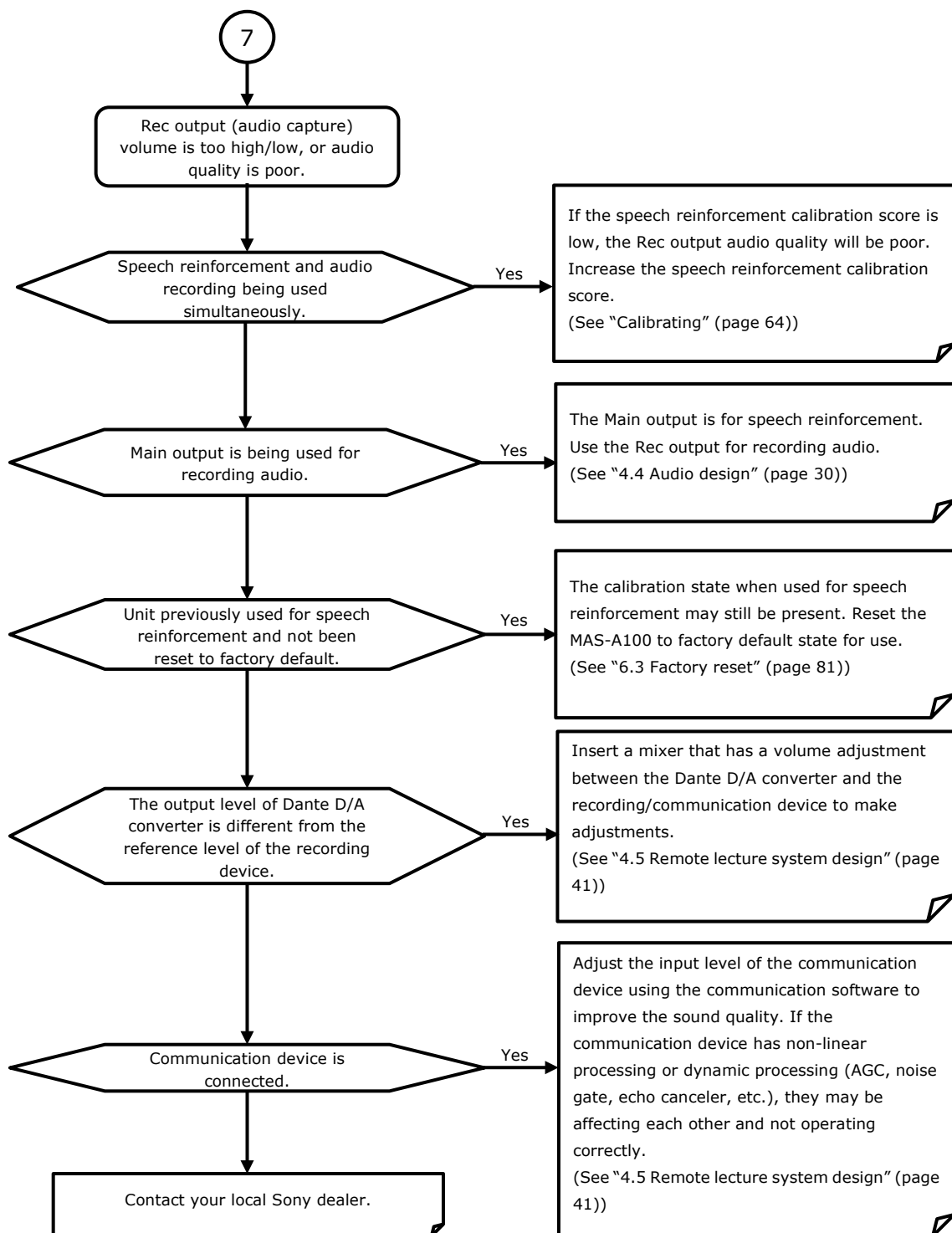












Appendix B. External Control

B.1. Setting external control

The MAS-A100 has an API that enables it to be controlled from an external system, such as AMX, Crestron, or Extron.

The API is disabled by factory default.

To enable the API, set [API control] to [Enable] on the [System] page of the [Settings] tab of MASM-1.

When the API is enabled, a network port of the MAS-A100 is opened, and the unit is accessible from external sources via that port, which is a potential security risk. Make sure that the API is enabled only when connected to a secure network.

B.2. Communication specifications

Internet protocol	IPv4
Transport protocol	TCP
IP address	IP address of the MAS-A100
Port	42382

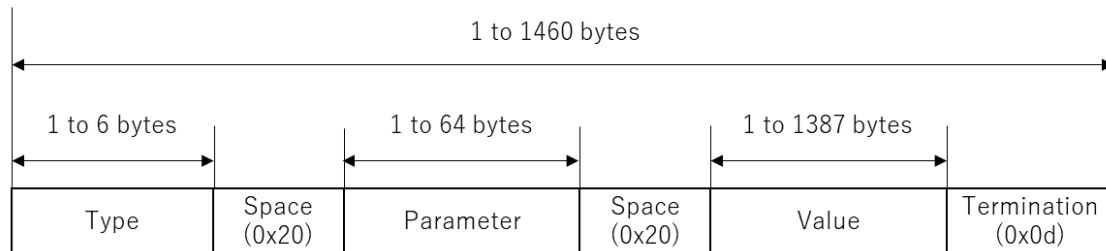
B.3. Format

Commands to an MAS-A100 and the response from an MAS-A100 consist of the following elements.

- Type
- Parameter
- Value
- Space-delimited text (SP) (0x20)
- Termination character (CR) (0x0d)

They are ASCII characters and case sensitive.

The format of the command and an example are shown below.



Hexadecimal code	53	45	54	20	4d	69	63	41	63	74	69	76	65	20	6f	6e	0d
Character	S	E	T		M	i	c	A	c	t	i	v	e		o	n	

B.4. Command

SET command

The SET command is used to configure the MAS-A100.

Type	Parameter	Value	Description
SET	MicActive	on	Set to Active state.
SET		off	Set to Standby state.
SET	Ch1MicMute	on	Set muting on.
SET		off	Set muting off.
SET	Ch1MicGain	-60 to 12	Set the microphone gain value.

GET command

The GET command is used to retrieve the status of the MAS-A100. Value is not specified.

Type	Parameter	Value	Description
GET	MicActive	None	Get the status (Active or Standby).
GET	Ch1MicMute	None	Get the status of muting (on or off).
GET	Ch1MicGain	None	Get the microphone gain value.

Response

The MAS-A100 returns a response indicating the result of the execution of the command.

The response is a combination of the result of the command, response type, parameter, and value as shown below.

Result	Type	Parameter	Value
SET success	RES_OK	Same as specified in the command	None
GET success	RES_OK	Same as specified in the command	Return value
Failure	RES_NG	Same as specified in the command	Error type

SET success

Command	Response
SET MicActive on	RES_OK MicActive
SET MicActive off	RES_OK MicActive
SET Ch1MicMute on	RES_OK Ch1MicMute
SET Ch1MicMute off	RES_OK Ch1MicMute
SET Ch1MicGain xxx	RES_OK Ch1MicGain

GET success

Command	Response	Description
GET MicActive	RES_OK MicActive on	Active state
	RES_OK MicActive off	Standby state
GET Ch1MicMute	RES_OK Ch1MicMute on	Muting on
	RES_OK Ch1MicMute off	Muting off
GET Ch1MicGain	RES_OK Ch1MicGain xxx	Microphone gain (where xxx is a value from -60 to 12)

Failure

The type of response is RES_NG if the command fails or cannot be executed. The value indicates the error symptom as shown below.

Value	Symptom	Solution
Timeout	Timeout	Try again later.
CommandNotExecutable	The command is not executable.	Restart the MAS-A100.
CommandNotSupported	The command is not supported.	Check that the type of the command is correct.
ParameterNotSupported	The parameter of the command is not supported.	Check that the parameter of the command is correct.
InvalidValue	The value of the command is invalid.	Check that the value of the command is correct.
BadCommandFormat	The format of the command is not correct.	Check that the format of the command is correct.
Failure	Other failure.	Restart the MAS-A100.

B.5. Crestron support

MAS-A100 units can be controlled from a Crestron controller using the external control API.

Download the module used for control from the following site.

https://applicationmarket.crestron.com/sony_mas-a100/

B.6. Sample program

The following code is an example of an intelligent module script for Symetrix Composer 8.0 that uses the API to control the microphone gain and mute settings of an MAS-A100 unit from a Symetrix device.

Example

```
-- For Symetrix Composer 8.0
-- MAS-A100 remote control lua script
-- This software is provided "as is".
--

MASComm = {}
function MASComm.new()
    local obj = {
        sock = TcpSocket.New(),
        MASCommTimer = Timer.New(),
        ipaddr = "",
        cur1 = -999,
        cur2 = -999,
        cur3 = -999,
        data = ""
    }
    return setmetatable(obj, {__index = MASComm})
end

function MASComm:ControlCheck(cur,new)
    return new >= 0.0 and new <= 1.0 and cur ~= new
end

function MASComm:send(data)
    self.data = data
    self.sock:Connect(self.ipaddr, 42382)
end
```

```

function MASComm:start()
    self.sock.ReadTimeout = 0
    self.sock.WriteTimeout = 0
    self.sock.ReconnectTimeout = 0
    self.sock.EventHandler = function(sock, evt, err)
        if evt == TcpSocket.Events.Connected then
            print("socket connected\r")
            self.sock:Write(data)
            print("Xmit:" .. data)
        elseif evt == TcpSocket.Events.Reconnect then
            print("socket reconnecting...\r")
        elseif evt == TcpSocket.Events.Data then
            rxLine = sock:Read(sock.BufferLength)
            if (nil ~= rxLine) then
                print("Got:" .. rxLine)
                sock:Disconnect()
                self.data=""
            end
        elseif evt == TcpSocket.Events.Closed then
            print("socket closed by remote\r")
        elseif evt == TcpSocket.Events.Error then
            print(string.format("Error: '%s'\r", err))
        elseif evt == TcpSocket.Events.Timeout then
            print("socket closed due to timeout\r")
        else
            print("unknown socket event\r")
        end
    end
end

self.MASCommTimer.EventHandler = function ()
    if self.data == "" and not self.sock.IsConnected then
        if self:ControlCheck(self.cur1,Controls.Inputs[1].Value) then
            self.cur1 = Controls.Inputs[1].Value
            data = string.format("SET Ch1MicGain %d\r",math.floor( self.cur1 * 72)-60)
            self:send(data)
        elseif self:ControlCheck(self.cur2,Controls.Inputs[2].Value) then

```

```

        self.cur2 = Controls.Inputs[2].Value
        if(self.cur2 > 0.5) then
            data = "SET Ch1MicMute on\x0d"
        else
            data = "SET Ch1MicMute off\x0d"
        end
        self:send(data)
    elseif self:ControlCheck(self.cur3,Controls.Inputs[3].Value) then
        self.cur3 = Controls.Inputs[3].Value
        if(self.cur3 > 0.5) then
            data = "SET MicActive on\x0d"
            self.cur1 = -999
            self.cur2 = -999
        else
            data = "SET MicActive off\x0d"
        end
        self:send(data)
    end
end
end
self.MASCommTimer:Start(0.25)
end

```

```

MASNo1 = MASComm.new()
MASNo1.ipaddr = "192.168.2.17"
MASNo1:start()

```

```

MASNo2 = MASComm.new()
MASNo2.ipaddr = "192.168.2.18"
MASNo2:start()

```