English

Professional Studio Audio

Audio Equipment
&
Communication Systems

Audio Equipment
Communication Systems

DIGITAL AUDIO MIXING CONSOLE

NT Series

NT880

NT660

NT110

NT

Tamura Resource Network Technology

The ocean is where life began and evolved.
It is said that all life forms were born in the ocean and some of them returned to the ocean at the end of the cycle of their evolution.

The concept of the NT series is to be the starting point of professional creative work that can constantly evolve and to continue to be like the ocean, a place for evolution.

High-speed data transfer protocol TR-LINK

Simplification of connection between devices

TR-LINK uses a single-mode fiber-optic cable.
A single fiber-optic cable transmits and receives synchronization signals and control signals in addition to 512 ch audio signals. As a result, a synchronization-signal cable and a control-signal cable, which were required for each device in addition to the audio cable, become unnecessary, and the connection between devices can be made only by a pair of fiber-optic cables.

Easy maintenance

The maintenance of the router unit and the DSP core, which constitute the center of the system, can be made by the replacement of each unit in stead of time-consuming board exchange. Only a fiber-optic cable is used between units. Therefore, the replacement of a unit is possible even under operation without affecting the system.

32 bit floating point transmission

Conventionally, MADI was used for connection of an IO unit and an audio processing unit. In MADI, however, the transmission is performed with 24-bit fixed-point data. Therefore, even if DSP performs high precision arithmetic, some data loss cannot be avoided with the MADI transmission. In TR-LINK, on the other hand, all the audio data is transmitted in the form of 32-bit floating-point. As a result, even if an audio processing unit and an IO unit are installed at remote locations, the transmission without data loss is possible as if they are connected within a single casing, as long as they are connected by TR-LINK. In addition, analog audio input of an IO unit is converted into 32-bit data within the IO unit, and analog audio output is directly converted from 32-bit data into analog audio within the unit.

Separation of units

A large amount of data exchange occurs between the DSP module and the routing module, and therefore, the interconnecting method via the backplane in the same casing was conventionally adopted. In this method, modules are electrically connected to each other and the probability that some trouble in a module affects others could not be reduced to zero. With TR-LINK, it is possible to transmit 512 channels of audio data as 32-bit floating point and the data transmission between modules can be done with fiber-optic cables. As a result, the DSP module and the routing module can be electrically separated as completely different units. This has made it possible to minimize the risk of trouble of a device spreading throughout the system.

Hybrid Audio Processing

Higher integrated processor

The NT series adopts TAMURA's own hybrid audio processing system using the DSP and the FPGA.
The combined use of superior features of both these devices significantly improves the arithmetic operation capacity and provides a higher integrated processor with high processing performance for the NT series.
The entire system has been significantly downsized, for example, a 1U-size DSP unit can perform 256-channel audio signal processing.
Power consumption has also been considerably reduced compared with conventional systems because of higher integrated circuits and a downsized system.

Higher integrated circuit with 44-bit high-precision arithmetic operation capacity

TAMURA has developed a new algorithm that can perform a 44-bit floating-point arithmetic operation for function such as an equalizer for which sound quality is particularly important.
The distortion produced by deviation is reduced by increasing the accuracy of the arithmetic coefficient, making it possible to achieve an unprecedentedly clear and transparent sound quality.

Availability and fault tolerance

Hot standby system

The redundant system based on the hot standby system is adopted as the router unit constituting the center of the system. The operation status is always mirrored to the standby system, and when a failure occurs, the operation immediately switches over to the standby system. This makes system downtime as little as possible.

Fast start

The boot-up time from the power-off state of the entire system is about 30 seconds. Even in the unlikely event of a serious system failure, requiring restarting of the entire system, the downtime can be kept to a minimum.

Firmware-based system

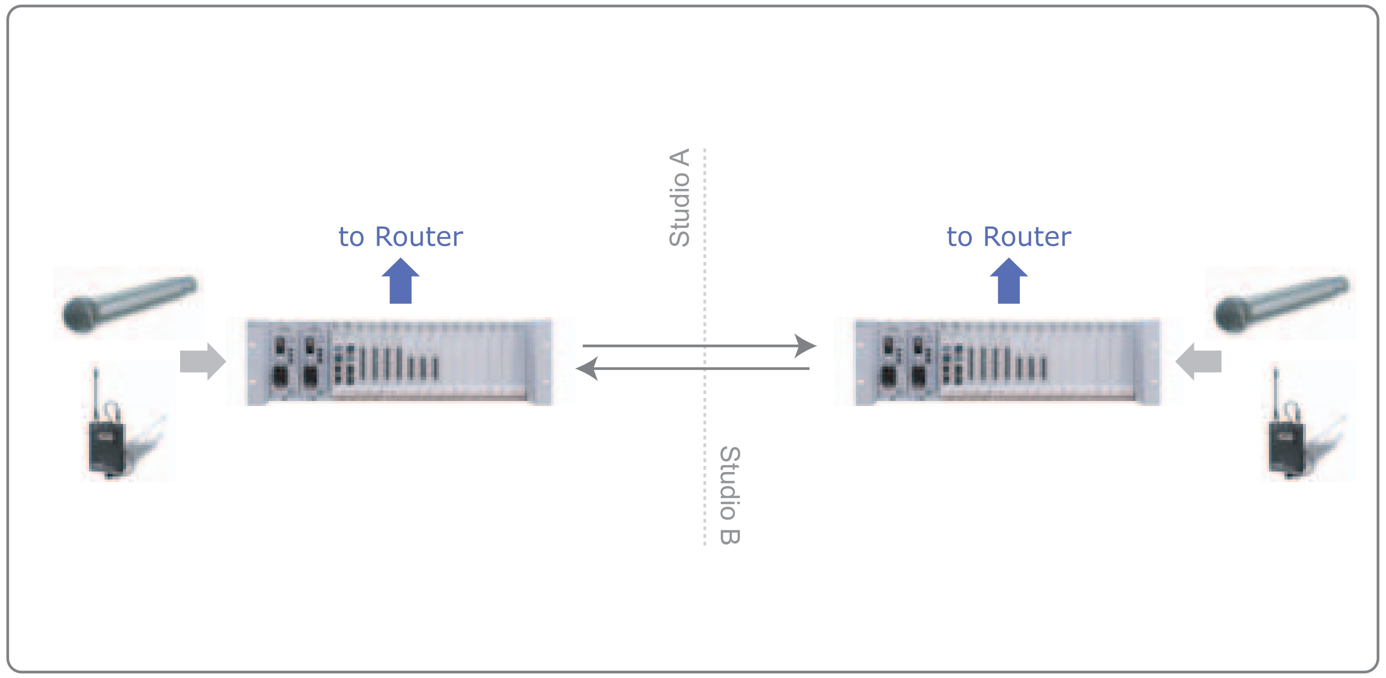
The system is built on the firmware basis without using general-purpose OS such as Windows and Linux.
Because the shutdown operation is unnecessary, the system can be immediately restarted at any time.
Because all operations are always saved in the backup memory, the previous state is recovered after a restart event. Even if the system restarts unintentionally due to sudden power-supply troubles, etc., the previous operating state is absolutely maintained.

IO Sharing

Sharing of input audio

Sharing of input audio
Audio sound inputted into a single IO frame can be shared by multiple systems. For example, this enables construction of a system in which microphones in two different studios can be used from either sides. This makes it possible to make each system an emergency backup system or use one as a premix mixer.

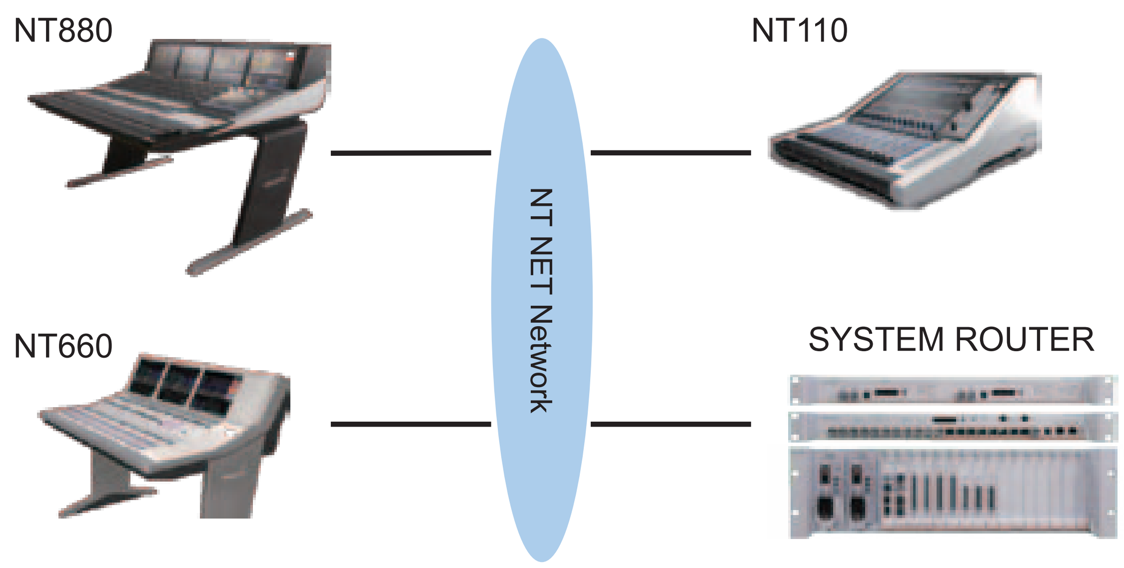
Control , such as gain control, of the microphone input shared by multiple systems can be made with any system. Also, by setting the protection control from any system, the gain control can be made only with a specific system. Audio input can be shared by at most eight systems.



NT NET System

Overview

NT NET is a function to connect the NT series with each other by network connection.
NT NET provides various functions to improve system operation efficiency and convenience.



[NT NET Network]
IP network for building NT NET system

[SYSTEM ROUTER]
Audio routing system composed ROUTER and IO FRAME of NT series

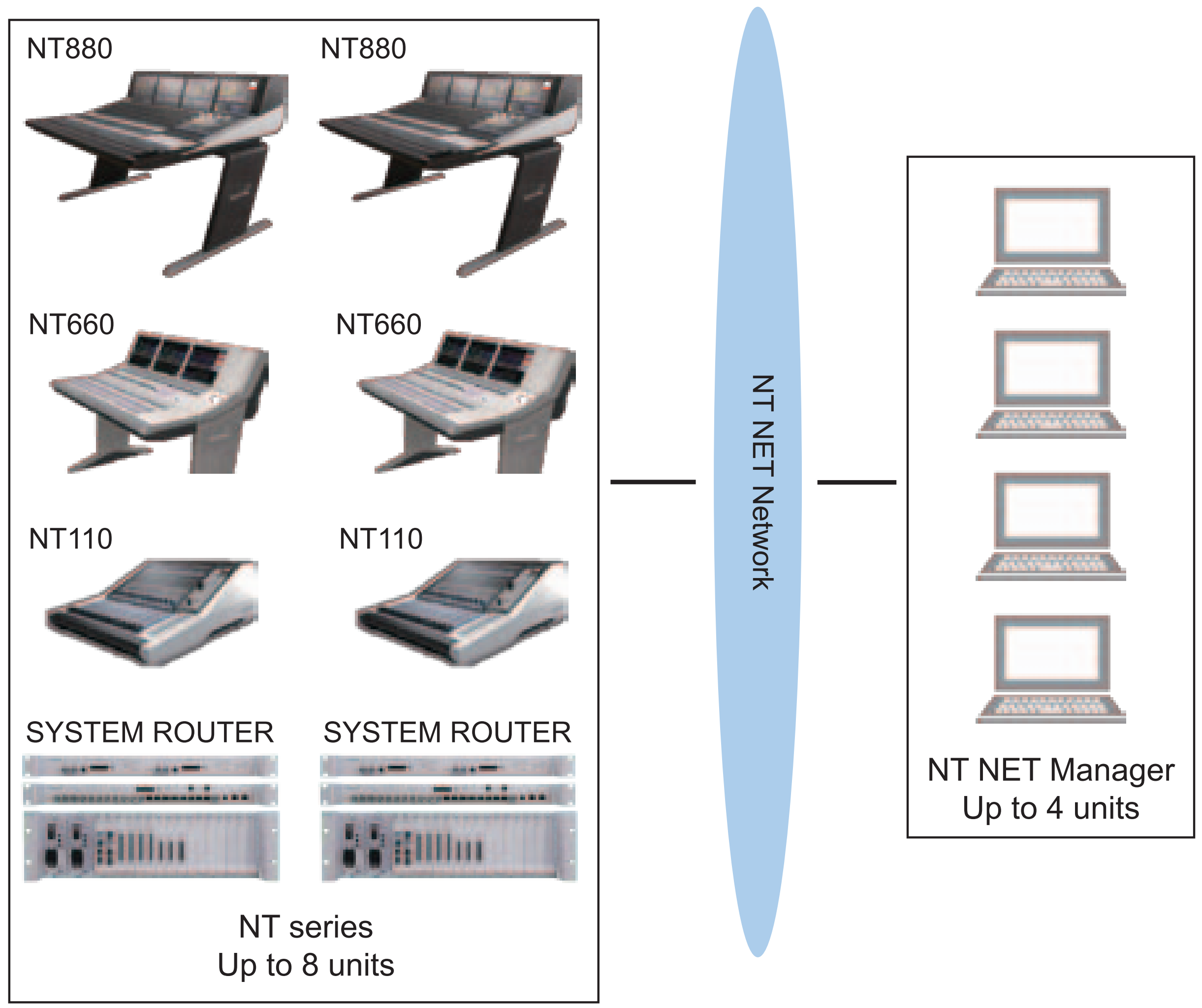
NT NET Corresponding models

Maker	Product
TAMURA	NT880 Digital Audio Mixing Console
TAMURA	NT660 Digital Audio Mixing Console
TAMURA	SYSTEM ROUTER
TAMURA	NT110 Digital Audio Mixer

* For each model, it is necessary that a software version compatible with NT NET is installed.

Maximum number of connections

The number of connected models of each model that can be connected to the network supported by NT NET is as follows.

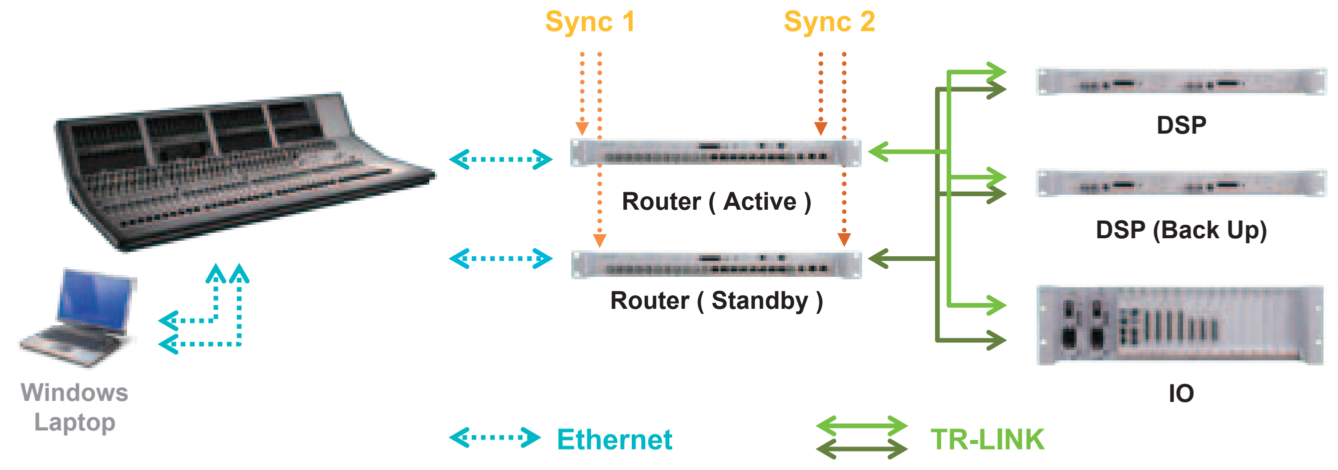


Application software for network setting and management of NT NET

*There is no restriction on combination of NT series.

Note. The maximum number of NT series connected units may be limited by each function of NT NET.
Please also refer to restrictions of each function.

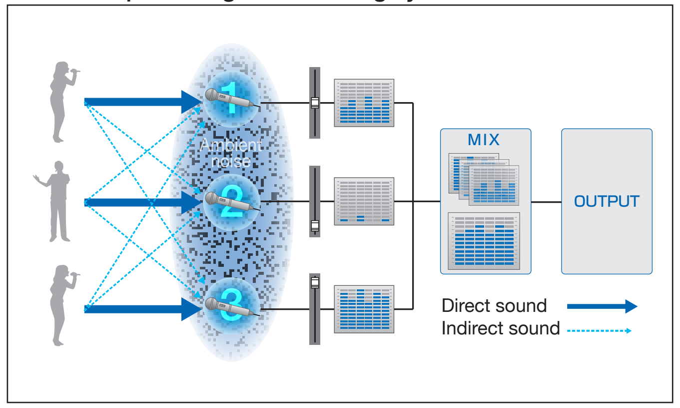
Connection diagram



AUTOMIX function

NT series AUTOMIX is a function to automate a part of the mixing operation.
At broadcasting and production sites where several microphones are used at the same time, the operator needs to control the faders of multiple microphone channels instantaneously and accurately depending on the situation.
The AUTOMIX function automates the fader operation of such microphone channels, reduces the operator's load, and thereby provides an environment where the operator can concentrate on the work of audio quality adjustment, etc.

Conceptual diagram of mixing system environment



Main Specifications of AUTOMIX

Item		Specifications
No. of Automix SHARC DSPs		Maximum 4
Specifications	No. of Automix channels	16ch
	Automix ch format	Mono
	Sample freq	FS 48k
Connection channel	Connect ch type	HA/Line Input Group M1/M2/M3
	Connect ch format	Mono/Stereo/5.1
	Connect ch signal path	Depends on the insertion path

The AUTOMIX of the NT series adopts the gain-sharing type functions and has the following features:

(1) Provides natural auditory sensation
- It does not sound like a noise gate
- No head missing at the beginning of talk
- No audible level fluctuations
- No imbalance in ambience

(2) No need for threshold-level setting
- The gate function does not operate with ambient noise even when the threshold is set to be low.
- The gate will not close even when the threshold is set to be high.
- Setting thresholds in a quiet environment does not cause problems even at the time of sudden clapping by the audience or music being played.

(3) No requirement of setting of the attack time, the hold time, etc.

(4) The state of no unnatural silence (no ambient) does not occur and the reverberation feeling is not terminated immediately after the end of talk.

(5) No unnatural disappearance of the ending of the sentence.

(6) When a new speaker starts talking, the quality of ambient noise does not change.

(7) No low-frequency pop noise due to gate operation

Specifications

System

- Sampling frequency — 48kHz / 96kHz
- Routing cross point — 10,240 x 10,240
- Maximum number of signal processing channels — 1,024ch
- Synchronous signal — Video (NTSC/PAL), Word, AES3 / AES3id
- DSP CORE — Maximum 5 DSP core units (including 1 backup unit)
- Number of TR-Link audio channels — 512ch

ROUTER

- Supply voltage — AC100-240V 50/60Hz
- Number of TR-Link ports — 20 ports
- Maximum number of signal processing channels — 1,024ch
- Synchronous signal input connector — BNC connector x 2, XLR connector x 2

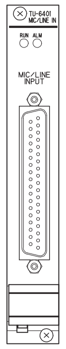
DSP CORE

- Supply voltage — AC100-240V 50/60Hz
- Number of signal processing channels — 256ch

IO FRAME

- Supply voltage — AC100-240V 50/60Hz
- Number of installed slots — 14 slots
- IO cards — 8ch Dsub MIC/LINE IN card, 8ch BNC AES IN card, HD/3G-SDI card, 8ch Dsub LINE OUT card, 8ch BNC AES OUT card, MADI IO card, GPIO card, Dante card

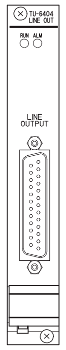
Option card



■ 8ch DSUB MIC/LINE IN Card

Audio interface card of analog 8ch input.
Mic/Line setting can be changed.

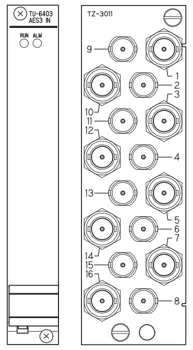
Number of Occupied slots	1 slot
Mic/Line input	balanced type
Number of Channels	8ch
[Mic input] Input level	-64dBu ~ +10dBu
[Mic input] Input inpedance	4kΩ or more
[Line input] Input level	-12~+12dBu (0.1dB step select)
[Line input] Input inpedance	600 / 10kΩ or more



■ 8ch DSUB LINE OUT Card

Audio interface card of analog 8ch output.

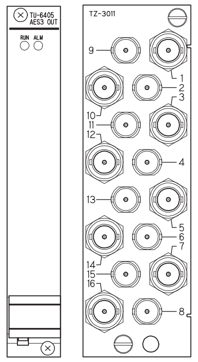
Number of Occupied slots	1 slot
Line output	balanced type
Number of Channels	8ch
Output level	-12 ~ +12dBu (0.1dB step select)
Output inpedance	55Ω



■ 8ch BNC AES3 IN Card

Audio interface card of 8 channel AES3 input. Change of the ON/OFF setting of SRC is possible.
In the 16-channel BNC REAR PANEL, 1-8 of the BNC connector corresponds to the slot on the left of two occupied slots, 9-16 to the input/output connector of the corresponding to the slot on the right.

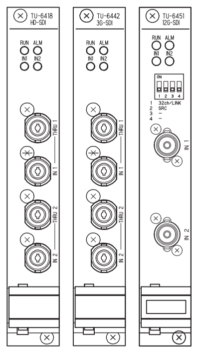
Number of Occupied slots	1 slot
Format	AES-3id
Number of Channels	8ch AES3
Input inpedance	75Ω unbalanced type
Input sampling frequency (SRC ON)	32~100kHz
Input sampling frequency (SRC OFF)	48 / 96kHz (Synchronized with the system clock)
Number of Input bits	16~24bit



■ 8ch BNC AES3 OUT Card

Audio interface card of 8 channel AES3 output.
In the 16-channel BNC REAR PANEL, 1-8 of the BNC connector corresponds to the slot on the left of two occupied slots, 9-16 to the input/output connector of the corresponding to the slot on the right.

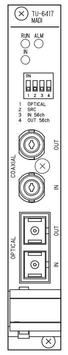
Number of Occupied slots	1 slot
Format	AES-3id
Number of Channels	8ch AES3
Output inpedance	75Ω unbalanced type
Output signal level	1 Vp-p
Output sampling frequency	48 / 96kHz (Synchronized with the system clock)
Number of output bits	24bit



■ HD-SDI Card / 3G-SDI Card / 12G-SDI Card

Interface card compatible with HD-SDI card / 3G-SDI / 12G-SDI.

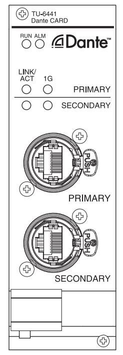
	HD-SDI Card TU-6418	3G-SDI Card TU-6442	12G-SDI Card
Number of Occupied slots	1 slot	1 slot	1 slot
Supported SDI formats	720p 50/59.94/60Hz 1035i 59.94/60Hz 1080i 50/59.94/60Hz 1080p 23.98/24/25/29.97/30Hz 1080psF 23.98/24Hz		2160/59.94p
		1080p 50/59.94/60Hz 1080psF 25/29.97/30Hz	
Embedded audio standard	SMPTE299M	SMPTE299M	SMPTE ST299
Input sampling frequency	48kHz	48kHz	48kHz
Number of Input bits	16~24bit	16~24bit	16~24bit
Number of Input channels	8ch / IN BNC	8ch / IN BNC	32ch
Loop Through output	Reclock active through output	Reclock active through output	



■ MADI Card

Audio interface card of MADI 64ch input / 64ch output.
Switching Optical In / Coaxial In, SRC ON/OFF setting, setting change of 64ch/56ch for IN/OUT are possible.

Number of Occupied slots	1 slot
Format	AES-10 / AES-10id
Number of input channels	48kHz : 56 / 64ch 96kHz : 28 / 32ch
Number of output channels	48kHz : 56 / 64ch 96kHz : 28 / 32ch
Channel alignment	Double channel
Input sampling frequency (SRC ON)	48 / 96kHz ±100ppm
Input sampling frequency (SRC OFF)	48 / 96kHz (Synchronized with the system clock)
Number of Input bits	16~24bit
Output sampling frequency	48 / 96kHz
Number of Output bits	24bit
[Coax] Input inpedance	75Ω unbalanced type
[Coax] Output inpedance	75Ω unbalanced type
[Opt] Supported optical cable	ISO/IEC 9314-3. MM 62.5/125nm Numerical Aperture 0.275



■ Dante Card

This card is audio interface card to connect to Dante network.
It is possible to maximum 64 channels input, 64 channels output.

Sampling frequency (Fs)	48kHz / 96kHz
Input / Output (Fs=48kHz)	Max 64 input, 64 output
Input / Output (Fs=96kHz)	Max 32 input, 32 output
Transmission Protocol	Dante
Dante Connector	RJ-45 type /Neutrik etherconConnector

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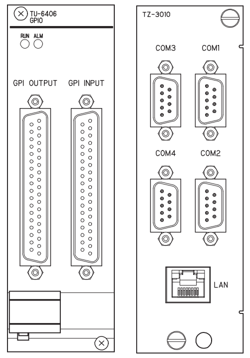


■ GPIO Card

Interface card for 24-input / 16-output of general-purpose control signals.

[GPI]

Function	Description
Link Function FU/BT	When On, the specified FU function is set to On or BT is On
Link Function Remote	When On, the specified remote function is set to On
Link Function AVL	When On, AVL function is set to On
System Tally 1	When On, indicator LAMP1 for OSC and TB prohibit control is lighted
System Tally 2	Indicator LAMP 2 is lighted
System Tally 3	Indicator LAMP 3 is lighted
Monitor Cut	When On, the specified monitor is disconnected
Monitor Dim	When On, the specified monitor is dimming
Output Matrix switching	When On, Out Source of specified TR-Link channel is altered
Send Ext Int Disable	When On, the Ext Int function of the specified Bus is disabled
Input Only	For GPI Link
TB interruption	When On, TB audio interruption is generated in the specified Bus
OSC interruption	When On, OSC interruption is generated in Master Bus
Moni Source switching	When On, Monitor Source is changed
GPI REM Sw	When On, console [REM] button is On

[GPO]

Function	Description
Link Function Remote	On output when the function is in the specified status
Console Mode Notification	On output for the specified Console Mode
OSC On Notification	On output when OSC is On
GPI Link	Output being linked with the specified GPI state
PFL On/Off Notification	Output PFL On/Off status
AFL On/Off Notification	Output AFL On/Off status
FU On	Output FU On/Off status of specified FU number
TB status Notification	Output of TB interrupt status to specified Bus
Mic On	Output of Mic On status of specified FU number

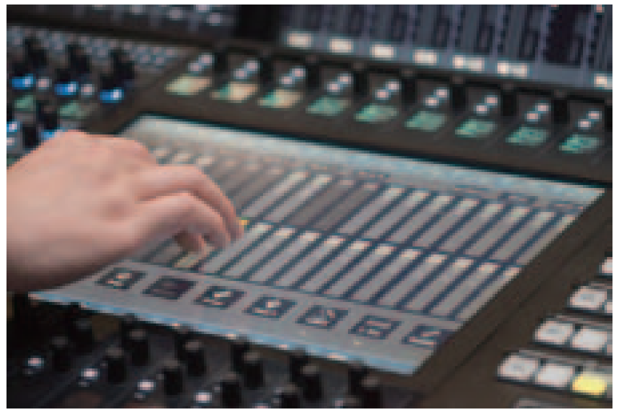
NT880

with Tamura Resource Network Technology

Excellent operability

Two parameter operation methods

Two methods are available for channel parameter operation, namely, the center-assign method, which assigns channels on a panel at a single location, and the channel-based method, which performs the operation for each channel as in the case of an analog console.
When you want to concentrate on a single channel sound, the center-assign method is most suitable because it allows you to operate all parameters at once.
On the other hand, the channel-based method is convenient when urgency is required, for example, during live broadcasting, because it allows the engineer to operate multiple channels at the same time.
These two operation methods are suitable for different situations.
NT880 allows operation using either method so that both methods can be selected in accordance with the situation and the level of preference of a mixing engineer.
For the channel-based method in particular, high operability for a quick response to the situation that changes moment by moment is achieved by placing 14 encoders per channel in order to minimize the function switching operation.



Channel layout editing functions

"Add new channels," "delete channels no longer in use," or "add a new microphone channel to existing active channels because another microphone has been added."
As in the case of these examples, it will be ideal if you can flexibly change the channel layout in accordance with the situation instead of having a channel layout that is fixed once it is set.
To enable such an operation, NT880 is provided with sophisticated channel layout editing functions (such as channel addition, deletion, copying, and cut and insert) on the touch panel.
This feature intuitively and instantaneously enables mixing engineers to set up an ideal channel layout.

Flagship model pursuing optimal ease of operation to enable high-level creative work

Expandable to large-scale systems

Number of physical faders

NT880 can be configured with up to 150 physical faders. Two or more consoles in different cabinets can be operated as a single console system as long as the number of installed faders is within the maximum number of faders. (*1)

Operator-specific section concept

When two or more engineers perform mixing operations at the same time, other engineers' work can be interrupted or their specific settings may be lost if one engineer needs to perform an operation that affects the entire console.
To prevent such inconveniences, TAMURA has introduced the section concept.
A 'section' specifies the operation range of one engineer. The extent of the effect from one operation is confined to the designated section only.
A mixing engineer can also set Pre-Fader Listen (PFL) and After-Fader Listen (AFL) solo functions independently for each section. Therefore, engineers are provided with conditions under which they appear to be working on separate console systems.
One console system can be divided into a maximum of four sections.

High-performance processor

NT880 has a control system that is built on the assumption of simultaneous operations by two or more engineers.
Even if there is a simultaneously imposed workload of four engineers, there is no delay in the response time of the operation panel displays or the adjustments made for audio.

(*1) There are cases in which restrictions apply to the installation position, distance, and other factors.

Specifications

Console

Supply voltage	AC100-240V 50/60Hz
Maximum number of physical faders	150 faders
Bank / Layer	6Bank / 2Layer
Number of fader groups	32Group

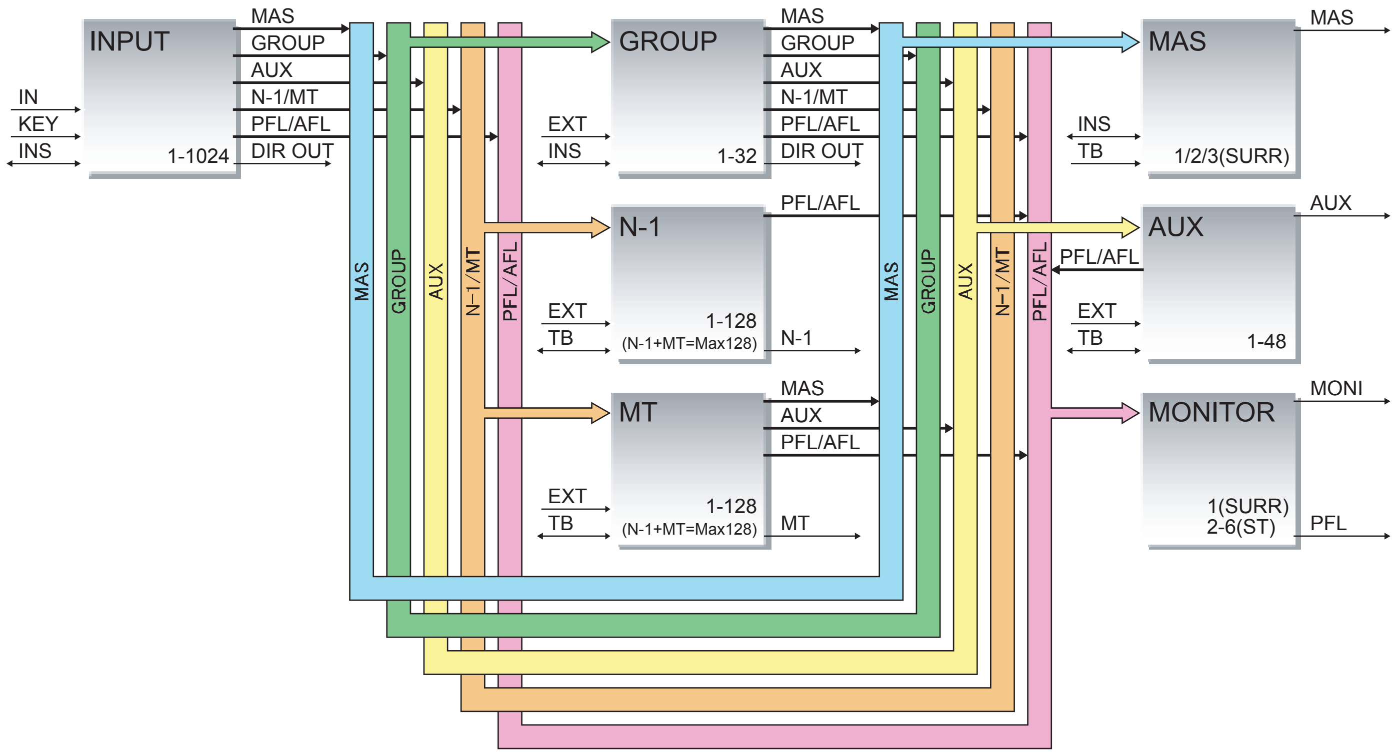
Audio channel (Fs=48kHz)

Master Bus	Maximum 24 buses (3 surround)
Group Bus	Maximum 32 buses
Aux Bus	Maximum 48 buses
N-1 / MT Bus	Maximum 128 buses
AFL	1 surround
AFL / PFL	3 stereo
PFL	1 stereo
Main Monitor	1 surround+stereo
Sub Monitor	5 channels (Stereo)

Audio control parameters

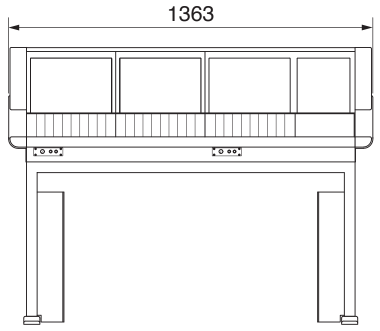
HA Gain	+10dBu~-64dBu
Trim	+24dB~-24dB
Delay	5000ms or more
Filter	Filter1 (HPF/Notch) Filter2 (LPF/Notch)
Equalizer	4Band (Support for all frequency bands)
Dynamics	Compressor 2 channels Gate/Expander 1channel

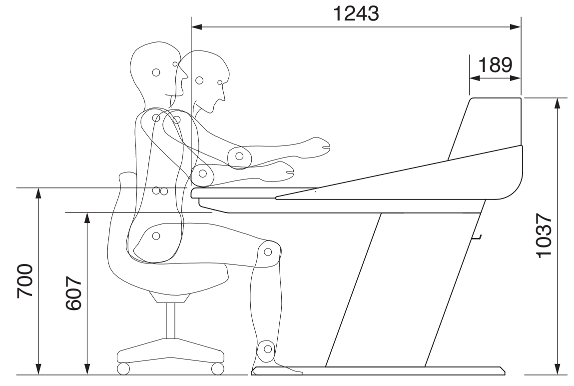
Audio block diagram



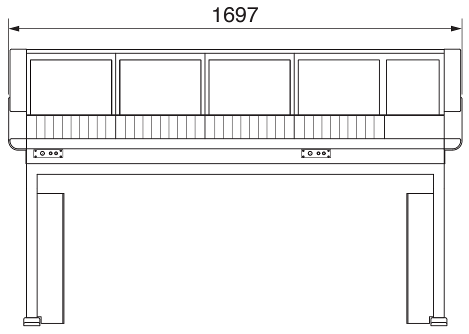
Dimensions

30 Fader

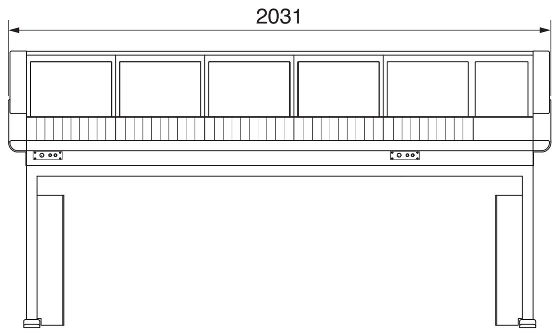




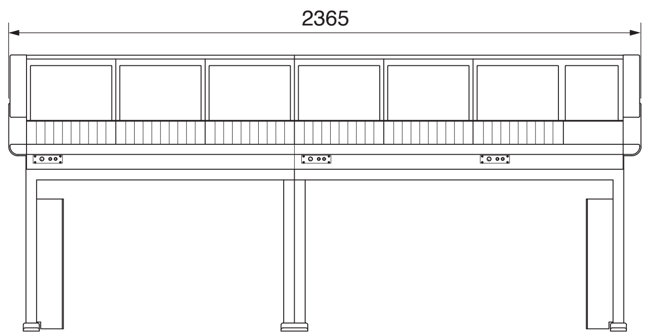
40 Fader



50 Fader



60 Fader



NT660

with Tamura Resource Network Technology

Flexible Operation

New parameter operation method

The arrangement of seven encoders on the compact surface allows channel-oriented operation, which is useful in urgent situations such as live broadcast.
Also, the equipment uses a new operation method, bay-oriented operation, in order to allow the user to concentrate on controlling one channel in hand.
In bay-oriented operation, functions to control channel parameters are incorporated into all the encoders in the same bay as that of the channel. This allows simultaneous access to most of the parameters on a channel.
You can freely switch between these two operation methods, instead of configuring initial settings to select either of them. It is possible to select the appropriate method according to the circumstances, which can realize efficient creation of contents.
When using all channel parameters, you can perform center assign operation, through which parameters are comprehensively manipulated on the touch panel.

Touch Panel Surround Panner

In order to support creation of high-level surround sound, it has been made possible to perform surround panning with the touch panel.
You can select mouse mode, which determines the pan position by taking into consideration in what direction and at what distance you drag, in addition to normal mode, in which the exact touched position is specified as the pan position.
Also, the use of the Pan Link function allows you to automatically specify the pan position of the R-side microphone according to that of the L-side microphone when using two monaural microphones as a stereo pair.
The equipment supports creation of surround sound during a broadcast requiring immediate responses, not simply by replacing a joystick but by allowing comfortable operation.

High-spec Compact Model Where Functions of the Highest-grade Model Are Kept within Reach

Greatly Enhanced Functions

Inheriting Enhanced Functions

You can use the same sound processing parameters as those of the higher-grade model NT880.
Two compressors are used for each individual channel, and algorithms for full-four-band EQ and the like are exactly the same.
Also, the equipment has a delay compensation function for multistage bus assignment, allowing creation of detailed sound.

User Level Setting

The equipment has the Administrator Lock mode, which limits the range of operation.
When an operator who does not understand the entire audio system, such as a director, uses the equipment, this mode can disable, in advance, functions that may lead to fatal erroneous operation.

Consolidated Control of Bus Outputs

As the process of content creation is becoming more complicated, the number of bus outputs to be monitored is increasing.
In an environment where installation spaces are limited, it may be difficult to arrange external meter units.
Therefore, the equipment is capable of simultaneously displaying the meter readings of 80 buses in the bottom of the channel meter.
You can always display the output meter readings of buses to be monitored without changing the screen or settings.
Since the operator can at any time freely change the buses to be metered, it is possible to build an appropriate metering system according to the circumstances.

DAW Control Functions

In order to ensure efficient use of facilities, post-production work is occasionally carried out even in a sub broadcast studio.
The equipment is compatible with DAW control functions in order to support post-production work in a sub studio. (Option)
Channels for DAW control are not held in the same specific layer, but can be freely placed in any bank and any layer, similarly to normal audio channels.
For example, on the same control surface, it is possible to control music tracks with a DAW while operating a narration recording microphone.
The equipment also incorporates other functions to support complicated post-production work, such as automation mode control and track arming.

Specifications

Console

Supply voltage	AC100-240V 50/60Hz
Maximum number of physical faders	20/30/40/50 faders
Bank / Layer	6Bank / 2Layer
Number of fader groups	32Group

Audio channels (Fs=48kHz)

Master Bus	Maximum 24 buses (3 surround)
Group Bus	Maximum 32 buses
Aux Bus	Maximum 48 buses
N-1 / MT Bus	Maximum 128 buses
AFL	1 surround
AFL / PFL	3 stereo
PFL	1 stereo
Main Monitor	1 surround+stereo
Sub Monitor	3 channels (Stereo)

Audio control parameters

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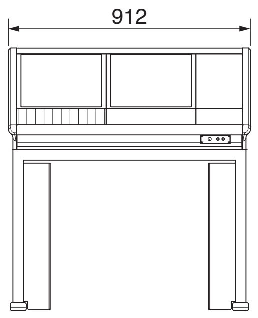
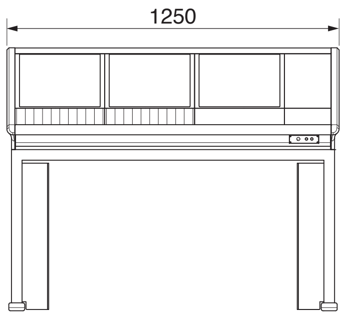
Delay	5000ms or more
Filter	Filter1 (HPF/Notch) Filter2 (LPF/Notch)
Equalizer	4Band (Support for all frequency bands)
Dynamics	Compressor 2 channels Gate/Expander 1channel

Dimensions

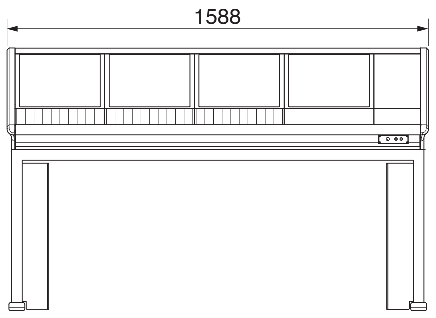
20 Fader



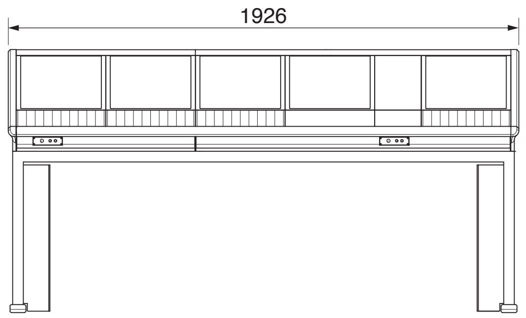
30 Fader

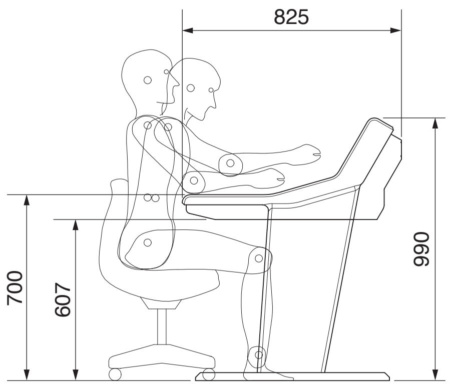


40 Fader



50 Fader





NT110

Digital Audio Mixer

Operability of trust

- 16 Analog input/output (MONO), AES3id2 input/output (STEREO), and 2 auxiliary input (STEREO) as the standard equipment
- Audio formats such as AES, MADI, SDI, DANTE can be linked with NT016 via 2 expansion slots. (option)
- External Remote Control for Input/Output can be achieved via GPIO cards installed in the expansion slots for various applications.

- Compact design mountable in a EIA 19 inch width rack
- 2 Layer (AB sides per each Layer), 3 Banks enable 80ch Logic CH with 16ch physical faders incorporated.
- Incorporated Surround sound monitor output enables Surround sound product ion at Outside broadcasting field.
- 2 sets of NT016 can be Cascaded to have a Physical 32ch. Fader Console (option)

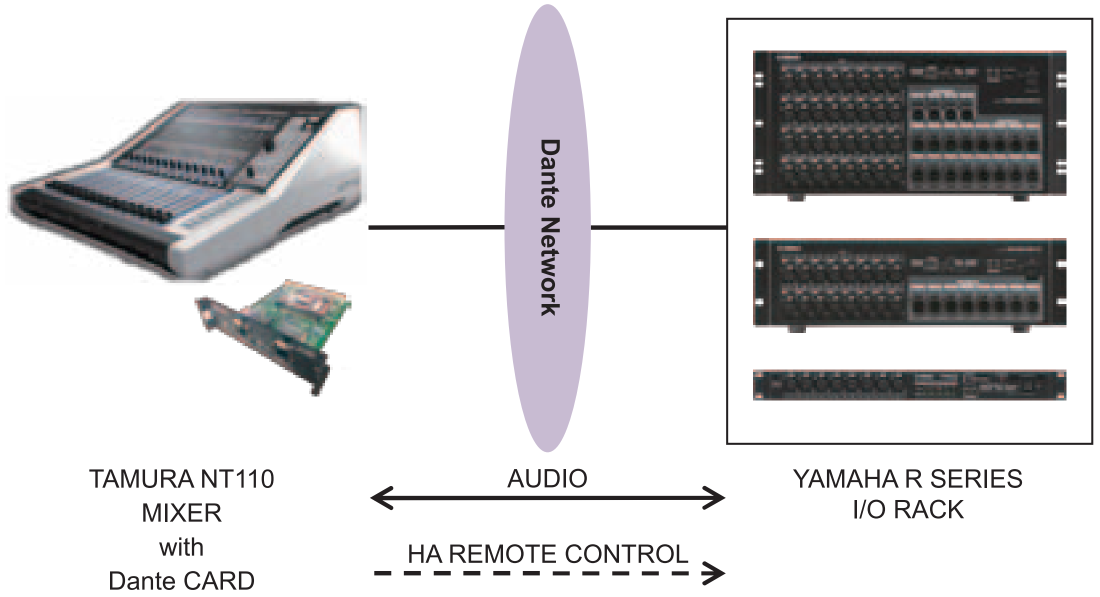
- Sampling Frequency;48K/96KHz, selectable for High-quality audio program production.
- Availability of Power redundancy which is a prime requirement for live broadcasting events for the highest reliability.
- Audio digital signal processing redundancy despite of the size of portable mixer. (option)

Portable Model with Inherited Functions RE-Liability of NT Series

Rio Remote

Overview

Rio Remote is a function to remotely control the head amp parameters(HA GAIN and +48V) of YAMAHA R SERIES I/O RACK from NT110 via Dante.
By connecting NT 110 and R SERIES to the same Dante network.
it remote control the head amp parameters of R SERIES in real time while mutually transmitting voice.



Corresponding models

As of October 2018

Maker	Product	
TAMURA	NT110 Digital Audio Mixer	Digital Audio Mixer
TAMURA	TU-6439 Dante CARD	NT110 Dante Add port card
YAMAHA	Rio3224-D	I/O RACK
YAMAHA	Rio1608-D	I/O RACK
YAMAHA	Ri8-D	I/O RACK
YAMAHA	Rio3224-D2	I/O RACK
YAMAHA	Rio1608-D2	I/O RACK

Multi Meter

Corresponding models

This is a multi meter that measures and displays the LOUDNESS value, VU value, PEAK value of the input signal.
AES 3 - 2009 and LTC (TIME CODE), GPI are carried in the input.

Main function

- Various LOUDNESS calculation display
- VU / PEAK / TRUE PEAK indication
- LTC indication
- AES 3 - 2009 digital audio input
- GPI (start / stop / pause of average LOUDNESS operation and load of PRESET)

Specifications

› Overall Rating

■ Dimensions (without Side panel)	490(W)×222(H)×606(D)mm (Protruding parts not included) 430(W)×220.5(H)×550(D)mm (FRONT/SIDE PANEL not included)
■ Weight	16.5 kg
■ AC	100 - 240V, 50/60Hz
■ DC	12V/14.8V
■ Power Consumption	150W
■ Operating free-air temperature range	-10~ 40˚C
■ Number of faders	16 Fader
■ Bank/Layer	3Bank/2Layer

› Audio Channels (Fs=48kHz)

■ Master Bus	2ch (5.1Surround+STEREO)
■ Summing Bus	16ch (MONO)
■ AFL Bus	1ch (5.1Surround+STEREO)
■ PFL Bus	1ch (Stereo)
■ Monitor Out	1ch (5.1Surround)
■ Headphone Out	1ch (Stereo)

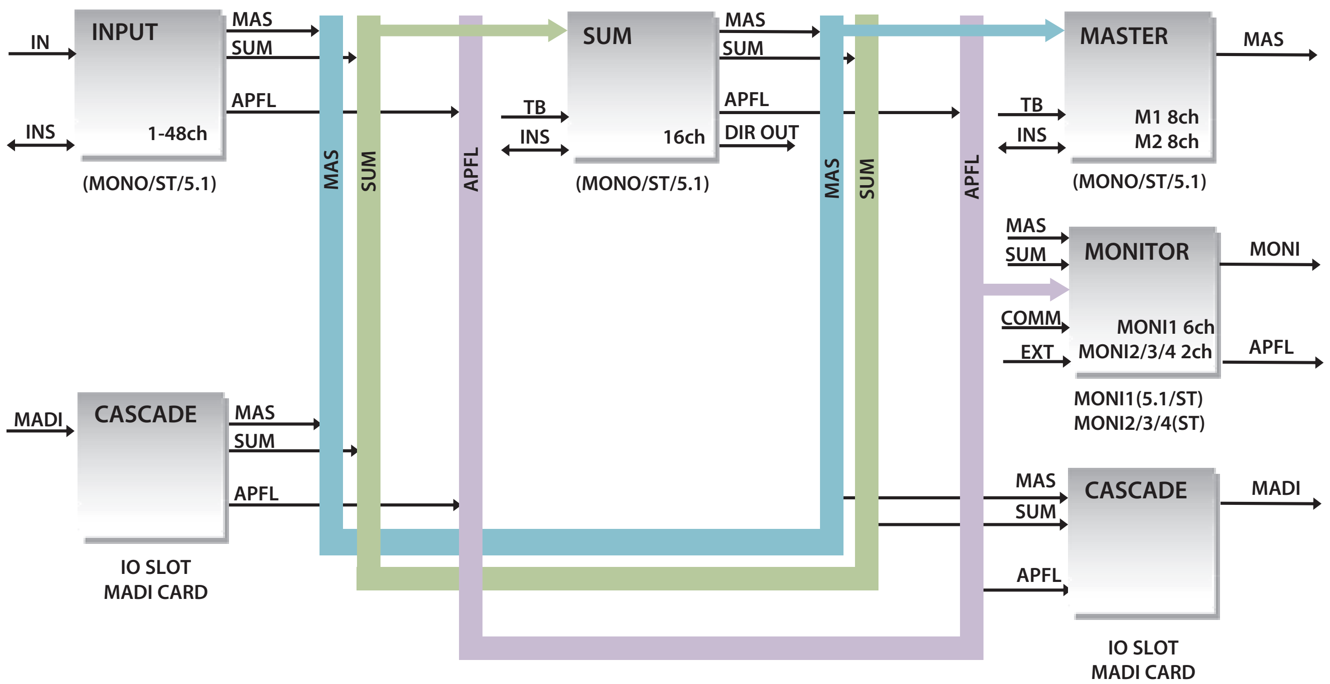
› Audio control parameters

■ Audio Reference Input Level	
(Analog MIC)	+10 ~ -64dBμ
(Analog LINE)	+4dBμ
■ Audio Reference Output Level	
(Analog LINE)	+4dBμ
■ Audio Reference Input/Output Level	
(Digital)	+10dBFS / -64dBFS
■ HA Headroom	20~30 dB

› Option

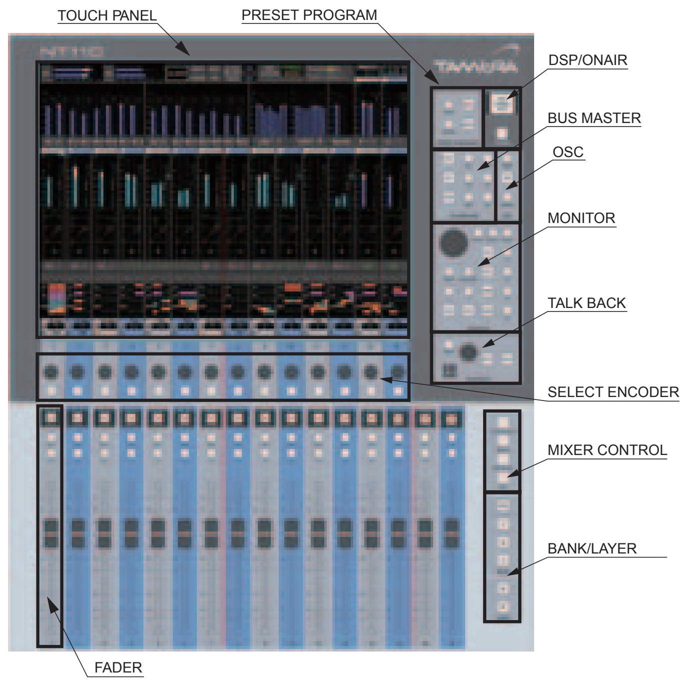
- ■ DSP CARD (BACK UP REDUNDANT DSP)
- ■ OPTION CARD
  - AES3id IO CARD (4ch IN + 4ch OUT BNC)
  - GPIO CARD
  - MADI CARD (1Coax/1Opt)
  - Dante CARD
  - MIC / LINE IN CARD
  - LINE OUT CARD
- ■ Multi Meter
- ■ Storage case

Audio block diagram

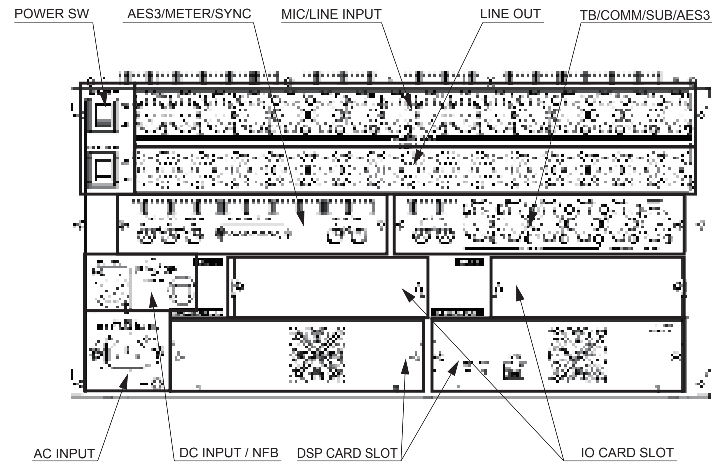


Control Panel Description

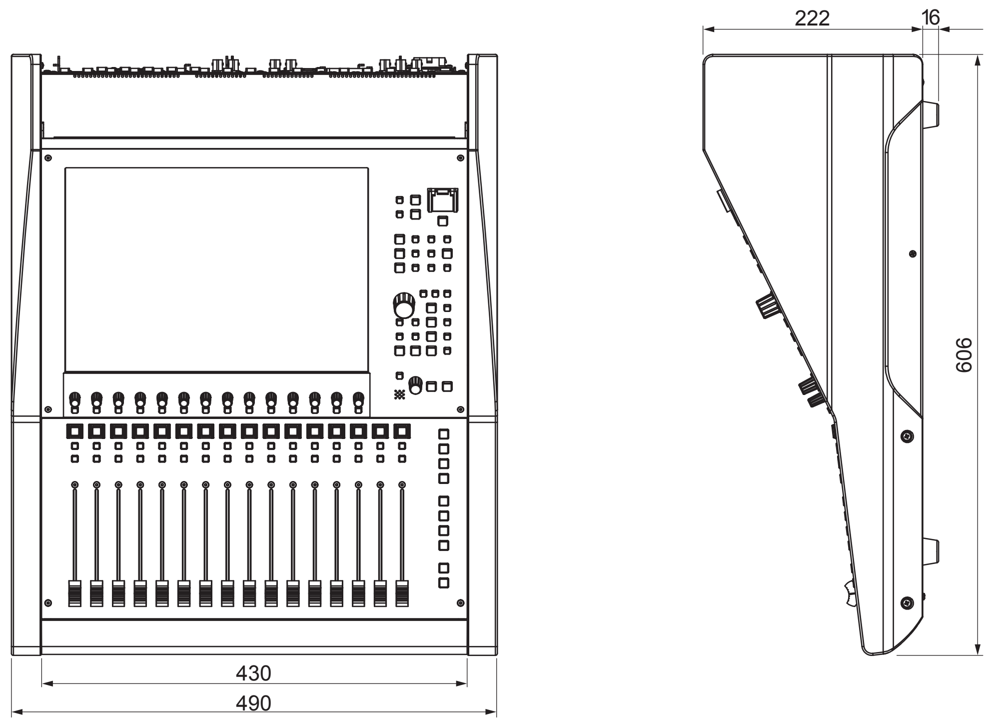
■ Front panel



■ Rear panel



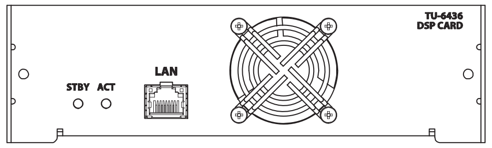
Dimensions



Option card

■ DSP Card

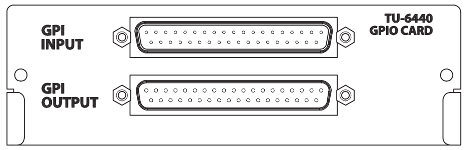
This card is a card with built in audio signal processing, audio routing and control functions.
It is possible to form a redundant system is to implement a card.



Sampling frequency (Fs)	48kHz / 96kHz
LAN Connector	RJ-45 type
Dimensions	171(W)×49.5(H)×304(D)mm
Weight	590g

■ GPIO Card

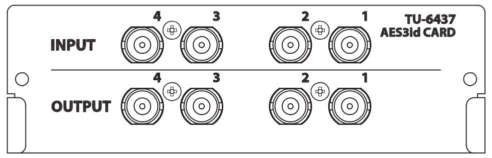
Interface card for 16-input / 16-output of general-purpose control signals.



General-purpose control signal inputs(GPI INPUT)	16ch electrically isolated opto-coupler inputs 37-pin D-type connector(male)
General-purpose control signal outputs(GPI OUTPUT)	16ch open-collector outputs 37-pin D-type connector(female)
Dimensions	129(W)x40(H)x152(D)mm
Weight	168g

■ AES3id Card

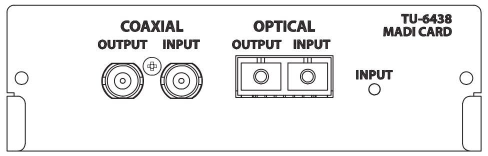
Audio interface card of 4ch AES3 input / 4ch AES3 output.



AES3 id INPUT	
Format	AES3id compliant
Number of Channels	4ch AES3
Input Sampling frequency	48kHz/96kHz (SRC Off) 30kHz~100kHz (SRC On)
Number of input bits	16~24bit
Connector	BNC(Coaxial/75Ω) x4
AES3 id OUTPUT	
Format	AES3id compliant
Number of Channels	4ch AES3
Output Sampling frequency	48kHz/96kHz
Number of output bits	24bit
Connector	BNC(Coaxial/75Ω) x4
General	
Dimensions	129(W)x40(H)x152(D)mm
Weight	210g

■ MADI Card

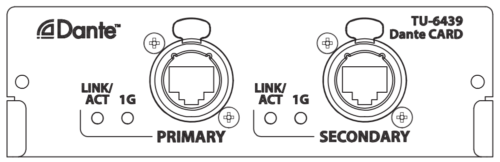
Audio interface card of MADI 64ch input / 64ch output



Format	AES10 compliant
Input Sampling frequency	48kHz/96kHz (SRC Off) 48kHz/96kHz±100ppm (SRC On)
Output Sampling frequency	48kHz/96kHz
Number of input bits	16~24bit
Number of output bits	24bit
Number of input channels	64ch/56ch (Fs 48kHz) 32ch/28ch (Fs 96kHz)
Number of output channels	64ch/56ch (Fs 48kHz) 32ch/28ch (Fs 96kHz)
Coaxial Connector	BNC (Coaxial/75Ω)
Optical Connector	MM 62.5/125μm (SC Connector)
Dimensions	129(W)x40(H)x152(D)mm
Weight	180g

■ Dante Card

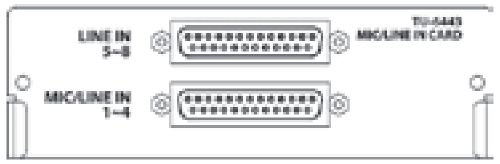
This card is audio interface card to connect to Dante network.
It is possible to maximum 64 channels input, 64 channels output.



Sampling frequency (Fs)	48kHz / 96kHz
Input / Output (Fs=48kHz)	Max 64 input, 64 output
Input / Output (Fs=96kHz)	Max 32 input, 32 output
Transmission Protocol	Dante
Dante Connector	RJ-45 type /Neutrik etherconConnector
Dimensions	129(W)x40(H)x152(D)mm
Weight	150g

■ MIC/LINE IN CARD

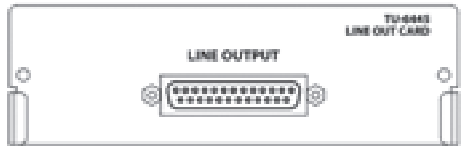
This card is audio interface card that inputs microphone level and line level analog audio signals.



MIC/LINE INPUT(CH1~CH4)	
Audio Reference Input Level	-64dBu - +10dBu
Headroom	20 - 36 dB
Input inpedance	More than 4kΩ
Phantom power supply(1ch)	48V/10mA
LINE INPUT(CH5~CH8)	
Audio Reference Input Level	0dBu/+4dBu
Input inpedance	More than 10kΩ
General	
Transmission frequency range (Fs=48kHz)	20 - 20,000Hz
Transmission frequency range (Fs=96kHz)	20 - 40,000Hz
Sampling frequency (Fs)	48kHz/96kHz
Dimensions	129(W)x40(H)x152(D)mm
Weight	210g
Connector	25pin D-type connector(female)x2

■ LINE OUT Card

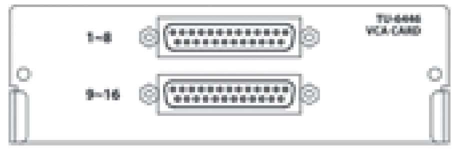
This card is audio interface card that outputs line level analog audio signals.



LINE OUTPUT(CH1~CH8)	
Audio Reference Input Level	0dBu/+4dBu
Output inpedance	less than 55Ω
General	
Transmission frequency range (Fs=48kHz)	20 - 20,000Hz
Transmission frequency range (Fs=96kHz)	20 - 40,000Hz
Sampling frequency (Fs)	48kHz/96kHz
Dimensions	129(W) x 40(H) x 152(D)mm
Weight	190g
Connector	25pin D-type connector (female)

■ VCA Card (for NT MATRIX)

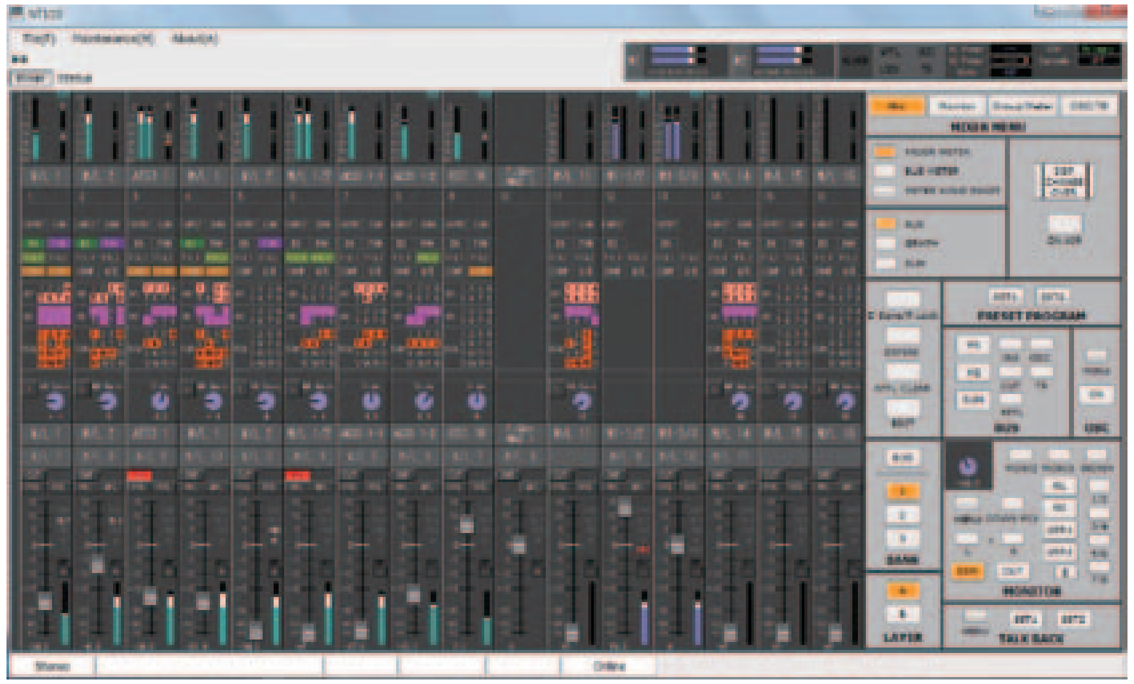
Interface card for input of VCA control signals.



VCA INPUT(CH1~CH16)	
Reference voltage	+5V DC
Compatible potentiometer	Linear curve, 10kΩ
General	
Dimensions	129(W) x 40(H) x 152(D)mm
Weight	160g
Connector	25pin D-type connector (male) x2

NT MIX

NT Mix (Windows software, free of charge) is used for displaying the touch panel of the NT110, establishing various settings, restoring the settings, etc.



Mixing operation of NT110 is performed in the Mixer menu.
When the connection status is Online, this screen operates linked to the control panel of the NT110 and can be used as a redundant control panel during operation.
The selection of Bank / Layer on this screen is independent from NT110. Therefore, it can also be used as an extended fader panel when the number of physical fader on NT110 is insufficient.

NT Mix download page
https://www.tamura-ss.co.jp/jp/

NT MATRIX

Audio Interface Unit

Overview

NT MATRIX is a system interface with a built-in DSP processor that performs routing matrix, mixing, and various processing of audio signals. It supports various forms of use by combining audio input and output cards and control cards. It also supports redundancy of power supply input and redundancy of audio signal processing unit (optional), and therefore is ideal for relaying, live broadcasting, program recording, and television and radio studios requiring high reliability.

Features

Function - rich function

(1) 160 ch x 160 ch AUDIO MATRIX ROUTER
(2) Configurable DSP audio signal processing
(3) Six card slots (two of which support 64 ch audio input and output)
(4) Analog, digital audio I/O cards and option for GPIO and VCA control cards
(5) LOGIC function for logical setting of button ON/OFF status of GPIO and the touch panel
(6) GUI application that allows flexible configuration of user interface
(7) Size appropriate for mounting on EIA 19 inch rack

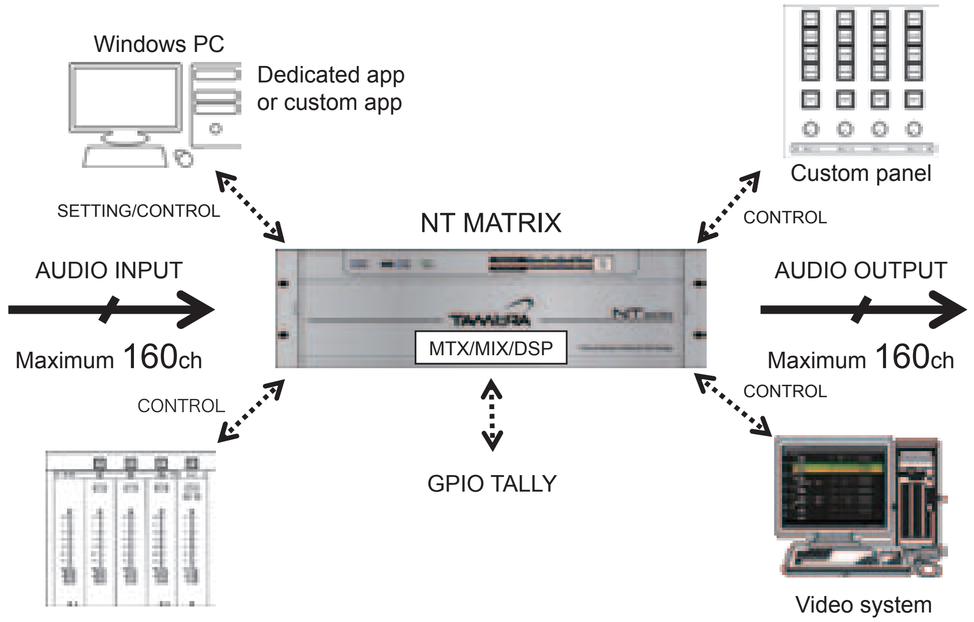
Original technologies - advanced technology

(1) Built-in high dynamic range audio signal processing by 32-bit floating point arithmetic. Mixing without considering the internal level diagram is possible
(2) Selection of the signal processing function by the DSP configuration

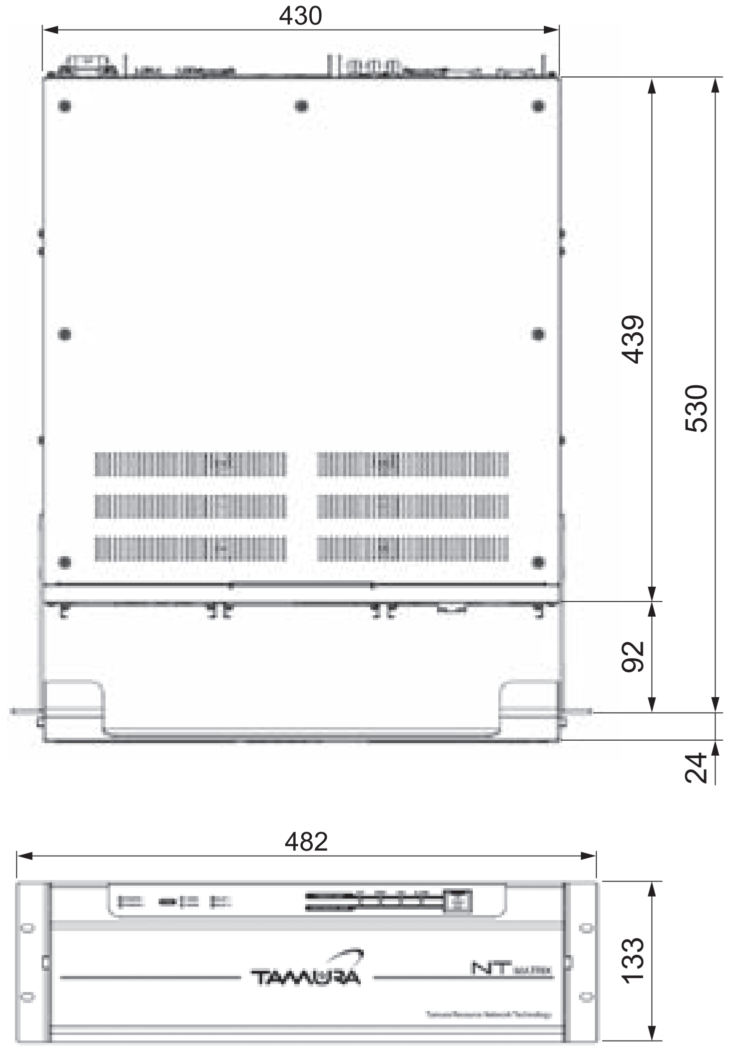
Operational safety - high safety

(1) Redundant configuration with two AC inputs for power supply
(2) DSP CARDs equipped with audio signal processor and controller can be redundantly configured to support various applications requiring high reliability (optional)
(3) The system is constructed on a firmware basis that does not use advanced OS and therefore achieves high stability and high-speed startup

NT MATRIX System



Dimensions



Custom UI

- Equipped as standard with GUI software that can customize operation parts
- DSP parameters are freely assigned to operation parts
- Parts such as buttons, faders, meters, lamps, texts, etc. are available as operation parts
- Customization is possible for the operation parts such as their color, characters, and sizes
- Graphical and design-friendly GUI can be constructed by the bitmap import function
- Test of GUI operation in the offline environment by EMULATE MODE

Specifications

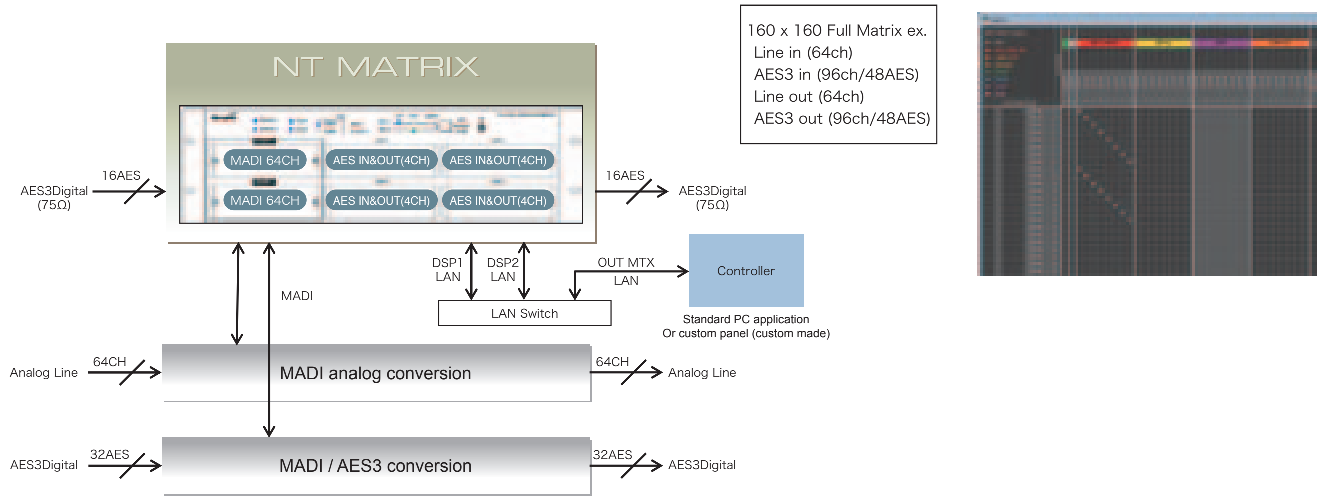
Items	Specification
AUDIO ROUTER	160ch x 160ch
DSP PROCESS	32in x 32out DSP x 6
DSP FUNCTION	32in x 32out Mix Matrix or Filter/Limitter , AUD , Internal OSC
CONTROL PORT	LAN/RS422SERIAL/GPIO/VCA
SYNCHRONIZED INPUT SIGNAL	WORD CLOCK/VIDEO
POWER SUPPLY	AC100-240V 50/60Hz
OPERATION TEMPERATURE	-10 ~ 40 ºC
EXTERNAL DIMENSIONS (WxDxH)	482 x 554 x 133

Option (Common to NT110 refer to P.30~31)

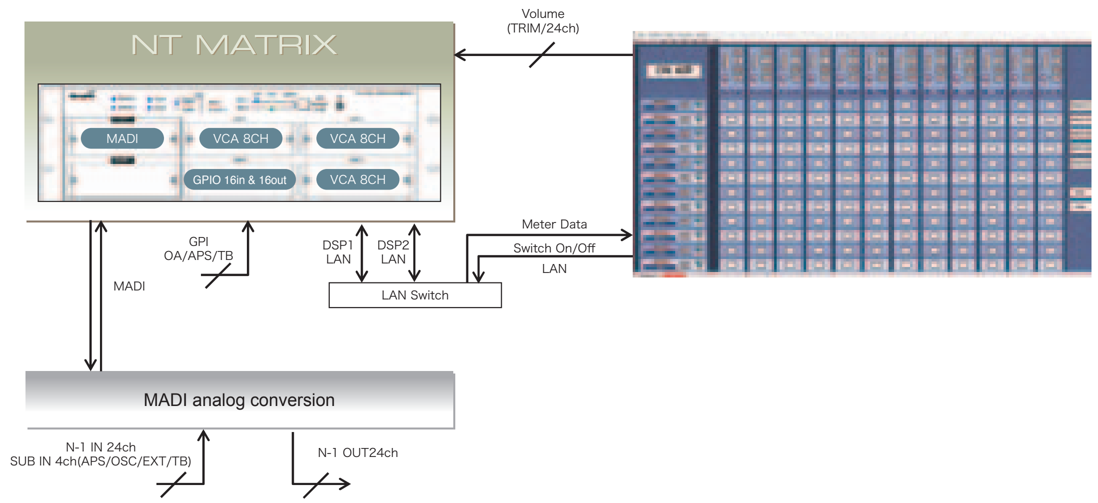
Items	Specification
DSP CARD	Redundancy DSP CARD
MIC/LINE INPUT CARD	MIC INPUT 4ch + LINE INPUT 4ch
LINE OUTPUT CARD	LINE OUTPUT 8ch
AES3id CARD	AES3id INPUT 4ch + AES3 id OUTPUT 4ch
MADI CARD	MADI INPUT 1ch + MADI OUTPUT 1ch(OPTICAL & COAXIAL)
Dante CARD	Dante 1ch (Primary & Secondary)
GPIO CARD	GPI INPUT 16ch + GPI OUTPUT 16ch
VCA CARD	VCA INPUT 16ch

Example of application

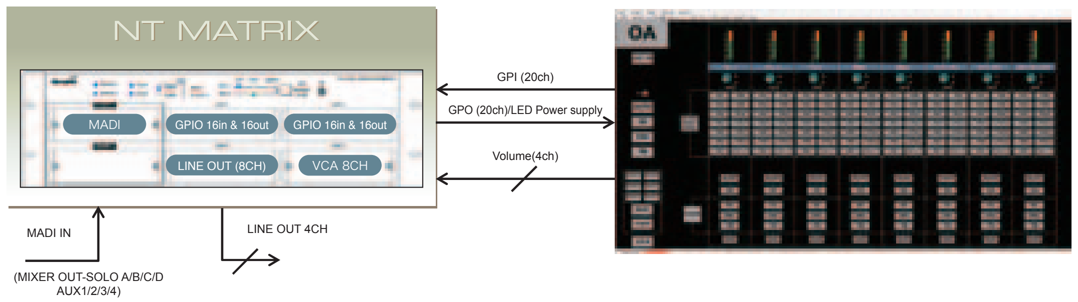
Audio Router(Matrix)



N-1 Sending back system

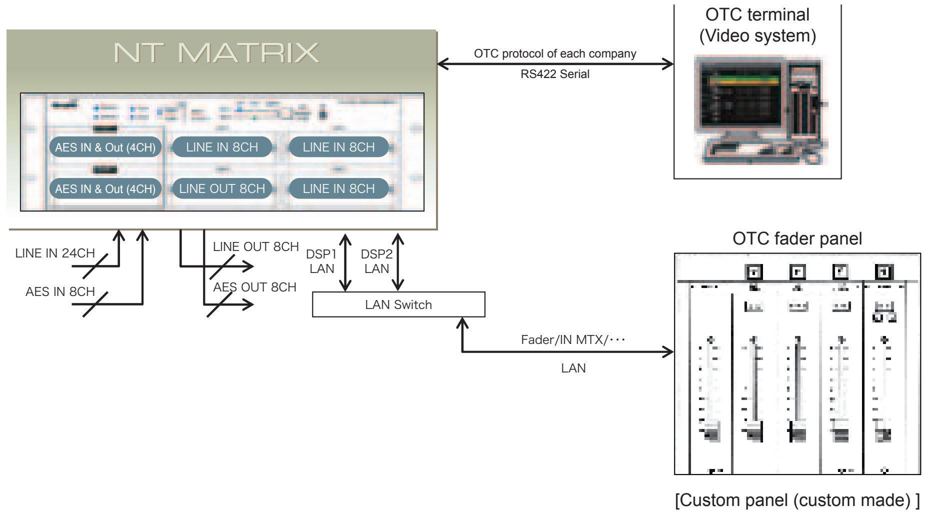


Output Matrix



Example of application

OTC system



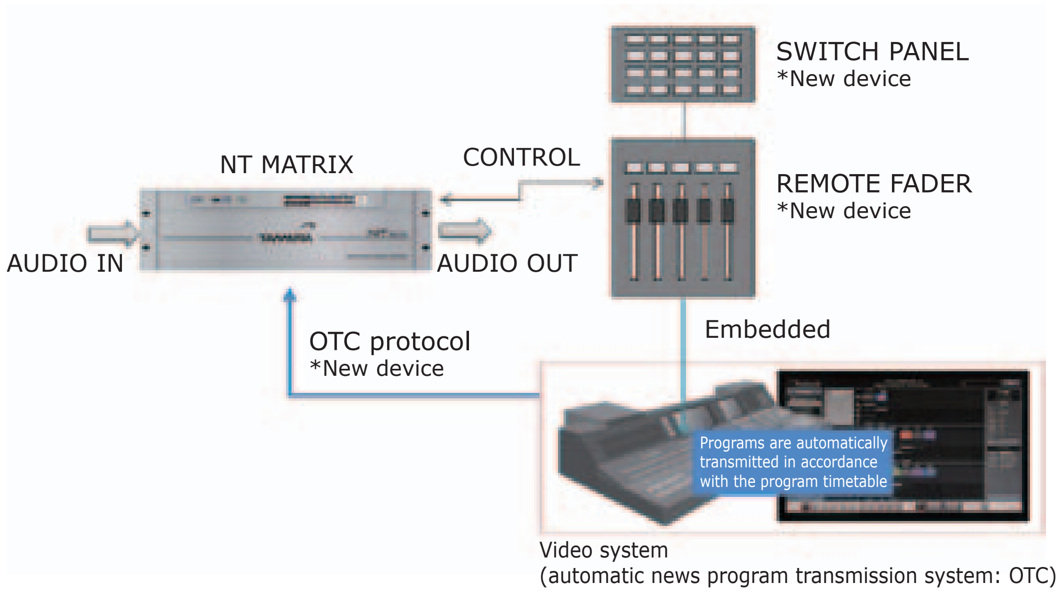
NT MATRIX OTC System

Overview

The NT MATRIX OTC system is equipped with a voice processing function for the automatic news program transmission system. It receives instruction for the audio part from the OTC system (video system), which automatically transmits programs in accordance with the program timetable (scenario, que sheet), and switches the audio MATRIX or controls the volume using the fader.

The NT MATRIX receives control information from the OTC system and processes audio.

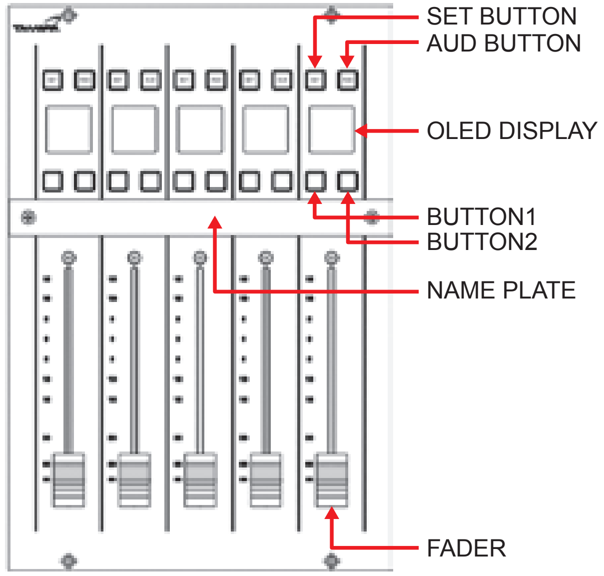
The audio can be controlled manually by operating the REMOTE FADER (OTC fader) or SWITCH PANEL. In addition to the REMOTE FADER, the CUSTOM UI for the NT MATRIX also allows audio control parameters to be manipulated manually.



2. Function

2-1 REMOTE FADER

TU-6453 REMOTE FADER is a 5-channel operation panel equipped with a motor fader. The display which displays the source name of a channel, fader name, and fader level, etc. and the illumination type pushbutton switch are carried.



(1) SET BUTTON

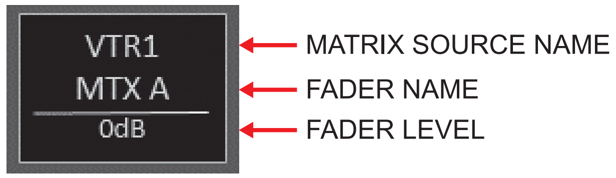
When the SET button is pressed while the TU-6454 SWITCH PANEL button is selected, the source defined for the SWITCH PANEL button is set as the input source for the channel.
The set source name is displayed in OLED DISPLAY.

(2) AUD BUTTON、BUTTON 1、BUTTON 2

The use of this button is defined on the connection destination device. Button operation can be set with DIP SWITCH on the rear panel.

(3) OLED DISPLAY

Displays channel information and REMOTE FADER information.



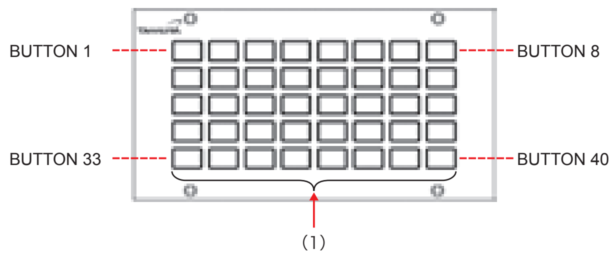
[MATRIX SOURCE NAME]
The name of the input source to the channel is displayed.

[FADER NAME]
Displays the name of the fader channel.

[FADER LEVEL]
Displays the level value set by the fader.

2-2 SWITCH PANEL

TU-6454 SWITCH PANEL is an operation panel equipped with 40 illuminated pushbutton switches.



(1) BUTTON 1 to 40

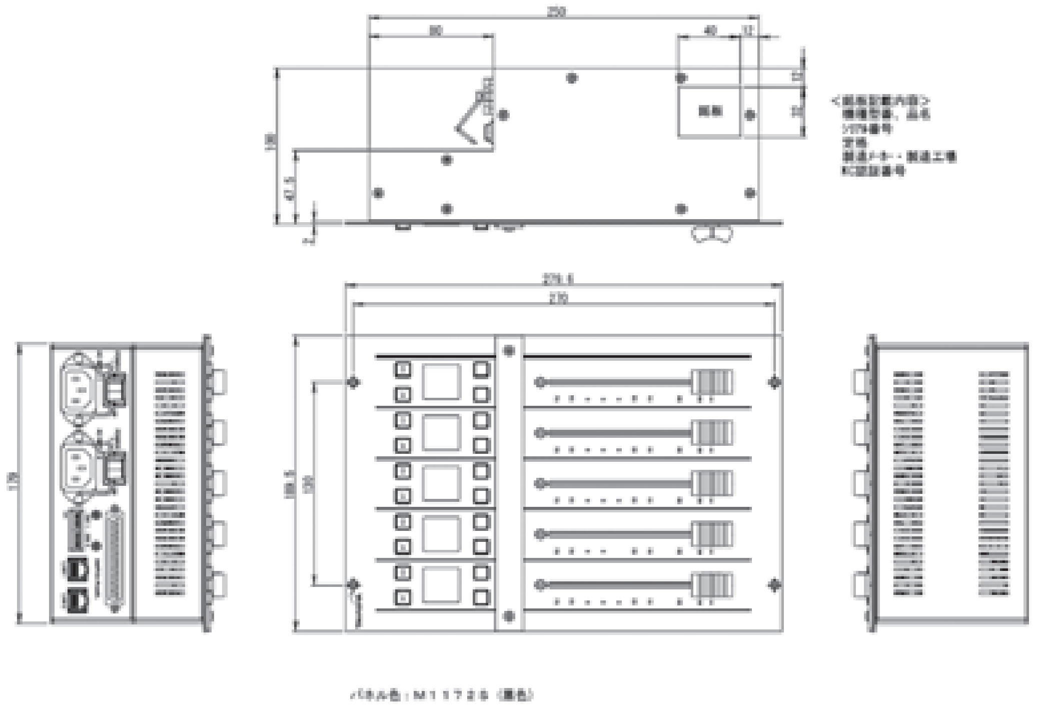
lluminated pushbutton switch. The operation and use of this button are defined in the connection destination device.

(2) CONTROL CONNECTOR

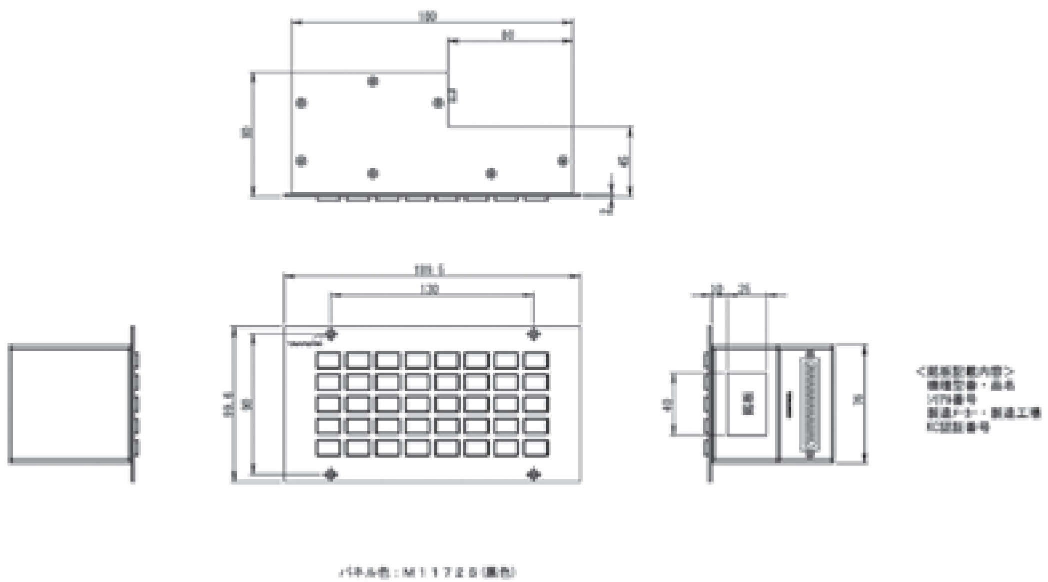
Connect to the device that controls this panel. Be sure to connect using the attached cable.

Dimensions

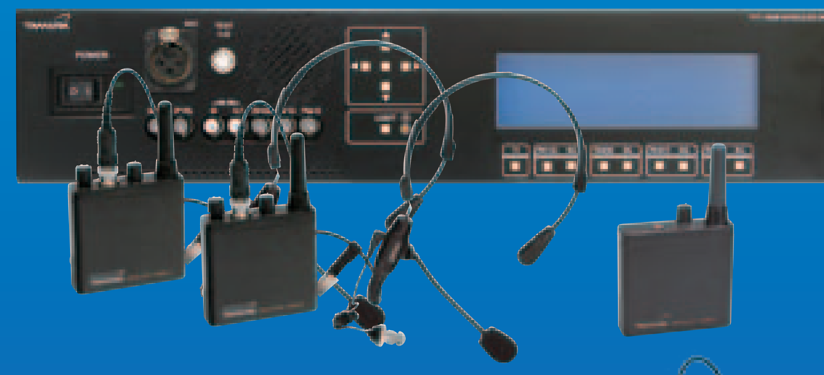
■ **REMOTE FADER Dimensions**



■ **SWITCH PANEL Dimensions**



INDEX



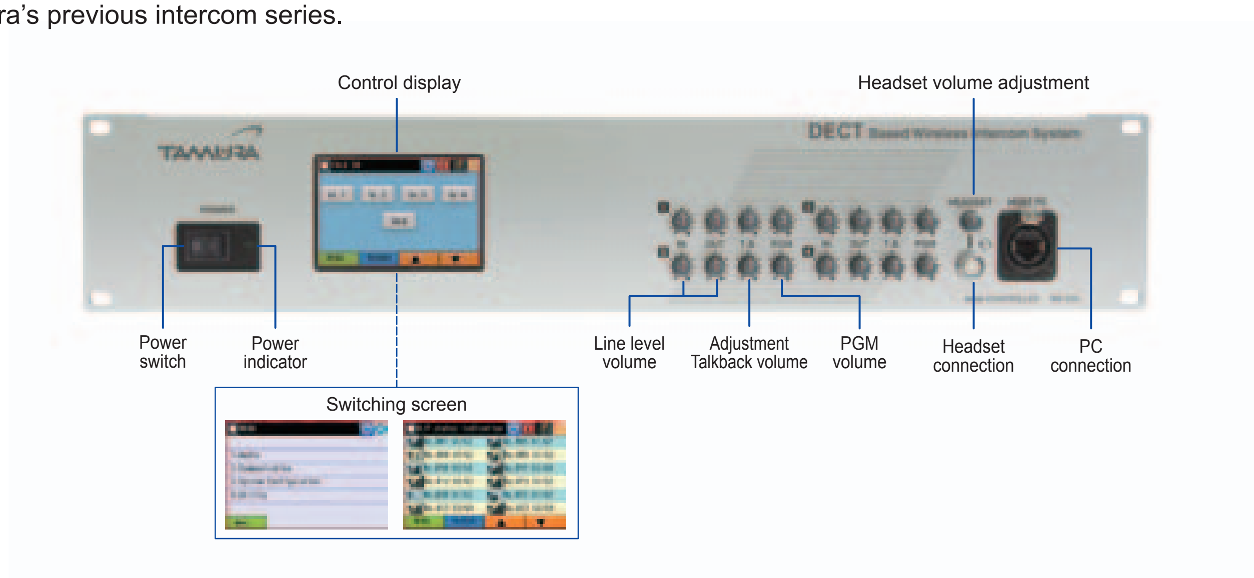
DECT Based Wireless intercom system

"DECT Based wireless intercom system" is Tamura Corporation's new digital wireless intercom system that inherits the intercom system technologies developed over time and complies with the new DECT format (ARIB STD- T101).

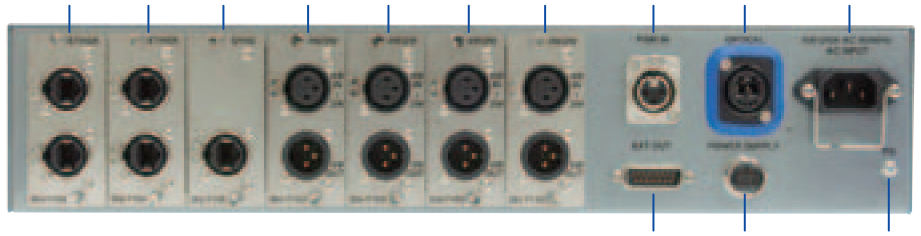
DECT Based wireless intercom system allows for improved convenience and large-scale system design, while maintaining the usability that enables intuitive operation.

DECT Based wireless intercom system is a communication tool that achieves both innovation and familiarity.

- The main controller allows for both touch-panel control and knob-based adjustments that inherit Tamura's previous intercom series.



- Audio lines can be organized into 4 groups per system. The main controller is equipped with 4W/2W external connections, which make linking with wired intercom systems as easy as before.



- Ethernet cable is used for the communication line between the antenna and the main controller. In addition, an optical cable is used to support longer distances, which can be extended to a maximum of 2.5 km.
  When a PoEHUB is used, Ethernet cable also allows for a star network configuration.

- 10 personal stations can be connected per antenna, and up to 60 personal stations can be connected per system for simultaneous calls only.

When combined with dedicated command-receiving devices, a total of 176 personal stations (48 personal stations for simultaneous calls and 128 personal stations for receiving commands) can be connected so as to enable large-scale system configuration.

Simultaneous calls only

Max. **60**
personal stations for simultaneous calls connected calls

Combined with dedicated command-receiving devices

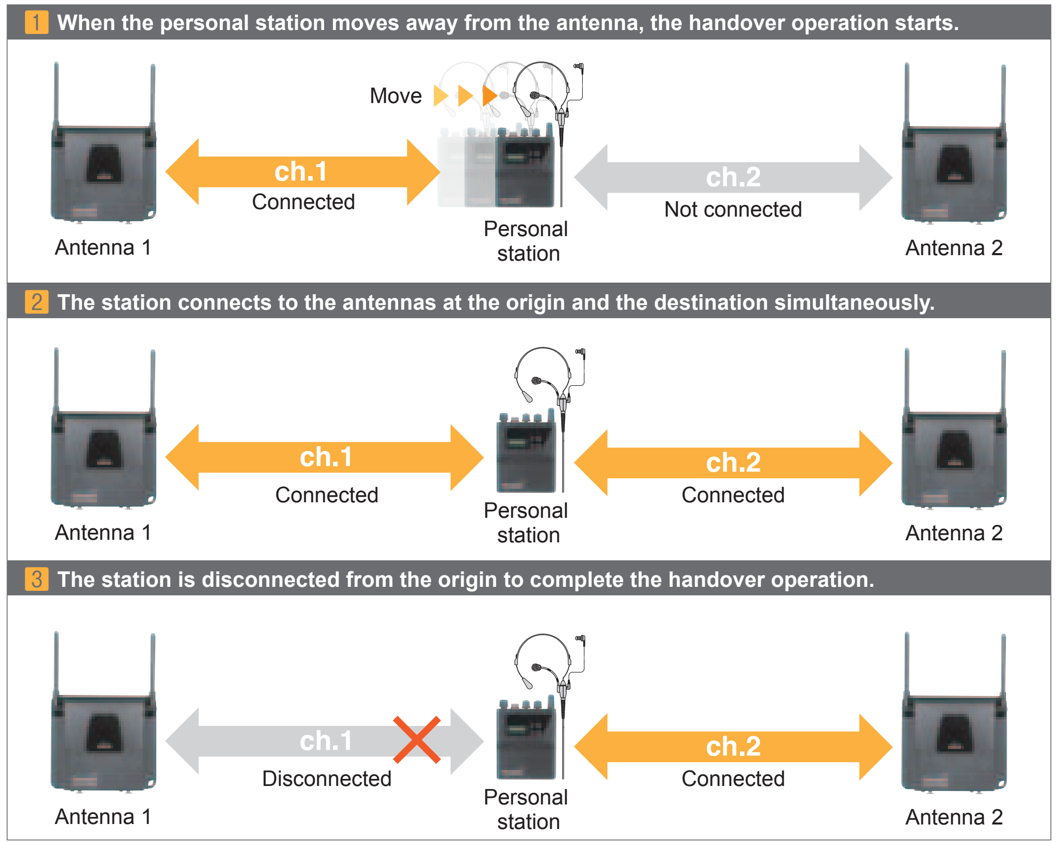
Max. **48**
personal stations for simultaneous calls connected

+

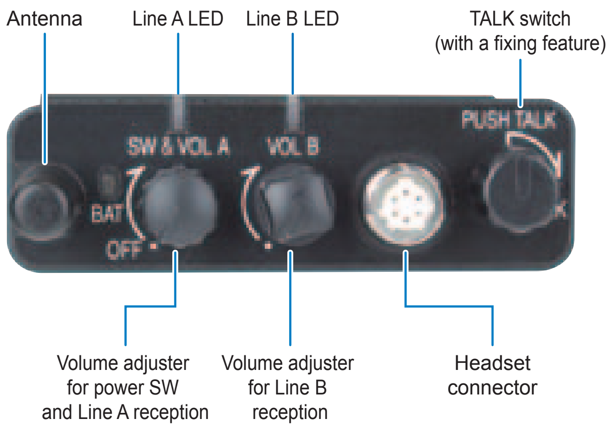
Max. **128**
personal stations for receiving commands connected

- Up to 16 antennas can be connected per main controller. When connecting 5 or more cell stations, connecting the main controller to the power supply can supply the power to all of the antennas.

- Seamless handover is adopted for the movement of personal stations between antennas. As the next antenna is detected beforehand, the seamless handover enables smooth transfer between antennas and seamless communication during station movements.



- Two audio groups can be assigned to the personal station to allow you to listen to the two groups simultaneously. Each volume adjuster is operated through knob control, which has been a popular feature of Tamura's previous intercom systems. It allows you to adjust the volume intuitively and respond instantly without the need to look at equipment when busy in the field.



- The system uses the 1.9-GHz band (DECT ARIB STD-T101), which does not overlap the Wi-Fi or other bands, reducing crosstalk. The system also reduces the crosstalk risk by channel transfer, when detecting any radio wave for an existing PHS or Tamura's previous digital wireless intercom systems.

Main Controller MK-C96

- Controls the entire system, when connected to an active antenna.
- Equipped with a line-specific volume adjustment feature for external interfaces.

External interface(4W/2W unit)	4 lines
Ethernet unit (for connecting to an active antenna)	4 lines
PGM input	1 line
Optical interface (for connecting to an active antenna)	1 line
Make-contact	4-line dry make-contact (Dsub15PIN)
Structure	Rack-mount type EIA=2U
Power supply	AC100V~240V
Power consumption	Approx. 40W
Environment	–10˚C~50˚C (excl. the display panel LCD)
Weight	7kg
Dimensions	H88×W480×D350 (mm)

Active Antenna MK-A96

- Communicates wirelessly with personal stations through control via the main controller.

No. of personal stations connected per antenna	When 10 personal stations are connected for simultaneous calls only and dedicated command-receiving devices are included: 8 command-receiving devices + 128 personal stations for simultaneous calls only
Structure	Wall-mounted and microphone stand-mounted
Power supply	Proprietary PoE or DC12V~24V
Power consumption	Approx. 9W
Environment	–10~50˚C
Weight	500g
Dimensions	H135xW153xD45 (mm) Excluding the dimensions of the protrusions

Power Supply MK-P96

- Used to supply power to active antennas. (Required when 5 or more active antennas are connected per main controller.)

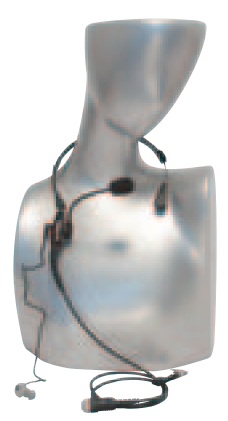
Output voltage	–55V
Power supply	AC100V~240V
Structure	Rack-mount type EIA=2U
Power consumption	Approx. 160W
Environment	–10~50˚C
Weight	6kg
Dimensions	H88×W480×D350 (mm)

Personal Station MK-B96

- Communicates wirelessly with active antennas.
- Supports the assignment of two audio groups and volume adjustment for each group.

Frequency characteristics	100Hz~7kHz
Power supply	AA alkaline battery x 2, or AA nickel metal hydride secondary battery x 2
Continuous use time	Approx. 8 hours (AA alkaline battery x 2) Approx. 12 hours (AA nickel metal hydride secondary battery x 2)
Environment	-10~50˚C
Weight	Approx. 218g (Contain an alkaline dry battery, Excluding leather cases)
Dimensions	H100×W85×D27 (mm) Excluding the dimensions of the protrusions

Headset MK-316C / HS-316C / HS-126D

		MK-316C (Condenser type)	HS-316C (Condenser type)	HS-126D (Dynamic type)
Appearance				
Microphone	Impedance	1.6kΩ	1.6kΩ	200Ω
	Sensitivity	– 73.0dB	–73.0dB	–86dB
	Frequency characteristics	100Hz~10kHz	100Hz ~ 10kHz	100Hz ~ 7kHz
Receiver	Impedance	16Ω	300Ω	8Ω
	Rated input	1mW	10mW	10mW
	Maximum permissible input	300mW	300mW	500mW
	Output sound pressure level	101.5dB	121dB	112dB
	Frequency characteristics	20Hz ~ 9kHz	100Hz ~ 3.5kHz	50Hz ~ 5kHz

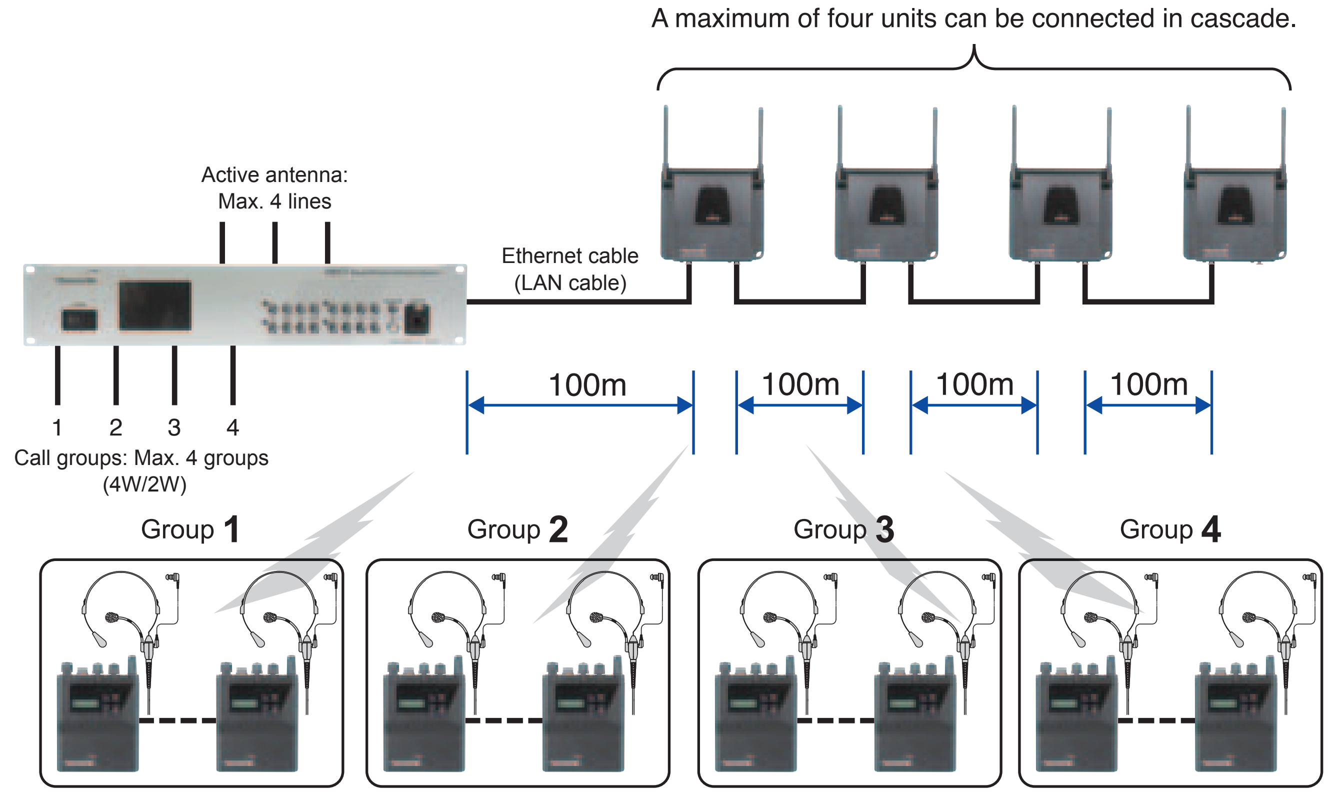
* HS-316C is exclusive for personal station.

Main System Specifications

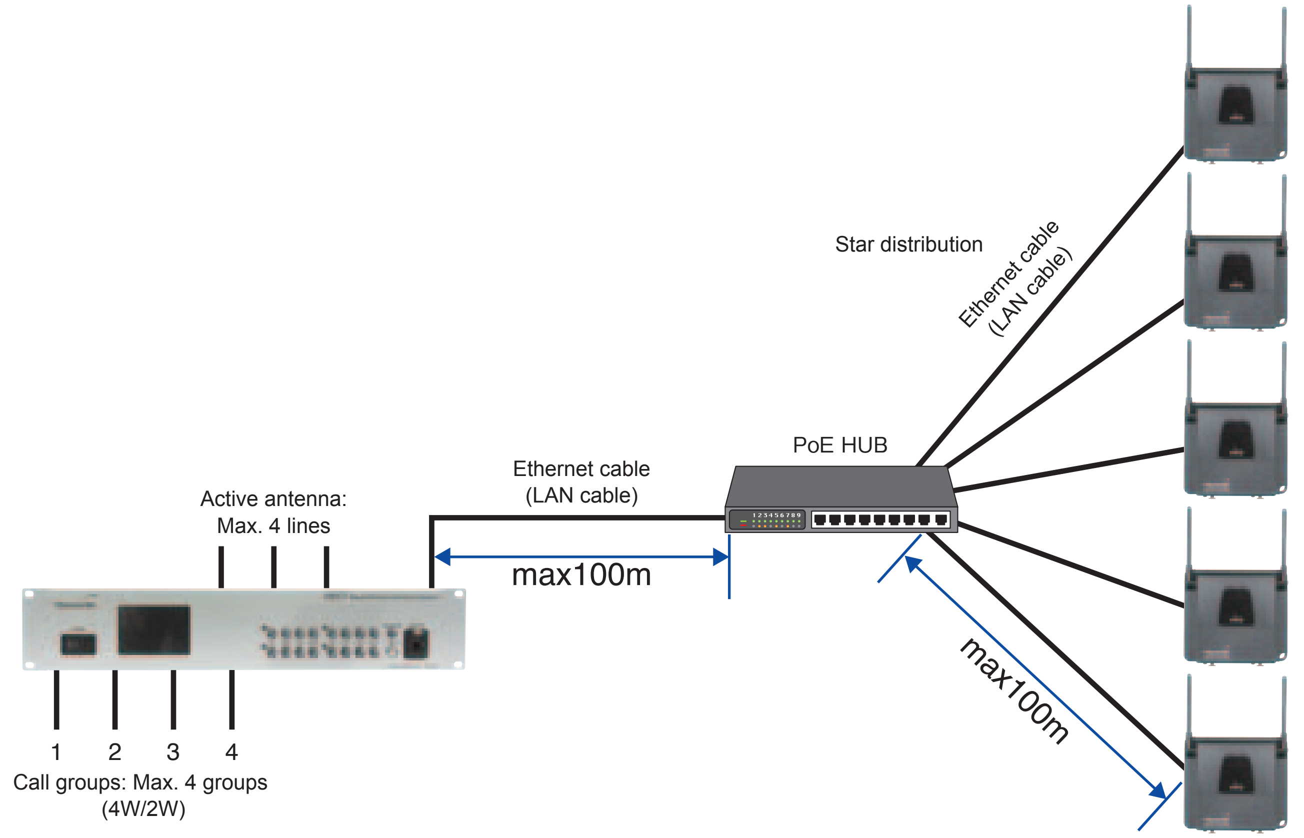
Item	Specification
Max. no. of stations connected per system	60 personal stations for calls or 48 personal stations for calls + 128 command-receiving devices
Max. no. of personal stations connected per active antenna	10
Max. no. of active antennas connected per system	16
No. of call groups per main controller	4
No. of personal station groups for simultaneous listening	2 (independent volume adjustments enabled)
Frequency characteristics	100Hz~7kHz
Radio system / Operating frequency	ARIB STD-T101 / 1.9GHz band
Personal station multipath support	Polarization diversity
Handover method	Seamless handover
Communication distance (line-of-sight)	Approx. 300m
Between the main controller and active antenna	Ethernet cable (max. 100m) or optical cable (max. 2.5km)
Continuous use time for personal station	Approx. 8 hours (AA alkaline battery x 2)

System Configuration Examples

Basic configuration

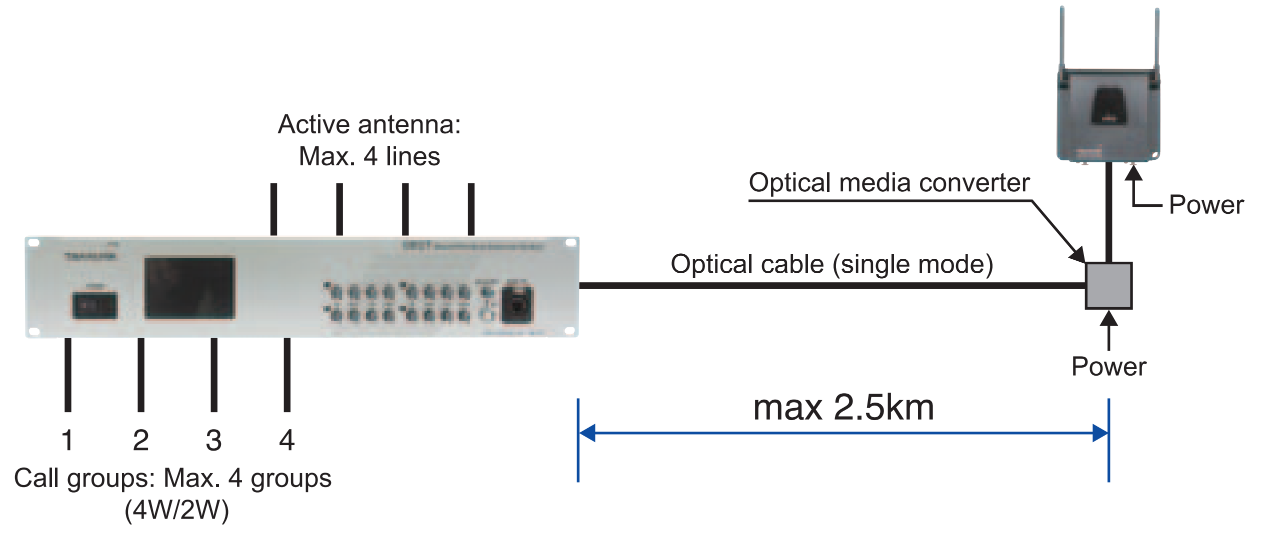


Configuration Example Using HUB

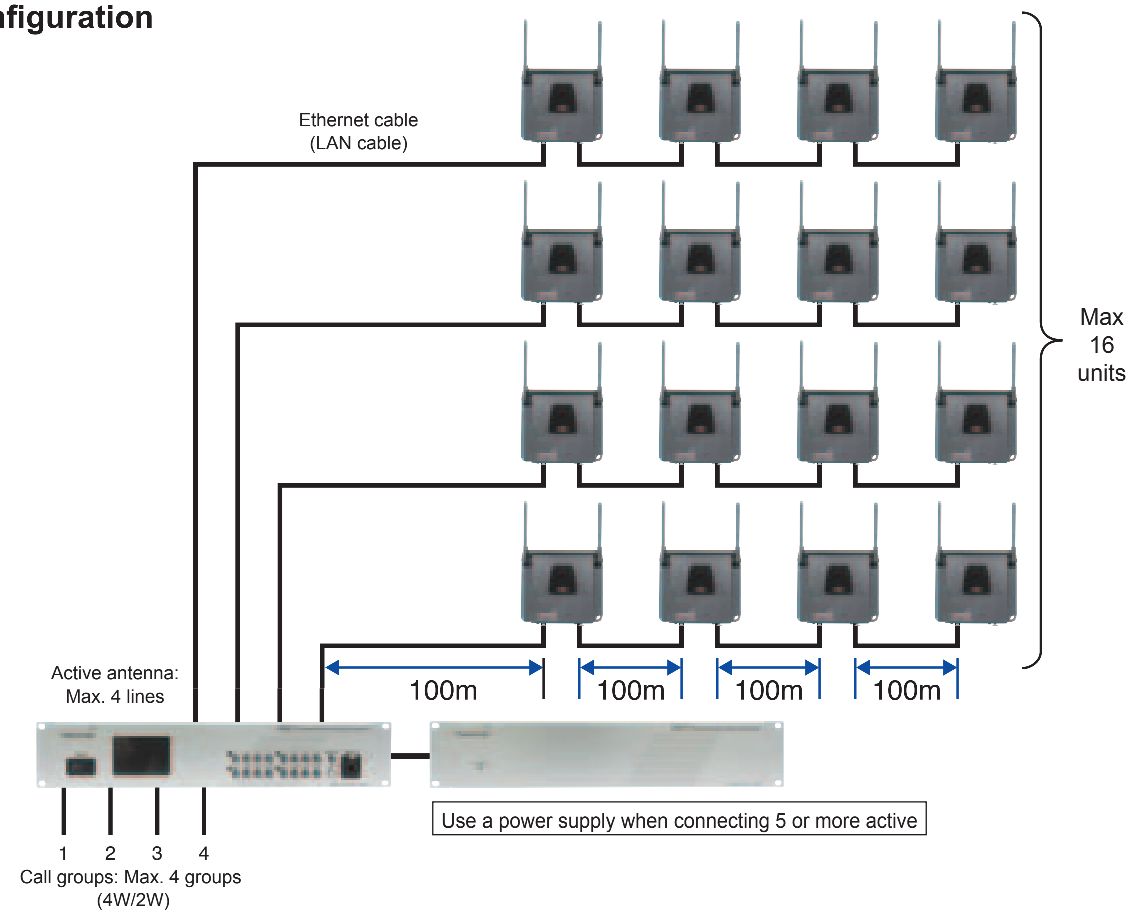


System Configuration Examples

When using optical cable



Max. Configuration



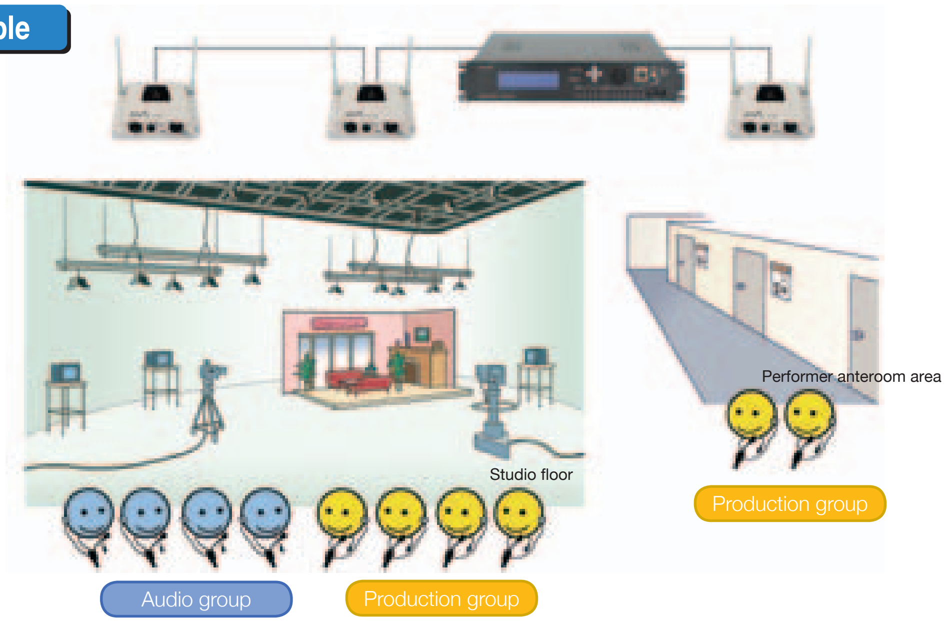
・PHS radio stations and different types of radio stations for digital cordless telephones operate in the operating frequency band of this equipment.
Due consideration has been given to this equipment so as to prevent radio wave interference with other radio stations that use the same frequency band.
However, should the equipment cause any harmful radio wave interference to another radio station, immediately stop emitting radio waves, and then contact our inquiry service desk, which is provided on the back of this catalog, to discuss how to prevent such crosstalk.
・All of the product screen images are inset composite images.

Digital Wireless Intercom System

Standard System

Leading the sector of simultaneous-call digital radio devices, Tamura's digital wireless intercom systems are used in a wide range of markets as highly reliable professional equipment.
Their simple operability and stable communication performance, which Tamura has always paid special attention to since the early development stages, allow for a wide range of applications not only in broadcasting stations, halls and theaters, but also for industrial use.

1 System example



2 System features

- ▶Radio station license is not required
- ▶Communication of higher quality than analog system
- ▶Quick connection
- ▶Use of optional CS control unit (for long distance) enables extended connection between master unit and slave unit up to 800 m (standard: 150 m)
  When the recommended cable [L-4E5C or DA206] is used

3 System standards

1) Used frequency: 1900 MHz band, 42 waves
2) Communication system: Multi-carrier TDMA-TDD system
3) Antenna power: 10 mW or less
4) Multiplicity: 4
5) Frequency switching: Synthesizer system by quartz control
6) Separation: 300 kHz (600 kHz separation in the same area)
7) Channel switching: MCA
8) Audio encoding system: 32 kbit/s ADPCM
9) Transmission rate: 384 kbit/s
10) Technical standard conformance: Conformity-certificated product
11) Radio station license: Not required

Base Station (BS) YFF-1870B

- Easily connected to wired intercom (2W/4W2 system)
- PGM input
- Independent use enabled using microphone/speaker
- Status display by LCD
- Up to 4 CS units can be connected
- Rack-mount type EIA=2U, JIS=2J

Specifications

Structure: Rack-mount type
Power supply: AC 100 V~240 V
Input/output: Microphone, SP, 2W/4W line, PGM, TEL
Environment: -10~+50℃ (excluding display panel (LCD) part)
Weight: Approx. 7.0 kg
Dimensions: Width: 480mm; height: 88mm; depth: 250mm
(not including protruding portions)

Cell Station (CS) YRW-1870B

- Diversity system
- Compact and easily installed temporarily
- Operable with one microphone cable
(power supplied from BS)

Specifications

Structure: Wall-mounting and microphone stand mounting system
Power supply: DC 24 V (supplied from the main device)
DC 12V (external power supply)
Number of calls: Simultaneous calls are possible in 1: 4
Antenna: Diversity operation with shared transmission/
reception and integrally structured case
Channel setting: Multi-channel access system
Standards: Technical standard conformance has been certificated
Environment: -10~+50℃
Weight: Approx. 430g (fittings included)
Dimensions: Width: 153mm; height: 135mm; depth: 45mm
(Excluding the dimensions of the protrusions)

Power UNIT YPL-1800A

Production on order

This product is necessary when five or more CS units are connected to one BS unit.

Specifications

Output voltage: 24V
Power supply: AC 100V
Environment: -10~+50˚C
Weight: Approx. 6.0kg
Dimensions: Width: 480mm; height: 88mm; depth: 350mm
(not including protruding portions)

Personal Station (PS) TWI-P190B

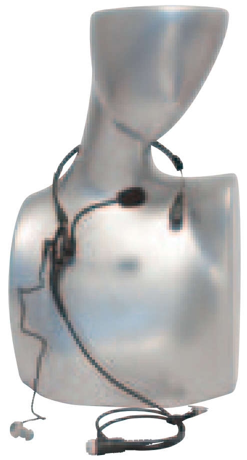
- Compact and Light focused on operability
- Operate with AA alkaline×2 or Nickel-hydrogen battery×2
- Big Volume Mode
*(Exclusive Headset is required Please ask us about details)
- Various kinds of setting information can be read and written using the Personal Station ID setting PC software.
- VOX Function (to Reduce Noise in Silence)
- Isolation Mode (with HS-316CTSW-002)
- CS by the main front of the switch, ID, can be set such as call group

Specifications

Structure: Compact, light, and Splash-proof IPX 4
Power supply: AA alkali cell×2 or Nickel-hydrogen battery×2
Continuous use time: 8 hours or more
Call: Bidirectional call
Antenna: Case-integrated (removal prohibited)
Gain 2.14 dBi or less
Channel setting: Multi-channel access system
Standards: Technical standard conformance has been certificated
Environment: -10~+50℃
Weight: Approx. 184g (batteries included)
Dimensions: Width: 67.7mm; height: 95.7mm; depth: 25.5mm
(Excluding the dimensions of the protrusions)

HEADSET

HS-316C



Specifications (HS-316C)

Microphone part (condenser type)
Impedance: 1.6kΩ±30%
Sensitivity: –73.0dB±4dB at 1kHz (0dB = 1V/0.5Pa)
Frequency characteristics: 100Hz~10kHz

Receiver part
Impedance: 300Ω (cord resistance included)
Rated input: 10mW
Maximum permissive input: 300mW
Output sound pressure level: 121dB at 1kHz (0dB=2×10^{-5} Pa)
Frequency characteristic: 100Hz~3.5kHz
*HS-316C is exclusive for personal station

HS-126D

Specifications (HS-126D)

Microphone part (dynamic type)
Impedance: 200Ω±20% at 1kHz
Inductance: 1.96mH±10%
DC resistance: 190Ω±10%
Sensitivity: –86dB±4dB at 1kHz (0dB = 1V/0.1Pa)
Frequency characteristics: 100Hz~7kHz –10dB

Receiver part
Impedance: 8Ω±15%
Inductance: 0.045mH±10%
DC resistance: 7.7Ω±10%
Maximum permissive input: 500mW
Output sound pressure level: 112dB±4dB at 1kHz (0dB=2x10-5 Pa)
Frequency characteristics: 50Hz~5kHz –20dB

Battery pack

BH-190

AA alkali cell×2

* Batteries are not included

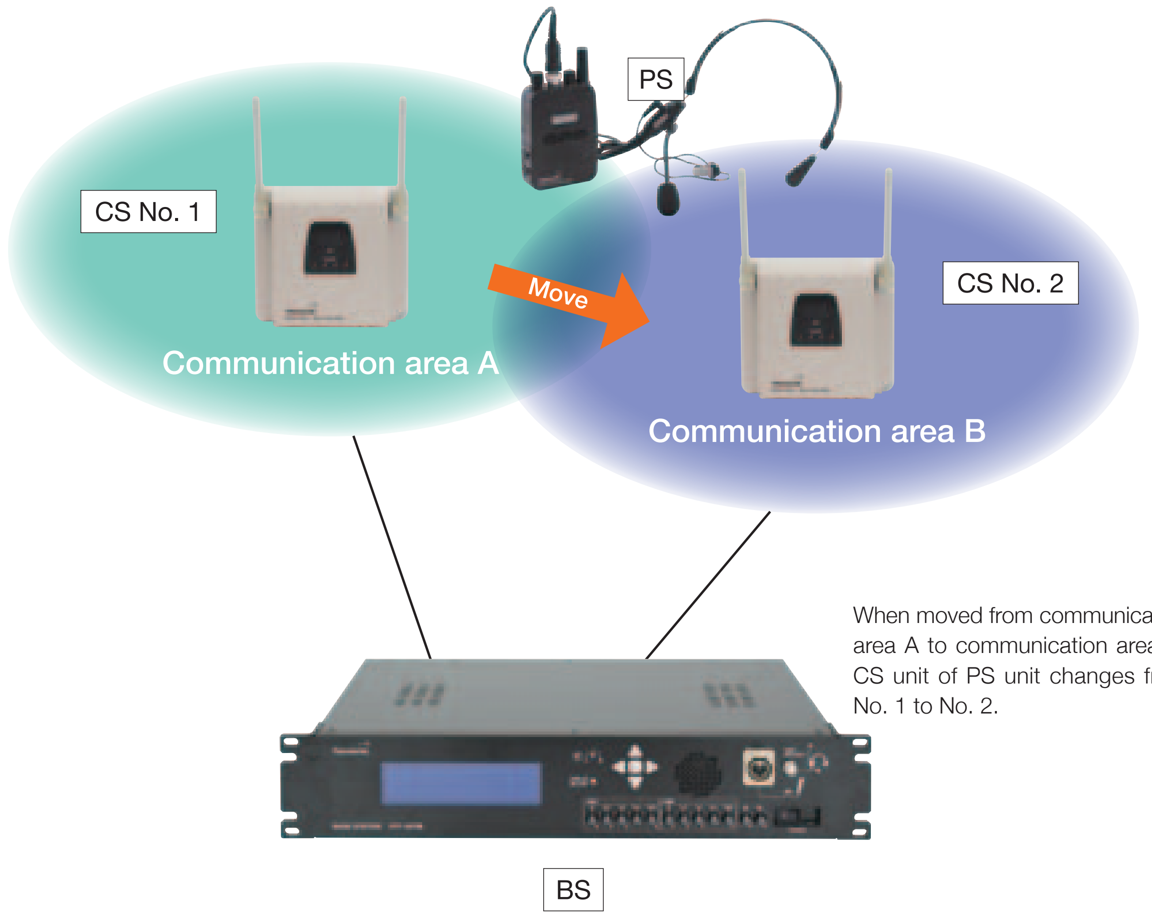
Outline of Digital Wireless Intercom System

1. One microphone cables connects between BS unit and CS unit, and between CS unit and CS unit
   Maximum 150 m between BS unit and CS unit, between CS units
   (recommended cable: CANARE L-4E5C or DA206)
2. Cascade connection up to 4 CS units is possible for one CS control unit in BS unit.
   Up to 4 CS control units can be mounted in one BS unit
   (When five or more CS units are connected to one BS unit, the Power UNIT [YPL-1800A] is necessary)
3. Up to 4 PS units can make a call to one CS unit.
4. PS units can be divided into 2 groups for use with one BS unit
5. No limit for use to the number of PS units dedicated for receiving command
   (When a PS unit dedicated for receiving command is used, the number of PS units that can be used is reduced by one)

Handover operation

When the extension of communication area is desired, handover setting is available by the setting of PS unit.

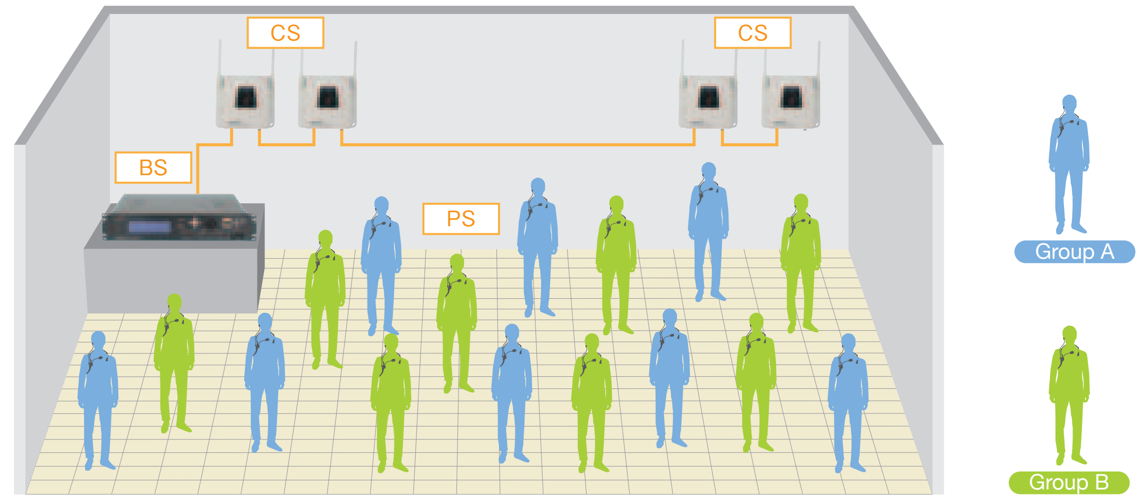
Handover: The CS units to which a PS unit is connected are switched automatically



When moved from communication area A to communication area B, CS unit of PS unit changes from No. 1 to No. 2.

System example -1

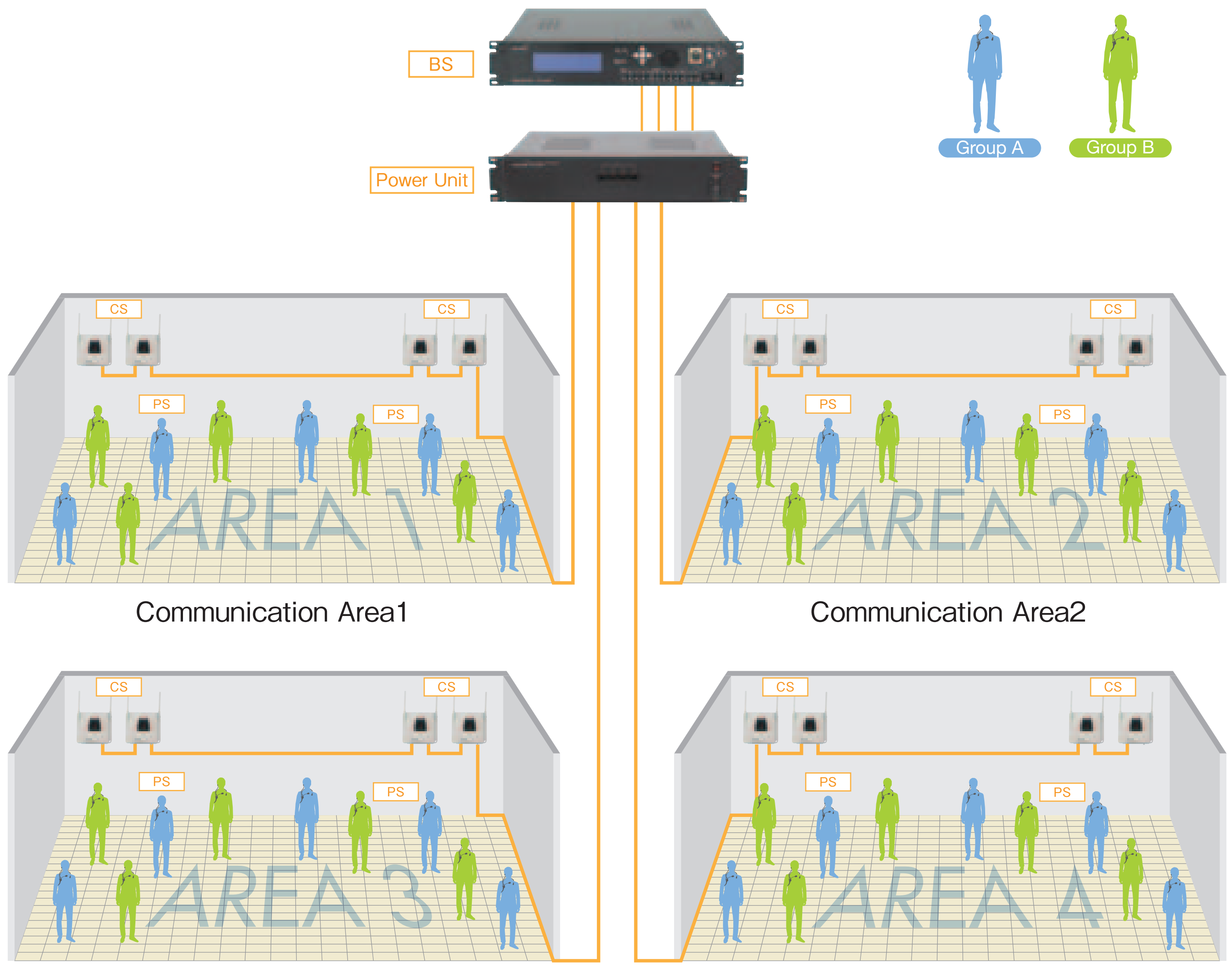
BS: One unit | CS: 4 units | PS: 16 units



Communication Area

System example -2

BS: One unit | Power Unit: One unit | CS: 16 units | PS: 40 units



Electrical characteristics

<table>
<tr><th colspan="2"></th><th>BS YF F-1870B
CS YRW-1870B</th><th>PS TWI-P190B</th></tr>
<tr><td rowspan="9">Common to high frequencies</td><td>Radio wave type</td><td colspan="2">G7D, G7E, G7X, G1D, G1E, G1X</td></tr>
<tr><td>Antenna type</td><td>λ/2 sleeve antenna</td><td>Whip antenna</td></tr>
<tr><td>Antenna impedance</td><td colspan="2">50Ω</td></tr>
<tr><td>Frequency range</td><td colspan="2">1895.15 ~ 1905.95MHz</td></tr>
<tr><td>Number of frequencies</td><td colspan="2">42 waves (control carrier 2 waves, communication carrier 40 waves)</td></tr>
<tr><td>Separation</td><td colspan="2">300kHz</td></tr>
<tr><td>Oscillation system</td><td colspan="2">Quartz control frequency synthesizer system</td></tr>
<tr><td>Frequency stability</td><td colspan="2">Within $\pm 3\times10^{-6}$</td></tr>
<tr><td>Modulation accuracy</td><td colspan="2">12.5% or less</td></tr>
<tr><td rowspan="6">Transmission</td><td>Antenna power</td><td colspan="2">10mW</td></tr>
<tr><td>Intensity of spurious radiation</td><td colspan="2">2.5μW or less (beyond band) 250nW or less (within band)</td></tr>
<tr><td>Modulation system</td><td colspan="2">$\pi/4$ shift QPSK</td></tr>
<tr><td>Audio frequency</td><td colspan="2">3.4kHz or less</td></tr>
<tr><td>Neighboring channel leak power</td><td colspan="2">600kHz mistuned 800nW or less,
900kHz mistuned 250nW or less</td></tr>
<tr><td>Occupied frequency band area</td><td colspan="2">Within 288 kHz</td></tr>
<tr><td rowspan="5">Reception</td><td>Reception system</td><td colspan="2">Double superhetrodyne</td></tr>
<tr><td>Reception sensitivity</td><td colspan="2">16 dBμV or less (bit error rate 1×10^{-2})</td></tr>
<tr><td>Spurious sensitivity</td><td colspan="2">47 dB or more</td></tr>
<tr><td>Neighboring channel selectivity</td><td colspan="2">50 dB or more (600 kHz detuning)</td></tr>
<tr><td>Body radiation</td><td colspan="2">4nW or less</td></tr>
<tr><td rowspan="7">Common</td><td>Line frequency characteristic</td><td colspan="2">3.4kHz or less</td></tr>
<tr><td>Line input/output</td><td>0dBm balanced</td><td>–</td></tr>
<tr><td>Microphone input</td><td>-60dBm balanced</td><td>-60dBm unbalanced</td></tr>
<tr><td>Speaker output</td><td>Inside 1W Outside 2W at 8Ω</td><td>–</td></tr>
<tr><td>External input</td><td>0dBm balanced</td><td>–</td></tr>
<tr><td>Used power supply/
power consumption</td><td>AC100V±15%: 3A
AC240V±15%
DC 12~24V: 1A (cell station only)</td><td>130 mA or less at DC 3V</td></tr>
<tr><td>Use environment</td><td colspan="2">Temperature; -10 ~ +50˚C, Humidity: Within 30~90%</td></tr>
</table>

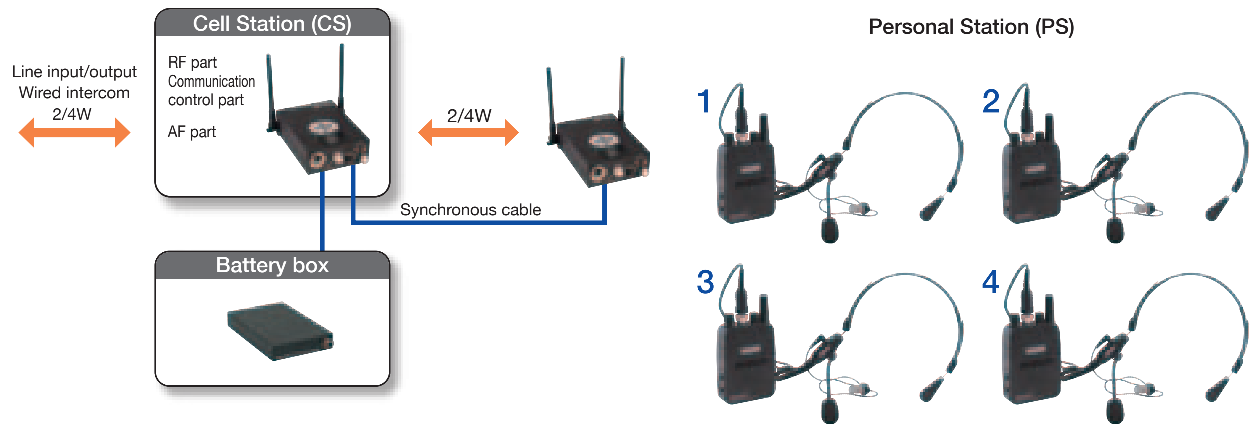
Digital Wireless Intercom System

Portable type System

This portable-type digital wireless intercom system is packed with high-functionality and high-performance features.
Since functions are integrated between the cell station and the main equipment, two party-line systems can be built on the single cell station.
As this product is battery-operated, it can also be used outdoors, where there is no power supply.

Basic system

Wired Intercom System, Connection with 2 groups is possible!



Cell Station (CS) YFP-1821B

Features

2W line of Clear-Com can be input directly
4 PS units can be connected to one CS unit.
Up to 5 CS units can be used simultaneously with synchronous connection
Continuous use time is 7 hours or more (when 8 AA alkali cells are used)
The following settings are available with a cross key on the front

- Setting of LINE input/output interface (4W/2W)
- Setting of talkback
- Setting of audio input/output level
- Confirmation of registration ID
- Setting of RF output level

Specifications

Audio frequency: 3.4 kHz or less
Audio encoding system: 32 kbit/s ADPCM
Line specification: 4W: IN 0 dBm, OUT 0 dBm
AIR: IN -20dBm
Microphone input: -50dBm (unbalanced 600Ω)
Speaker output: 15 mW or more (at 8Ω)
Power supply: DC8.0~16.0V (negatively grounded)
Structure: Portable type
Standards: Technical standard conformance has been certificated
Environment: -10~+50℃
Weight: Approx. 430g
Dimensions: Width: 125mm; height: 125mm; depth: 37mm
(including antenna)

Battery box MK-D96

YFP-1821B Battery box
8 AA alkali cells are used
Weight: Approx. 160g
(Cable, batterys are not including)
Dimensions: Width: 89mm; height: 142mm; depth: 22mm
(not including protruding portions)

Analog Wireless Intercom System

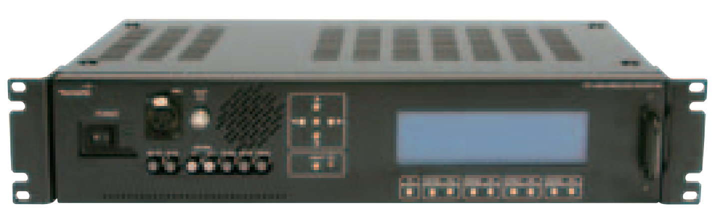
Antenna power 1mW or less, for surface movement business Wireless Intercom System

This intercom system is equipped with simple operability and basic performance as wireless equipment.
Using an antenna distribution method, the system can cover dead zones of radio waves with multiple antennas.

1 : 4 Simultaneous Communication System

Land mobile station

Base Station YFF- 4530



Specifications

Structure: Rack mounting type
Power supply: AC 100 V
Number of calls: 1:8 simultaneous calls
Circuit configuration: Unit structure
Number of antennas: 2 (transmission/reception shared)
Channel setting: Station selection is easy with quartz control PLL synthesizer system
Standards: Technical standard conformance has been certificated
Environment: -10~+50℃
Weight: Approx. 7.0 kg
Dimensions: Width: 480mm; height: 88mm; depth: 250mm
(not including protruding portions)

Personal Station YMT- 4120

Specifications

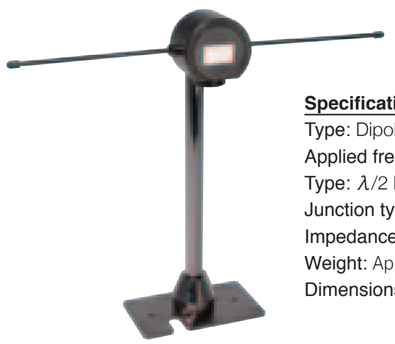
Structure: Compact, light, and drip-proof
Power supply: AA alkali cell×2 Continuous use time is 20 hours
Call: Interactive simultaneous call
Antenna: Helical antenna or whip antenna for transmission/reception
Channel setting: Station selection is easy by quartz control PLL synthesize system
Environment: 10~+50℃
Weight: Approx. 220g (battery pack YBA-4120 included)
Dimensions: Width: 85.4mm; height: 82mm; depth: 22.5mm
(not including protruding portions)

Command receiving device YRT- 4120

Specifications

Structure: Compact, light, and drip-proof
Power supply: AA alkali cell×2 Continuous use time is 23 hours
Call: Interactive simultaneous call
Antenna: Helical antenna or whip antenna for transmission/reception
Channel setting: Station selection is easy by quartz control PLL synthesize system
Standards: Technical standard conformance has been certificated
Environment: –10~+50℃
Weight: Approx. 210g (battery pack YBA-4120 included)
Dimensions: Width: 85.4mm; height: 82mm; depth: 22.5mm
(not including protruding portions)

Antenna CAW-4510



Specifications

Type: Dipole type
Applied frequency: 413~454MHz
Type: λ/2 half wavelength type
Junction type: M type
Impedance: 75Ω
Weight: Approx.800g (attachment base included)
Dimensions: Width: 325mm; height: 287.5mm; depth: 91.5mm

Battery pack

YBA-4120

Production on order

AA alkali cell×2

* Batteries are not included

PBA-4120

Production on order

Nickel-hydrogen battery (2.4 V)

Electrical characteristics

		Base Station YFF-4530	Personal Station YMT-4120	Command receiving device YRT-4120
Frequency common	Radio wave type	F3E/F2D (however, use is inhibited only with F2D)		–
	Antenna type	Half wavelength dipole antenna	Helical antenna or whip antenna	
	Antenna impedance	75Ω	50Ω	
	Frequency rand	Transmission: 454MHz band, Reception: 413MHz band	Transmission: 413MHz band, Reception: 454MHz band	Reception: 454MHz band
	Number of frequencies	Downward (master unit transmission): 24 waves, upward (slave unit) 72 waves		
	Separation	12.5 kHz (Interleave 6.25 kHz)		
	Oscillation system	Quartz control PLL synthesizer system		
	Frequency stability	With in ±4ppm		
	Compander characteristic	Transmission compressor 2: 1, Reception expander 1: 2		
Transmission	Antenna power	With in 1mW (+20 -50%) Total of antenna terminal		–
	Strength intensity of spurious radiation	2.5 μW or less		–
	Modulation scheme	Direct frequency modulation		–
	Voice frequency	3 kHz or less(300Hz~3kHz)		–
	Neighboring channel leak power	60 dB more than carrier wave power		–
	Occupied frequency Bandwidth	With in 8.5 kHz		–
	Frequency deviation	±2.5kHz less than when carrier frequency without modulation		–
Reception	Reception system	Double superheterodyne		
	Reception sensitivity	0 dB μV or less at SINAD 12 dB		
	Squelch sensitivity	Tone SQ: 0 dBμV or less, Noise SQ: 0 dBμV or less		
	Cabinet radiation	4nW or less		
Common	Audio frequency characteristic	Within 300 Hz~3kHz		
	Line input/output	0 dBm, balanced 600Ω (4W) -20 dBu, unbalanced 220Ω (2W)	–	
	Microphone input	-60 dBm, balanced 600Ω	−60 dBm, unbalanced 600Ω	–
	Speaker output	Inside; 1W(8Ω), Outside: 1W(8Ω)	15 mW or more (at 8Ω)	
	Program input	-20~+10 dBm unbalanced 600Ω	–	
	Power supply use range	AC100V ± 15%: 1.5A	At 3.0 V: 150 mA or less	At 3.0 V: 120 mA or less
			Warning lamp flashes at 2.3 V or less	
	Use environment	Temperature: -10~+50˚C, Humidity: with in 35~90%		

Frequency within license

[BS]
Downward transmission frequency

Channel No.	Frequency (MHz)	Channel No.	Frequency (MHz)
1	454.05000	2	454.05625
3	454.06250	4	454.06875
5	454.07500	6	454.08123
7	454.08750	8	454.09375
9	454.10000	10	454.10625
11	454.11250	12	454.11875
13	454.12500	14	454.13125
15	454.13750	16	454.14375
17	454.15000	18	454.15625
19	454.16250	20	454.16875
21	454.17500	22	454.18125
23	454.18750	24	454.19375

[PS]
Upward transmission frequency

Channel No.	Frequency (MHz)	Channel No.	Frequency (MHz)
1	413.70000	2	413.70625
3	413.71250	4	413.71875
5	413.72500	6	413.73125
7	413.73750	8	413.74375
9	413.75000	10	413.75625
11	413.76250	12	413.76875
13	413.77500	14	413.78125
15	413.78750	16	413.79375
17	413.80000	18	413.80625
19	413.81250	20	413.81875
21	413.82500	22	413.83125
23	413.83750	24	413.84375

Channel No.	Frequency (MHz)	Channel No.	Frequency (MHz)
25	413.85000	26	413.85625
27	413.86250	28	413.86875
29	413.87500	30	413.88125
31	413.88750	32	413.89375
33	413.90000	34	413.90625
35	413.91250	36	413.91875
37	413.92500	38	413.93125
39	413.93750	40	413.94375
41	413.95000	42	413.95625
43	413.96250	44	413.96875
45	413.97500	46	413.98125
47	413.98750	48	413.99375

Channel No.	Frequency (MHz)	Channel No.	Frequency (MHz)
49	414.00000	50	414.00625
51	414.01250	52	414.01875
53	414.02500	54	414.03125
55	414.03750	56	414.04375
57	414.05000	58	414.05625
59	414.06250	60	414.08875
61	414.07500	62	414.08125
63	414.08750	64	414.09375
65	414.10000	66	414.10625
67	414.11250	68	414.11875
69	414.12500	70	414.13125
71	414.13750	72	414.14375

A-type 1.2 GHz band OFDM digital wireless microphone system

Main features

High sound quality
Audio transmission mode Uncompressed 24-bit/48 kHz and ADPCM

Superior radio wave propagation
Maximum ratio combining diversity

Low latency
1 ms or less

Remote terminal function
Monitoring of transmitter settings with an operating terminal

PC management function
Monitoring of the transmitter state, logger function

Large-capacity rechargeable battery
Operable for 6 consecutive hours

Transmitter TWO-H120A / TWO-T120

- Using a replaceable handheld microphone capsule
- The two-piece type uses an internal transmitting antenna, offering low-loss, superior radio wave propagation characteristics
- Always operable remote terminal functions
- Using a large-capacity rechargeable battery for long-time continuous operation (6 hours)

TWO-T120
Digital wireless microphone (two-piece type)

TWO-H120A
Digital wireless microphone (handheld type)

		TWO-H120A (handheld type)	TWO-T120 (two-piece type)
RF	Operating frequency band	1240.325 MHz to 1259.675 MHz (excluding 1251.700 MHz to 1253.300 MHz) Maximum number of channels used: 23CH 800 kHz step, 32 groups	
	Antenna power	20 mW/10 mW/2 mW	
	Occupied bandwidth	600 kHz	
	Transmission system	OFDM	
Audio	Frequency characteristics	20 Hz to 22 kHz	
	Total harmonic distortion ratio	0.01% or less	
	Dynamic range	120 dB (A-weighted)	
	Sampling frequency	48 kHzv	
	Low-cut frequency	60/80/100/125 Hz, 12 dB/oct	
	Level setting	Gain (3 dB step) -21 dB to +21 dB	Sensitivity (1dB step) LINE setting: +4 dB to -16 dB MIC setting: -24 dB to -75 dB
	Information compression	Linear PCM/ADPCM	
Indicators	Display	LCD	
	LED	Power supply state display Audio level display	
Power supply	Battery	Dedicated battery (lithium rechargeable battery) and AA cell battery (×2) (optional)	Dedicated battery (lithium rechargeable battery)
	Operating time (10 mW output, 25°C)	(When the dedicated battery is used) 6 hours or more (When AA cell batteries are used) 1 hour or more	(When the dedicated battery is used) 6 hours or more
General	Dimensions	φ37mm, height 185 mm (not including antenna)	Width 68.4 mm, height 110 mm, depth 22.8 mm (not including protruding portions)
	Weight	223g (Not including microphone capsule / Including battery pack)	160g (Including battery pack)
	Operating temperature/humidity	0°C to +40°C, 20% to 90% (No condensation)	
	Standard	Compliant with ARIB STD-T112	

Receiver — TWO-R120

TWO-R120

Digital wireless receiver
(2-ch implementation, 1U type)

- Design that uses the 200 MHz band for antenna input to reduce coaxial cable losses
- Digital audio output with a built-in SRC, compatible with 48 kHz/96 kHz external synchronization and output
- Incorporating a LAN port for external remote control
- PC applications for the receiver are available

RF	Operating frequency band	1240.325 MHz to 1259.675 MHz (excluding 1251.700 MHz to 1253.300MHz) Maximum number of channels used: 23CH 800 kHz step, 32 groups
	Reception sensitivity	-85 dBm or less
Audio	Frequency bandwidth	20 Hz to 22 kHz
	Total harmonic distortion ratio	0.01% or less
	Dynamic range	120 dB (A-weighted)
	Analog output	Balance (600Ω load)
	Analog output level	LINE output: -15 dBu (reference level), +21 dBu (maximum level) MIC output: -51 dBu (reference level), -15 dBu (maximum level)
	Sampling frequency	48 kHz
	Number of quantization bits	24 bits
	Headphone output	Unbalance (8~64Ω load)
	Headphone output power	50 mW or more (35Ω load)
	Digital output	AES/EBU (AES3id-compliant) Sampling frequency: 48 kHz/96 kHz SRC: ON/OFF
	Digital output reference level	-36 dBFS to -18 dBFS (2 dB step)
	Digital audio external synchronization signal	WORD CLOCK 48 kHz or 96 kHz

Indicators	Display	VFD
	LED	Receive status display, audio level display, ALARM display, WORD synchronization display, LOCAL signal display, headphone output selection state display
Interface	Antenna input connector (ANT A/B)	BNC-J (50Ω)×2 (DC 12V superimposition)
	Antenna output connector (ANT A/B)	BNC-J (50Ω)×2
	Analog audio out put connector	XLR-3-32 equivalent×2
	Digital audio output connector	BNC-J (75Ω)×2
	Digital audio synchronization signal I/O connector:	Input: BNC-J (75Ω)×1 Output (through): BNC-J (75Ω)×1
	Headphone output jack	φ6.3 stereo phone jack×1
	LAN connector	RJ-45 modular jack×1
General	Power supply	AC100V, 50Hz / 60Hz
	Dimensions	Width: 480 mm; height: 43.7 mm; depth: 430 mm (not including the rubber feet protruding portions)
	Weight	7kg
	Operating temperature/humidity	-10°C to +50°C, 20% to 90% (No condensation)
	Standard	Compliant with ARIB STD-T112

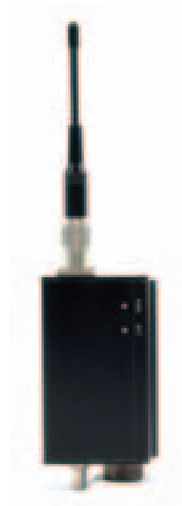
Antenna — TWO-A120 / TWO-AY120

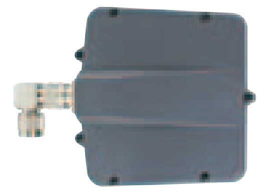
TWO-A120

Antenna with a built-in down converter
(omnidirectional)

TWO-AY120

Antenna (directional)





- Offering a transmission distance of up to 200 m (When using 5D-FB), thanks to a down converter system
- Capable of expanding the reception area by mixing antenna outputs with the mixing/distributing device
- A sleeve ground omnidirectional antenna and a 120-degree wide-angle small directional antenna are available

RF	Input frequency	1240 MHz to 1260 MHz
	Output frequency	198 MHz to 219 MHz
	Attenuator	0 dB / 20 dB
	Antenna input terminal	N-J (50 Ω) × 1
	Antenna output terminal	BNC-J(50 Ω) × 1
General	Power supply	DC 12 V (supplied by the receiver or the mixing/distributing device)
	Dimensions	Width: 36 mm; height: 117.1 mm; depth: 72 mm (not including protruding portions)
	Weight	413g (not including antenna)
	Operating temperature/humidity	-10°C to +50°C, 20% to 90% (No condensation)
	Standard	Compliant with ARIB STD-T112

Mixing/distributing device — TWO-D120

TWO-D120

Antenna mixing/distributing device

- Capable of expanding the reception area by mixing antenna outputs with the mixing/distributing device

RF	Passing frequency	198 to 219 MHz
	Passing gain	0 dB
Interface	Antenna input connector (ANT A/B)	BNC-J (50 Ω) × 4 (DC 12 V, local 45 MHz superimposition)
	Antenna output connector (ANT A/B)	BNC-J (50 Ω) × 8
	Local input	BNC × 1 (50 Ω)
	Local output	BNC × 1 (50 Ω)
General	Power supply	AC 100 V, 50 Hz/60 Hz
	Dimensions	Width: 480 mm; height: 43.7 mm; depth: 430 mm; (not including the rubber feet protruding portions)
	Weight	4.5kg
	Operating temperature/humidity	-15°C to +50°C, 20% to 90% (No condensation)
	Standard	Compliant with ARIB STD-T112

Charger — TWO-BC120

TWO-BC120

Charger

Display		• Power supply LED × 1 • Charging LED × 6 • LCD × 1
Functions	Normal charging	Charging the rechargeable battery pack
	Recovery charging	Restoring and charging a rechargeable battery pack that is in an over-discharged state
	Storage mode	Adjusting the remaining power of a rechargeable battery pack to a level appropriate for long-time storage
	Rechargeable battery monitoring	• Voltage- Remaining power • Temperature • Cell lifetime • Charging time
Power supply		DC +12 V Supplied by an AC adapter
Dimensions		Width: 410 mm; height: 110 mm; depth: 249.3 mm (not including protruding portions)
Weight		3.1kg
Operating temperature/humidity		-10°C to 40°C, 20% to 90% (No condensation)

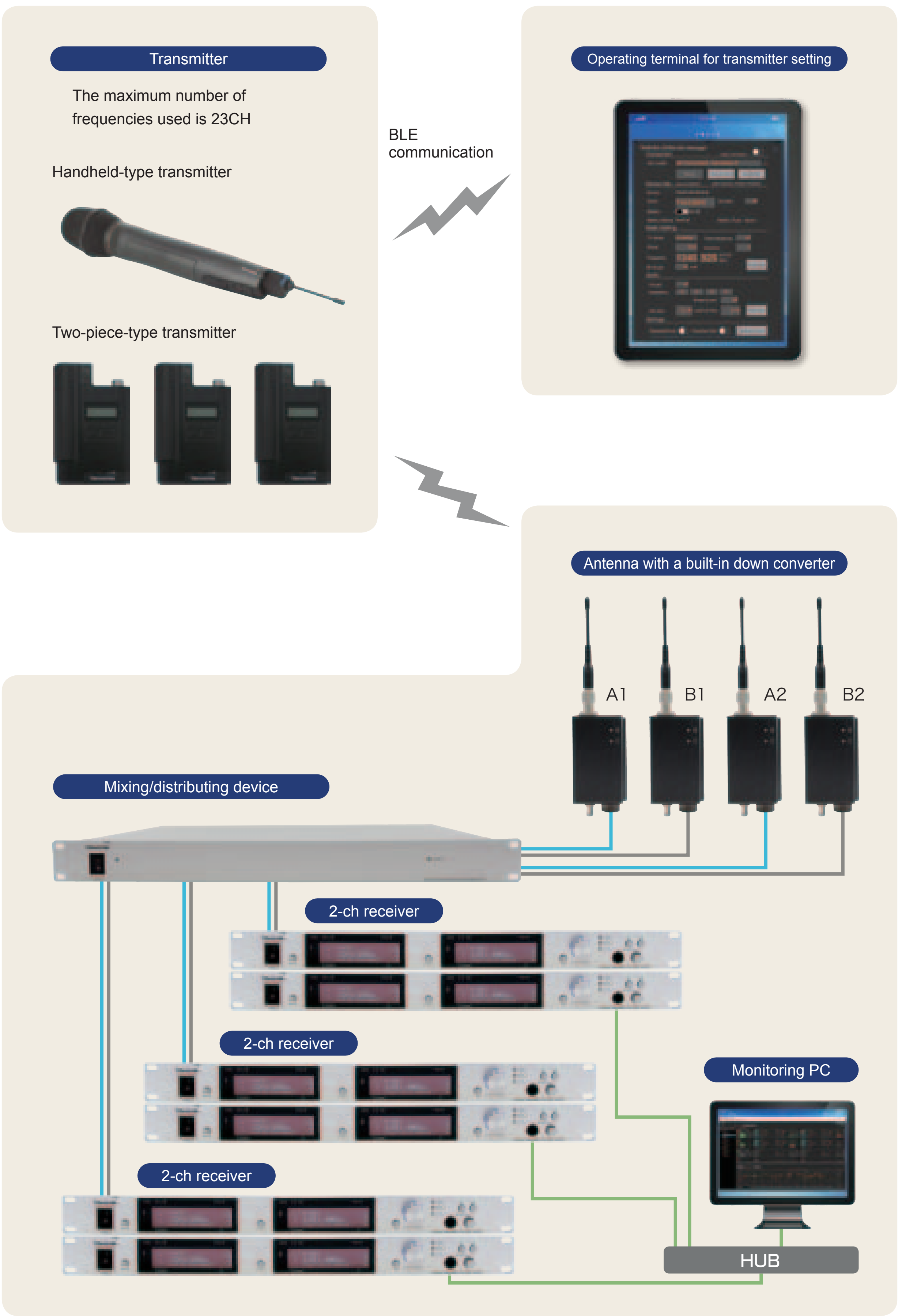
Remote repeater — TWO-RM120

TWO-RM120

Remote repeater

Bluetooth	Communication distance	10 m
	Maximum number of slaves	8
	Standard	Bluetooth v4.0 (Bluetooth Low Energy)
Interface	LAN connector	RJ-45 modular jack × 1
Display switch	LED	LINK/ACT, SPEED
General	Power supply	DC-48V (supplied by PoE)
	Dimensions	Width: 82 mm; height: 120.5 mm; depth: 22.9 mm (not including protruding portions)
	Weight	120g
	Operating temperature /humidity	-10 to +50°C, 20 to 90% (No condensation)

Example configuration of a general-purpose system



Wireless Monitoring System

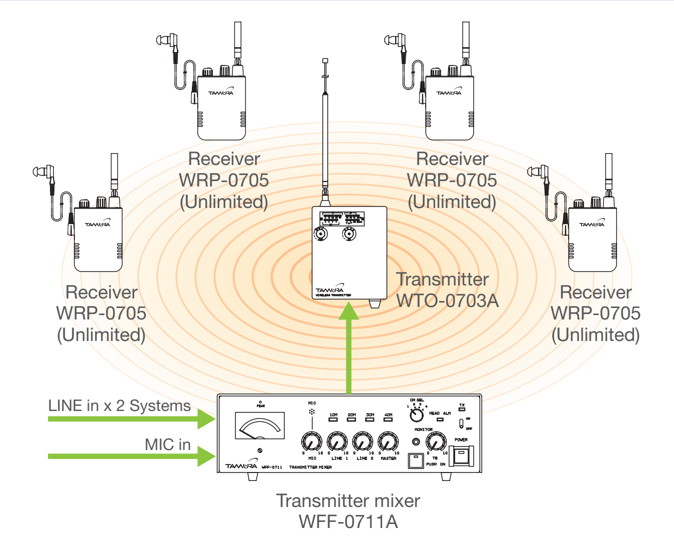
FM70MHz Band Output 10 mW License Free

Whether indoors or outdoors, the system ensures stable communication areas with high sound quality.
The system's small light-weight receiver and small transmitter in the same shape are easy to carry around with no burden.

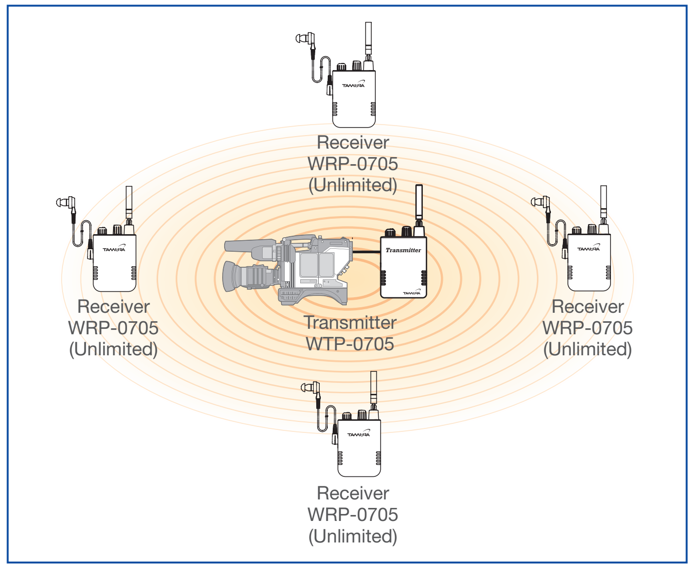
Main features

- **The transmitter mixer can change transmitter frequency and turn it on and off by remote control.**
- **It is possible to use a microphone cable to connect between the transmitter mixer and the transmitter.**
- **There are two types of transmitters: a long-antenna type designed for a long transmission distance (WTO-0703A) and a portable type designed for mobility (WTP-0705).**
- **One frequency is selected from 4 frequencies (74.58MHz / 74.64 / 74.70 / 74.7) for a channel. Up to 3 frequencies can be used in the same area.**
- **Focusing on stability, a transmission output of 10 mW (0.01 W) is designed even with a small power, achieving high-quality sound with a frequency response of 100 Hz to 8 kHz.**

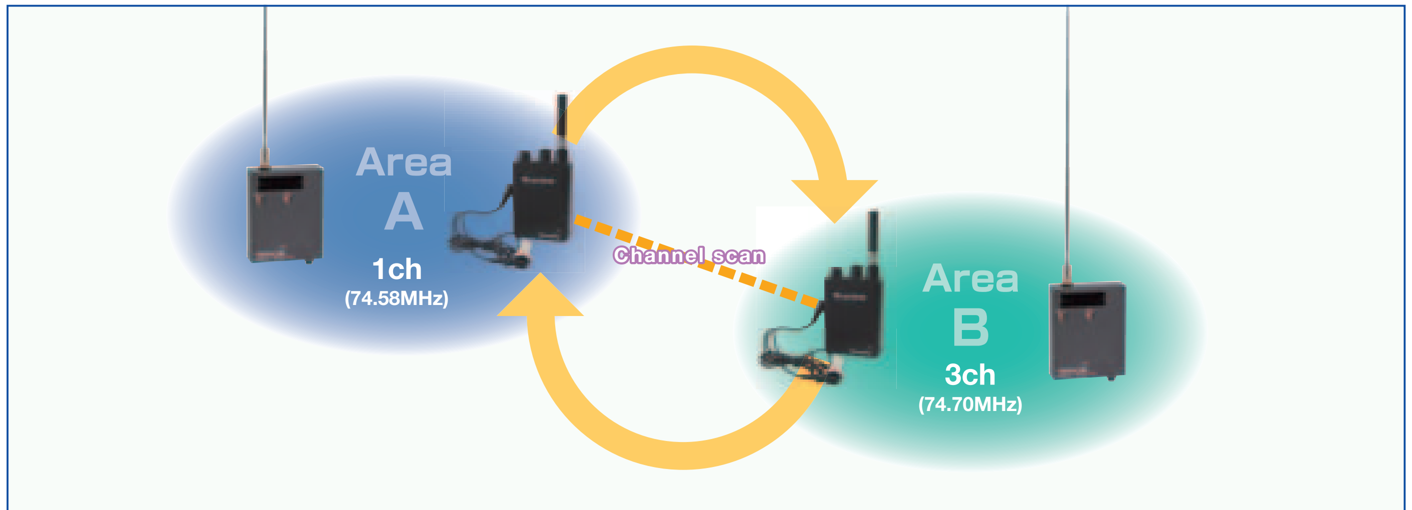
Example 1



Example 2



Channel scan function



Since the receiver has a channel scan function, it is possible to automatically change the channel, even if the channel of the transmitter is changed or the system is moved to a different area.

Transmitter WTP-0705

Rf Power Output	10mW
The Number of Channels	4 channels PLL type
Input	-60 / -20 / +4 dBm (600Ω)
Structure	Portable size
Power Source	DC1.5V (AA alkaline)
Available Time	More than 10 hours (1 AA alkaline)
Environment	-10℃～+50℃
Weight	100g (Including battery)
Dimensions	Width: 60mm; Height: 80mm; Depth:19mm (not including protruding portions)

Transmitter WTO-0703A

Rf power output	10mW
The Number of Channels	4Channels PLL Type
Input	-60 / -20 / +4 dBm (600Ω)
Structure	Wall-mounting and microphone stand mounting system
Power Source	DC10V(supplied by WFF-0711A) DC7～15V(External Power) More than 18 hours(AA alkaline×2)
Environment	-10℃～+50℃
Weight	680g (Including batteries)
Dimensions	Width: 105mm; Height: 130mm; Depth: 35mm (not including protruding portions)

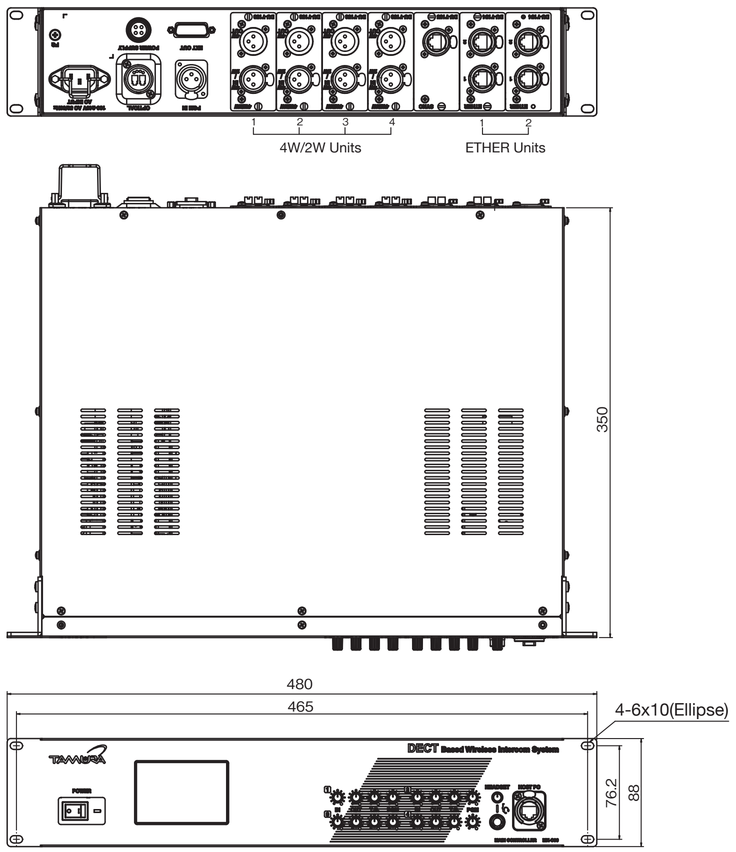
Transmitter mixer WFF-0711A

Input	LINE ×2 MIC ×1(600Ω)
Output	LINE ×1 (Transformer Balanced)
Channel Change	Remote Control, 4 Channels
Frequency Characteristic	100Hz ～ 10kHz
Environment	-10℃～+50℃
Power Source	AC100V / DC12V
Wight	2.5kg
Dimensions	Width: 260mm; Height: 77mm; Depth:250mm (not including protruding portions)

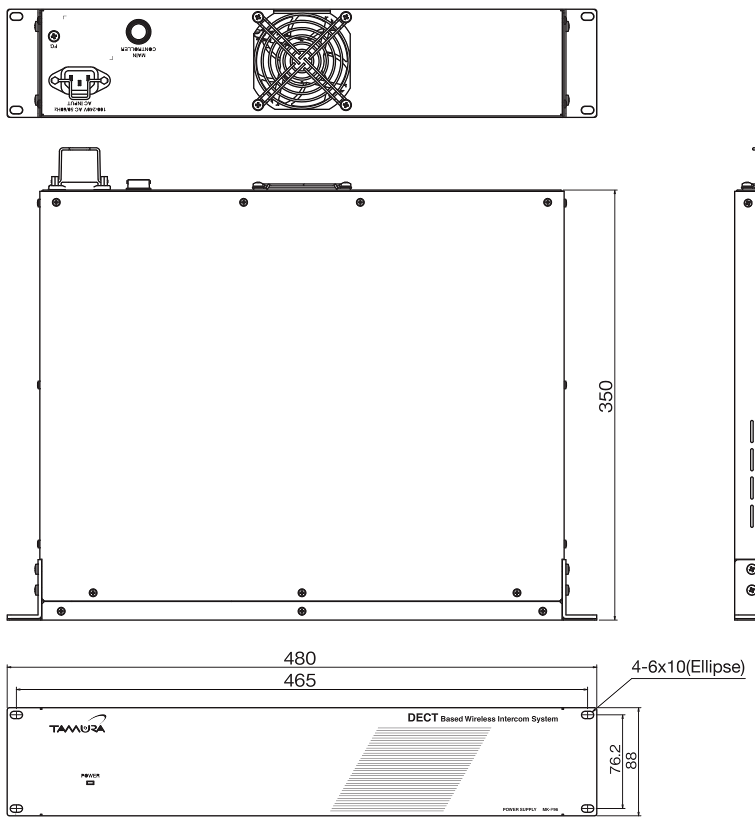
Receiver WRP-0705

Receive Sensitivity	Less than 2μV (SINAD12)
Type of Reception	Space Diversity
The Number of Channels	4 Channels PLL type
Structure	Portable size
Power Source	DC1.5V (AA alkaline)
Available Time	More than 18 Hours
Environment	-10℃～+50℃
Wight	100g (Including Battery)
Dimensions	Width: 60mm; Height: 80mm; Depth: 19mm (not including protruding portions)

MK-C96
Maincontroller

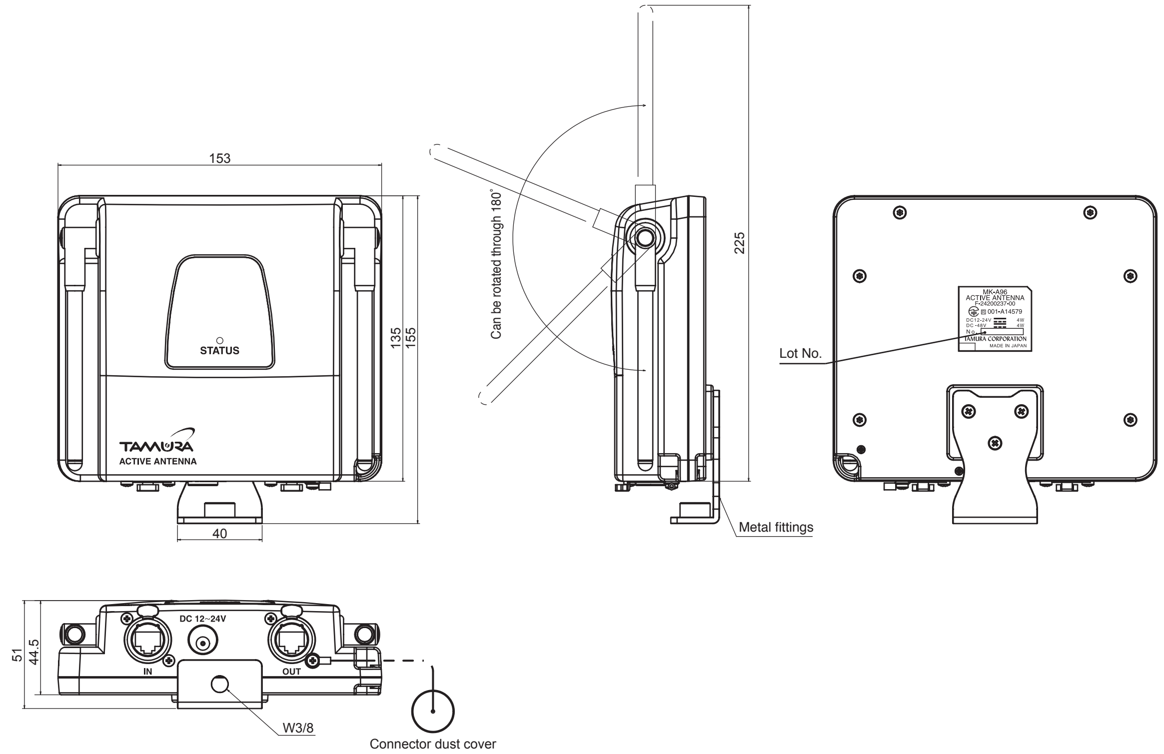


MK-P96
Subcontroller

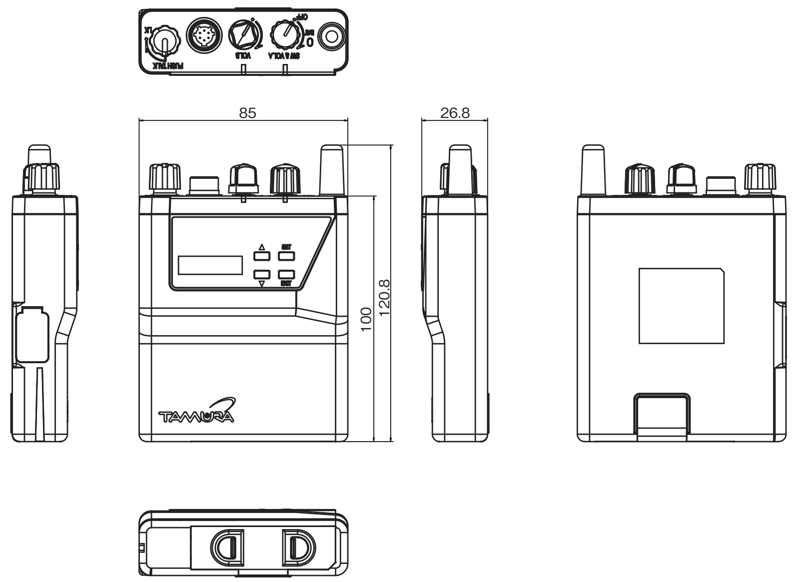


Dimensions DECT Based Wireless Intercom System

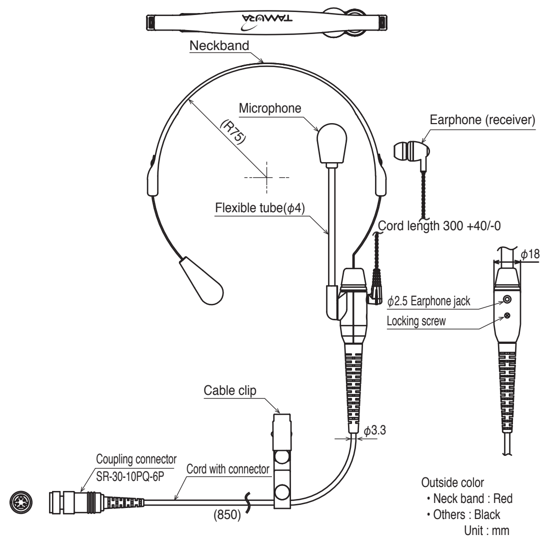
MK-A96
Activeantenna



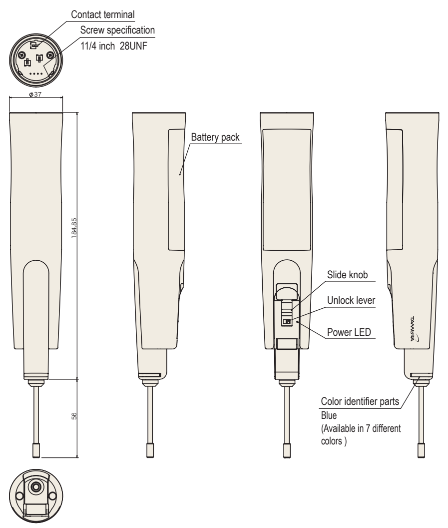
MK-B96
Personal Station



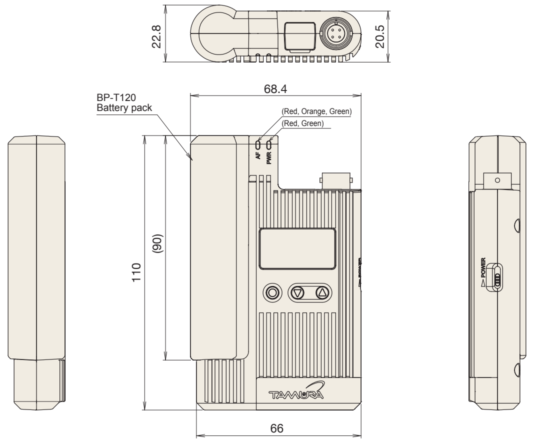
MK-316C
HEADSET



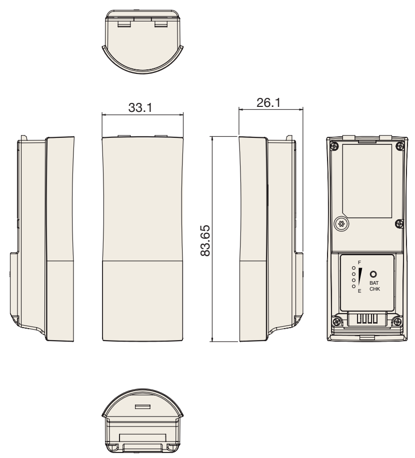
TWO-H120A
Digital wireless microphone (handheld type)



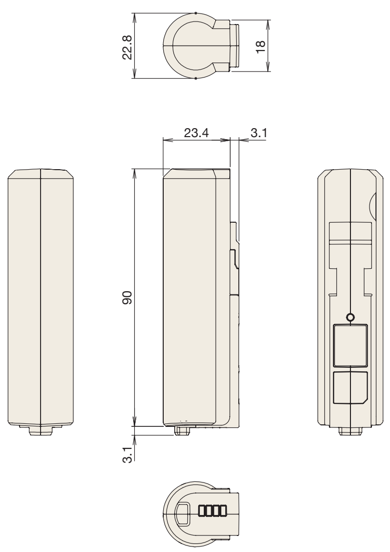
TWO-T120
Digital wireless microphone (two-piece type)



BP-H120
Battery pack handheld type (for TWO-H120)

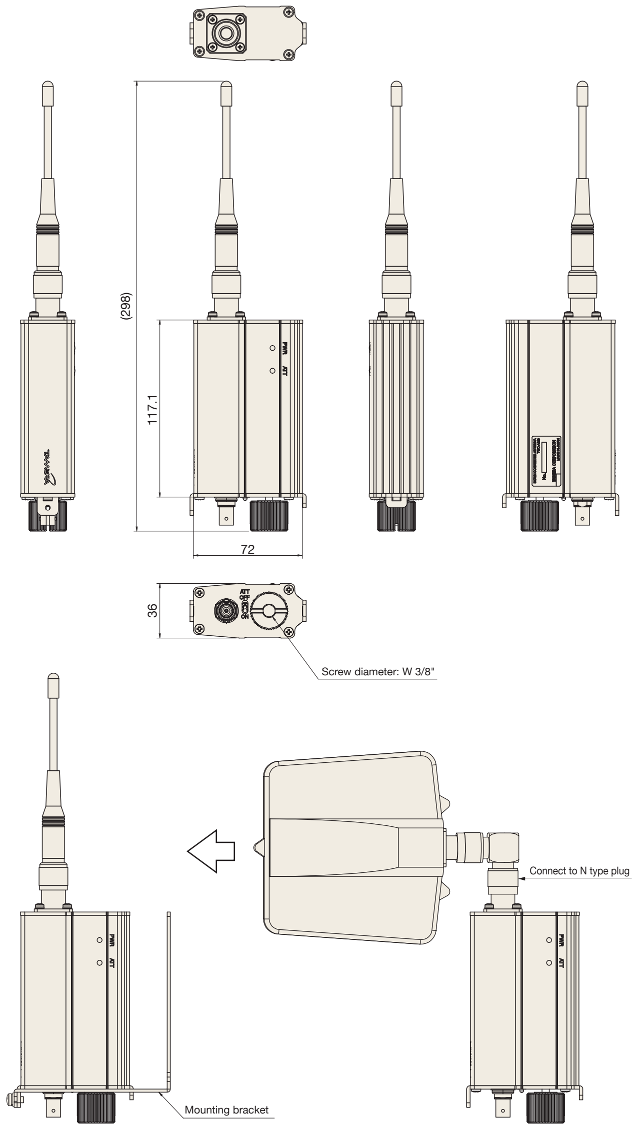


BP-T120
Battery pack two-piece (for TWO-T120)



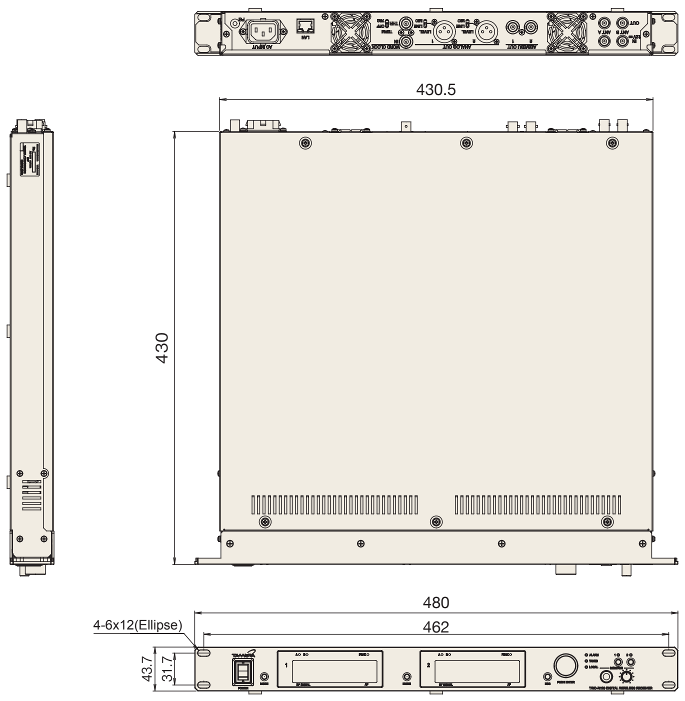
TWO-A120
Antenna with a built-in down converter (omnidirectional)

TWO-AY120
Antenna (directional)

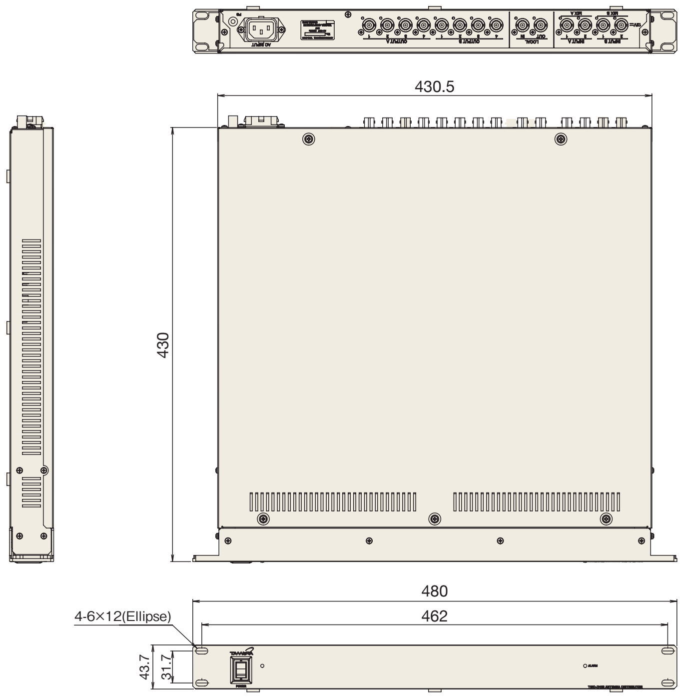


Dimensions OFDM Digital Wireless Microphone System

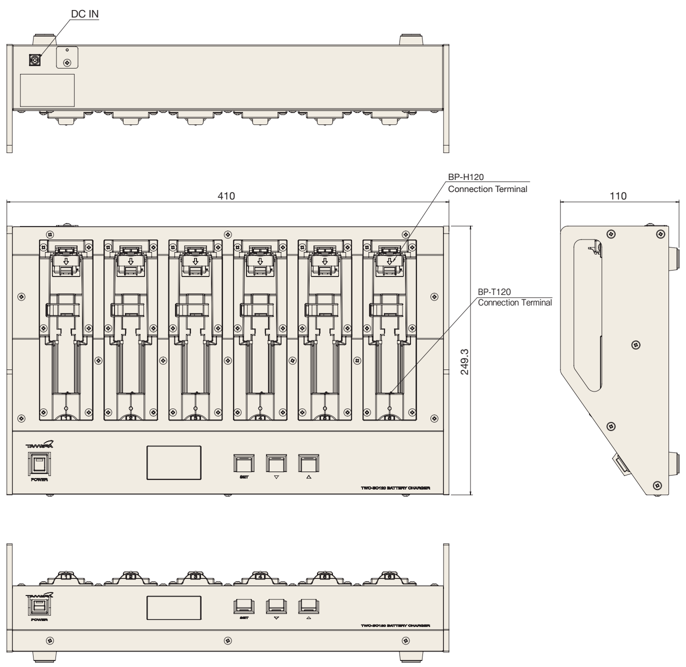
TWO-R120
Digital wireless receiver (2-ch implementation, 1U type)



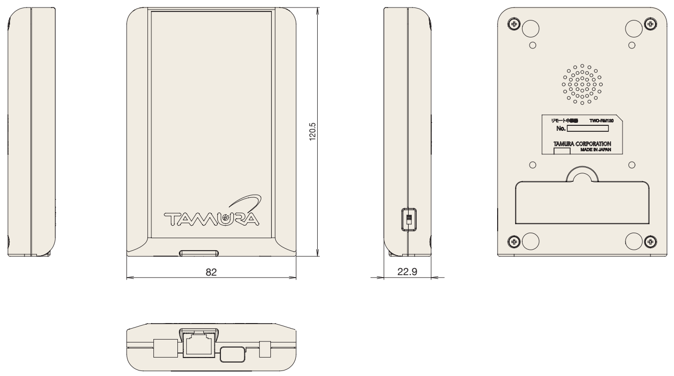
TWO-D120
Antenna mixing/distributing device



TWO-BC120
Charger

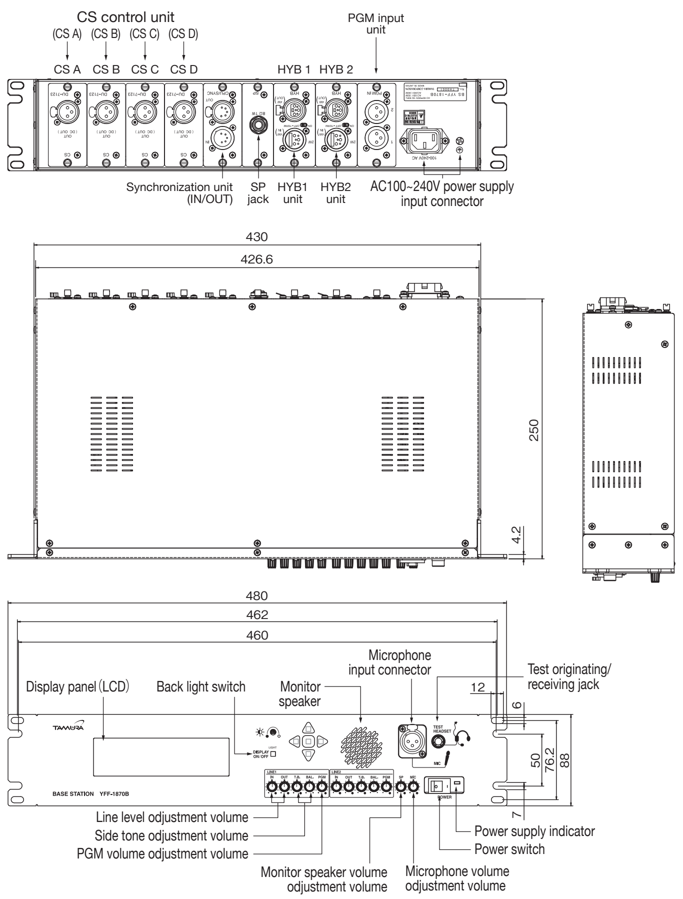


TWO-RM120
Remote repeater

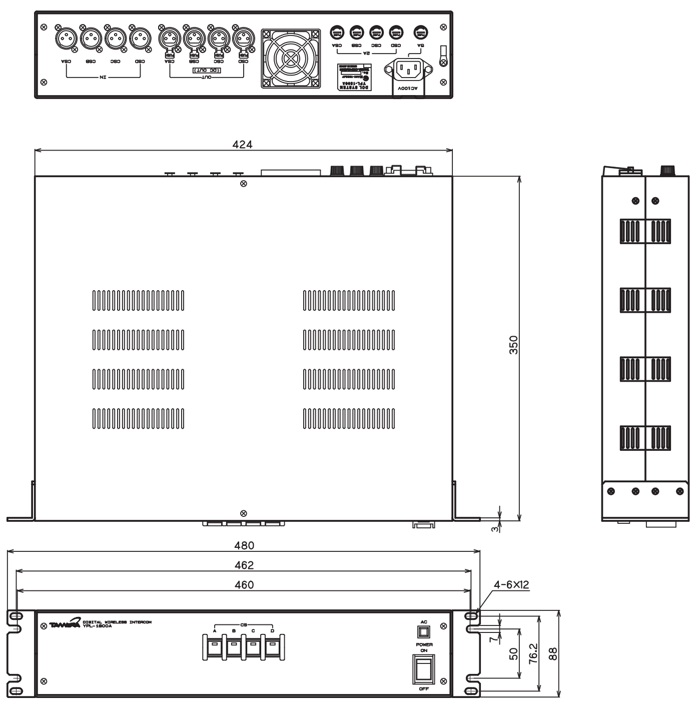


Dimensions Digital Wireless Intercom System

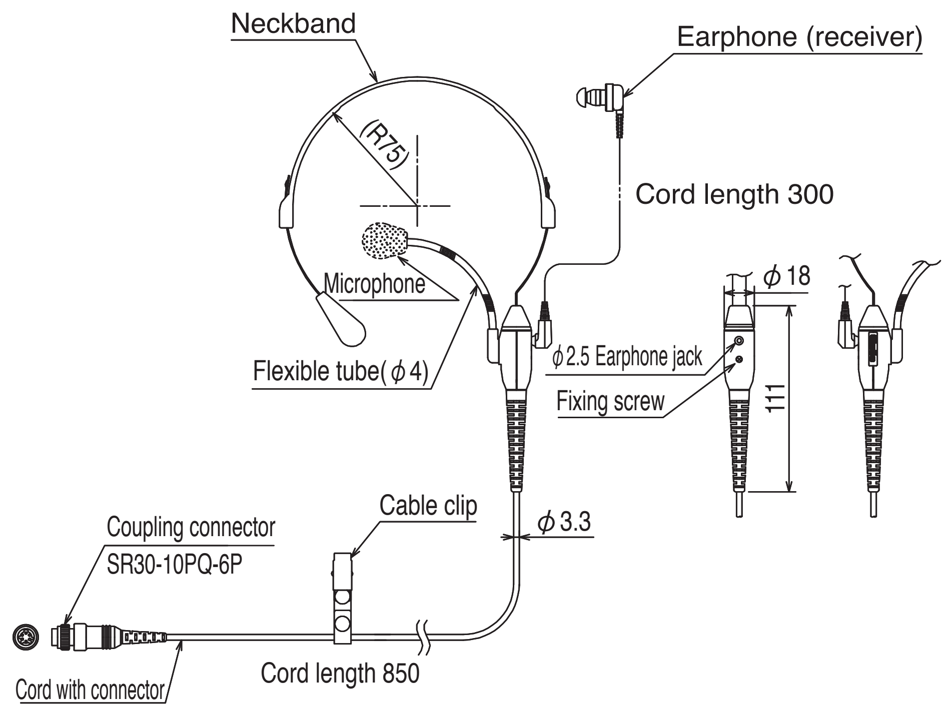
YFF-1870B
Base Station (BS)



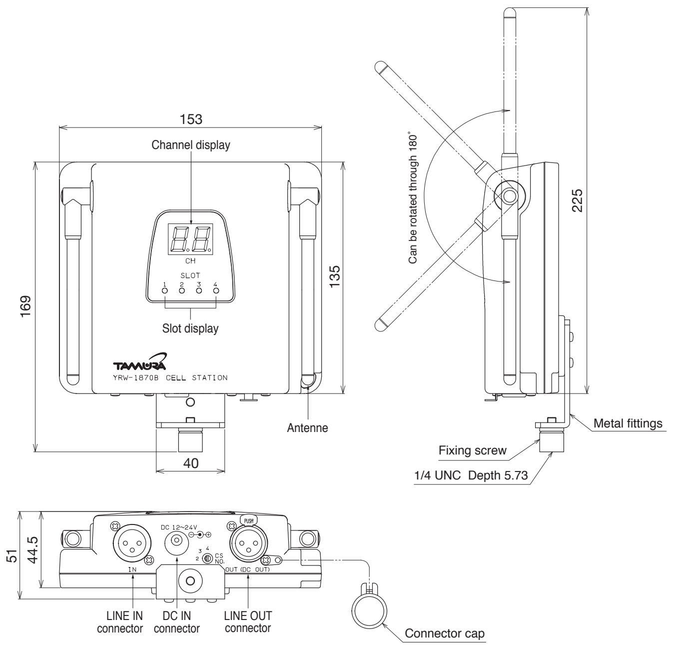
YPL-1800A
Power UNIT



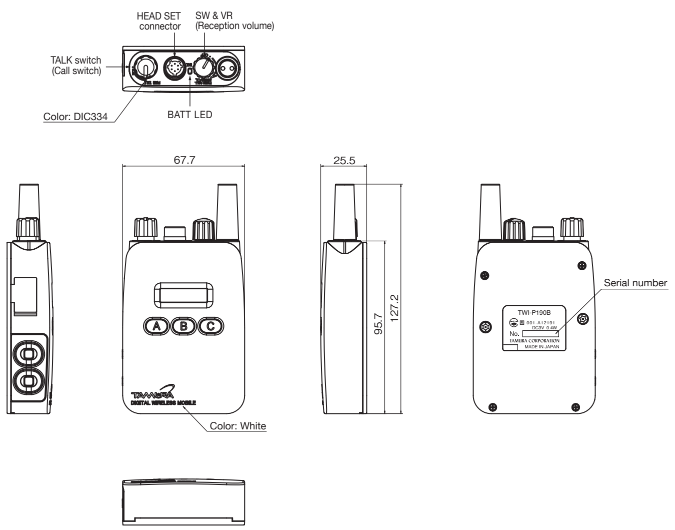
HS-316C
HEADSET



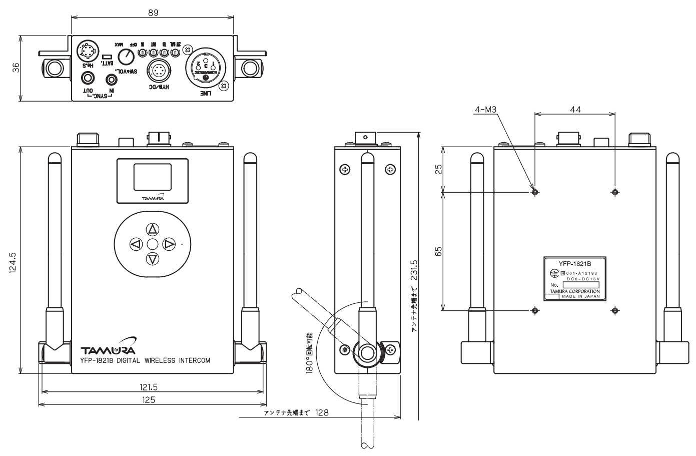
YRW-1870B
Cell Station (CS)



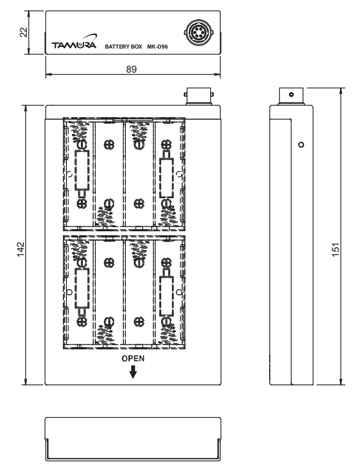
TWI-P190B
Personal Station (PS)



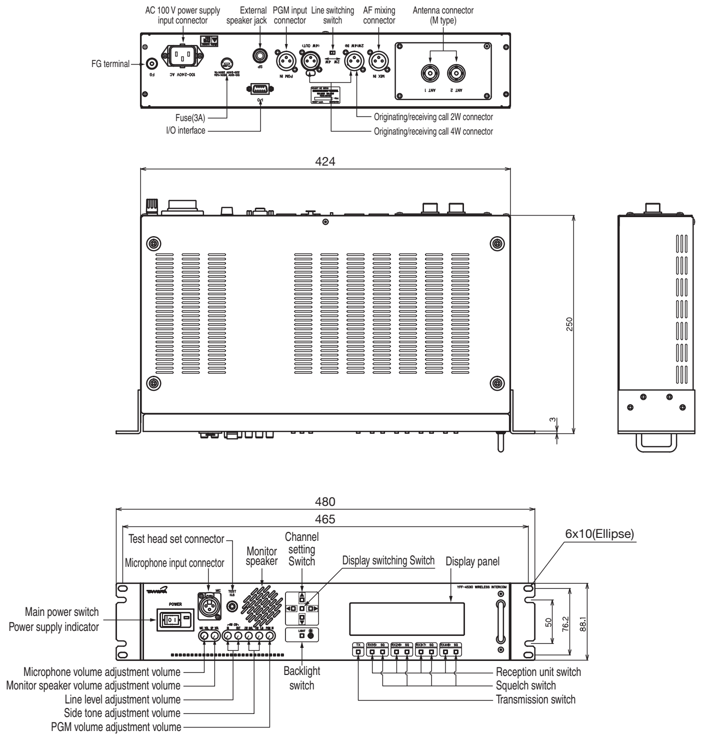
YFP-1821B
Cell Station (CS) Portable type System



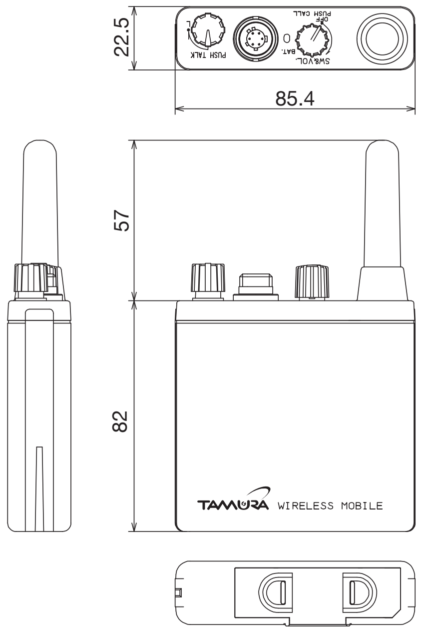
MK-D96
Battery box



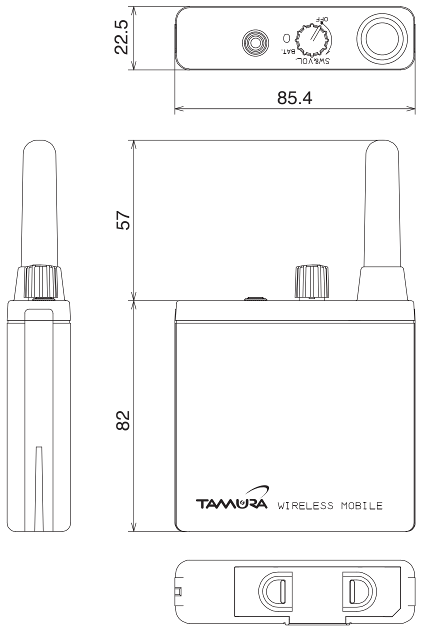
YFF-4530
Land mobile station Base Station



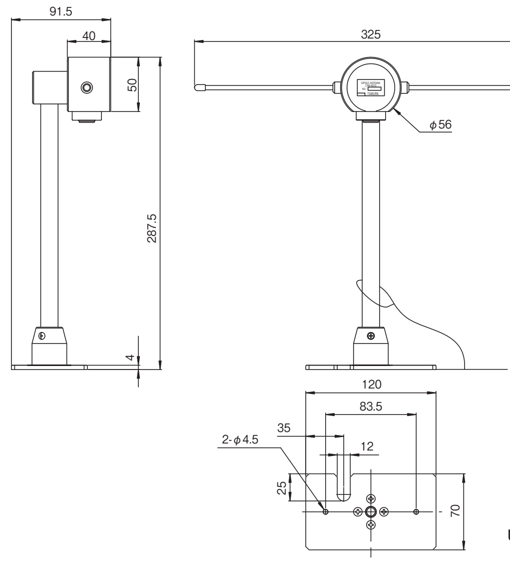
YMT-4120
Personal Station



YRT-4120
Command receiving device

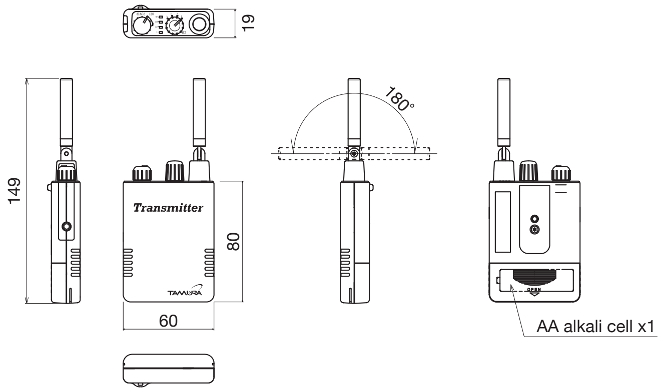


CAW-4510
Antenna

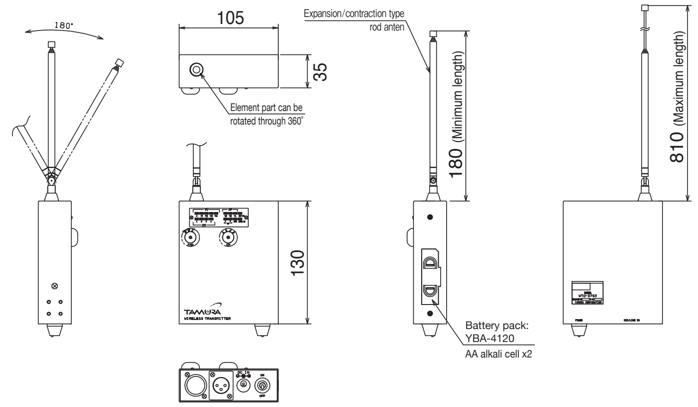


Unit: mm

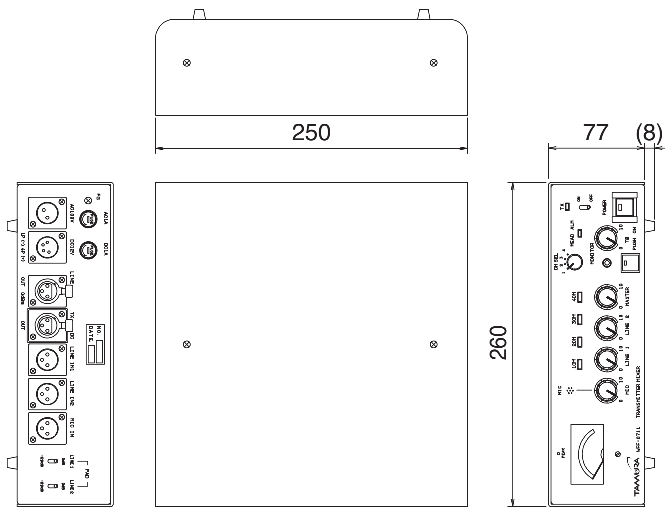
WTP-0705
Transmitter



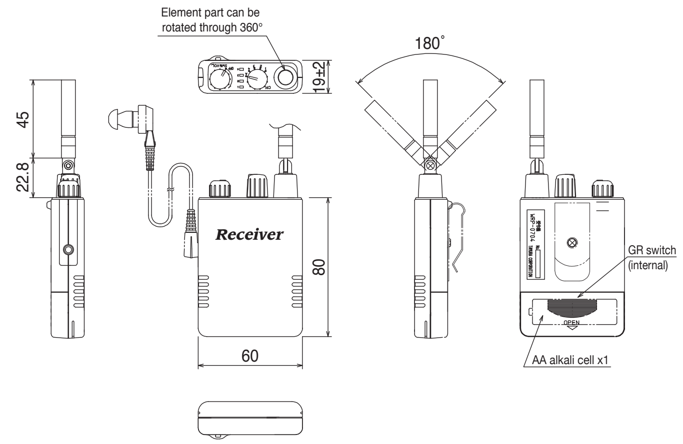
WTO-0703A
Transmitter



WFF-0711A
Transmitter mixer



WRP-0705
Receiver



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