Audio Equipment
&
Communication Systems
Audio Equipment
Communication Systems

DIGITAL AUDIO MIXING CONSOLE
NT Series

NT880

NT660

NT110
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

- **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.

- **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.

- **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 processing 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.

- **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.

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.

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.

IO Sharing

- **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**

<table>
<thead>
<tr>
<th>Maker</th>
<th>Product</th>
</tr>
</thead>
<tbody>
<tr>
<td>TAMURA</td>
<td>NT880 Digital Audio Mixing Console</td>
</tr>
<tr>
<td>TAMURA</td>
<td>NT660 Digital Audio Mixing Console</td>
</tr>
<tr>
<td>TAMURA</td>
<td>SYSTEM ROUTER</td>
</tr>
<tr>
<td>TAMURA</td>
<td>NT110 Digital Audio Mixer</td>
</tr>
</tbody>
</table>

*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.

- **NT NET Manager**
  - Up to 4 units
- **NT series**
  - Up to 8 units

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

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)
- DSP CORE: Maximum 5 DSP core units (including 1 backup unit)
- Number of TR-Link audio channels: 512ch

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 Dub/MC/Line IN card
  - 8ch BNC AES IN card
  - HD/SDI card
  - 8ch Dub LINE OUT card
  - 8ch BNC AES OUT card
  - MADI IO card
  - GPIO card
  - Dante card

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.

Main Specifications of AUTOMIX

<table>
<thead>
<tr>
<th>Item</th>
<th>Specifications</th>
</tr>
</thead>
<tbody>
<tr>
<td>No. of Automix SHARC DSPs</td>
<td>Maximum 4</td>
</tr>
<tr>
<td>No. of Automix channels</td>
<td>16ch</td>
</tr>
<tr>
<td>Automix ch format</td>
<td>Mono</td>
</tr>
<tr>
<td>Sample freq</td>
<td>FS 48k</td>
</tr>
<tr>
<td>Connect ch type</td>
<td>HA/Line Input Group</td>
</tr>
<tr>
<td>Connect ch format</td>
<td>Mono/Stereo/5.1</td>
</tr>
<tr>
<td>Connect ch signal path</td>
<td>Depends on the insertion path</td>
</tr>
</tbody>
</table>

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.
### Option card

**8ch DSUB MIC/LINE IN Card**
Audio interface card of analog 8ch input. Mic/Line setting can be changed.

<table>
<thead>
<tr>
<th>Number of Occupied slots</th>
<th>1 slot</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic/Line Input</td>
<td>balanced type</td>
</tr>
<tr>
<td>Number of Channels</td>
<td>8ch</td>
</tr>
<tr>
<td>[Mic input] Input level</td>
<td>-12dBu to +12dBu (0.1dB step selected)</td>
</tr>
<tr>
<td>[Line input] Input level</td>
<td>-12dBu to +12dBu (0.1dB step selected)</td>
</tr>
<tr>
<td>[Line input] Input Impedance</td>
<td>50Ω / 10kΩ or more</td>
</tr>
</tbody>
</table>

**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.

<table>
<thead>
<tr>
<th>Number of Occupied slots</th>
<th>1 slot</th>
</tr>
</thead>
<tbody>
<tr>
<td>Format</td>
<td>AES-3id</td>
</tr>
<tr>
<td>Number of Channels</td>
<td>8ch AES3</td>
</tr>
<tr>
<td>Input Impedance</td>
<td>75Ω unbalanced type</td>
</tr>
<tr>
<td>Input sampling frequency (SRC ON)</td>
<td>32~100kHz</td>
</tr>
<tr>
<td>Input sampling frequency (SRC OFF)</td>
<td>48 / 96kHz (Synchronized with the system clock)</td>
</tr>
<tr>
<td>Number of input bits</td>
<td>16 ~ 24bit</td>
</tr>
</tbody>
</table>

**8ch BNC AES3 OUT Card**
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.

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

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

<table>
<thead>
<tr>
<th>HD-SDI Card</th>
<th>3G-SDI Card</th>
<th>12G-SDI Card</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of Occupied slots</td>
<td>1 slot</td>
<td>1 slot</td>
</tr>
<tr>
<td>Supported SDI formats</td>
<td>75Ω / 50Ω, 9.94MHz, 1035i 59.94 / 60Hz, 1035i 59.94 / 60Hz, 1080i 24 / 25 / 29.97 / 30Hz, 1080i 24 / 25 / 29.97 / 30Hz, 2160p 59.94p</td>
<td></td>
</tr>
<tr>
<td>Input sampling frequency</td>
<td>1035i 59.94 / 60Hz, 1035i 59.94 / 60Hz, 1080p 25 / 29.97 / 30Hz</td>
<td></td>
</tr>
<tr>
<td>Number of input bits</td>
<td>16 / 24bit</td>
<td>16 / 24bit</td>
</tr>
<tr>
<td>Loop Through output</td>
<td>80Ω IN BNC, 75Ω IN BNC</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Dante Card</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sampling frequency (Fs)</td>
</tr>
<tr>
<td>Number of Input bits</td>
</tr>
<tr>
<td>Number of Input bits</td>
</tr>
<tr>
<td>Transmission Protocol</td>
</tr>
<tr>
<td>Dante Connector</td>
</tr>
</tbody>
</table>

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

#### [GPI]

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Link Function FUNST</td>
<td>When On, the specified FU function is set to On or BT is On</td>
</tr>
<tr>
<td>Link Function Remote</td>
<td>When On, the specified remote function is set to On</td>
</tr>
<tr>
<td>Link Function AVL</td>
<td>When On, AVL function is set to On</td>
</tr>
<tr>
<td>System Tally 1</td>
<td>Indicator LAMP 1 is lit when Indicator LAMP 1 for OSC and TB prohibits control is lighted</td>
</tr>
<tr>
<td>System Tally 2</td>
<td>Indicator LAMP 2 is lit when System Tally 2 for OSC and TB prohibits control is lighted</td>
</tr>
<tr>
<td>System Tally 3</td>
<td>Indicator LAMP 3 is lit when System Tally 3 for OSC and TB prohibits control is lighted</td>
</tr>
<tr>
<td>Monitor Cut</td>
<td>When On, the specified monitor is disconnected</td>
</tr>
<tr>
<td>Monitor Dim</td>
<td>When On, the specified monitor is dimming</td>
</tr>
<tr>
<td>Output Matrix switching</td>
<td>When On, Out Source of specified TR-Link channel is altered</td>
</tr>
<tr>
<td>Send Ext Int Disable</td>
<td>When On, the Ext Int function of the specified Bus is disabled</td>
</tr>
<tr>
<td>Input Only for GPI Link</td>
<td>When On, the specified Bus is input only for GPI Link</td>
</tr>
<tr>
<td>TB interruption</td>
<td>When On, TB audio interruption is generated in the specified Bus</td>
</tr>
<tr>
<td>OSC interruption</td>
<td>When On, OSC interruption is generated in Master Bus</td>
</tr>
<tr>
<td>More Source switching</td>
<td>When On, Monitor Source is changed</td>
</tr>
<tr>
<td>GPI REM Sw</td>
<td>When On, console [REM] button is On</td>
</tr>
</tbody>
</table>

#### [GPO]

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Link Function Remote</td>
<td>On output when the function is in the specified status</td>
</tr>
<tr>
<td>Console Mode Notification</td>
<td>On output when the specified Console Mode is set to On</td>
</tr>
<tr>
<td>OSC On Notification</td>
<td>On output when OSC is On</td>
</tr>
<tr>
<td>GPI Link</td>
<td>Output input/1st output with the specified GPI state</td>
</tr>
<tr>
<td>PFL On/Off Notification</td>
<td>Output PFL On/Off status</td>
</tr>
<tr>
<td>AFL On/Off Notification</td>
<td>Output AFL On/Off status</td>
</tr>
<tr>
<td>FU On</td>
<td>Output FU On/Off status of specified FU number</td>
</tr>
<tr>
<td>TB status Notification</td>
<td>Output of TB interrupt status to specified Bus</td>
</tr>
<tr>
<td>Mic On</td>
<td>Output of Mic On status of specified FU number</td>
</tr>
</tbody>
</table>
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. (*)

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.

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

Audio control parameters
- HA Gain: +10dBu~+66dBu
- 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 1 channel

Audio channel (P=48kHz)
- Master Bus: Maximum 24 buses
- 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)

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

Dimensions

1383 x 243 x 189
30 Fader

1687 x 2031 x 2365
50 Fader

2031 x 2031 x 2031
60 Fader

Audio block diagram
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.

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.

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

#### Console
- **Supply voltage**: AC100-240V 50/60Hz
- **Maximum number of physical faders**: 20/30/40/50
- **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
- **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 1 channel

### Dimensions

#### NT10 Fader
- **20 Fader**: 912 x 1588
- **30 Fader**: 1250 x 825
- **40 Fader**: 1588 x 825
- **50 Fader**: 1826 x 900

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Digital Audio Mixer
Operability of trust

- 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)

- 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.

- 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 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.

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<tr>
<td>TAMURA</td>
<td>TU-6439 Dante CARD</td>
<td>NT110 Dante Add port card</td>
</tr>
<tr>
<td>YAMAHA</td>
<td>Rio3224-D</td>
<td>I/O RACK</td>
</tr>
<tr>
<td>YAMAHA</td>
<td>Rio1608-D</td>
<td>I/O RACK</td>
</tr>
<tr>
<td>YAMAHA</td>
<td>R8-D</td>
<td>I/O RACK</td>
</tr>
<tr>
<td>YAMAHA</td>
<td>Rio3224-D2</td>
<td>I/O RACK</td>
</tr>
<tr>
<td>YAMAHA</td>
<td>Rio1608-D2</td>
<td>I/O RACK</td>
</tr>
</tbody>
</table>

Multi Meter

- 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.

- 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° to 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 ~ -64dBu
  (Analog LINE): +4dBu
- Audio Reference Output Level
  (Analog LINE): +4dBu
- 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

Dimensions

Control Panel Description

Front panel
- TOUCH PANEL
- PRESET PROGRAM

Rear panel
- POWER SW
- AC INPUT
- DC INPUT / NFB
- IO CARD SLOT
- MADI CARD
- MADI CARD
- MADI CARD
- MADI CARD

NT110 with Tamura Resource Network Technology
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 by implementing a card.

Sampling frequency (Fs) 48kHz / 96kHz
LAN Connector RJ-45 type
Dimensions 171(W)x49.5(H)x304(D)mm
Weight 550g

■ MIC/LINE IN CARD
This card is audio interface card that inputs microphone level and line level analog audio signals.

General-purpose control
signal inputs (MIC INPUT)
37-pin D-type connector (male)
Dimensions 125(W)x40(H)x152(D)mm
Weight 185g

■ LINE OUT Card
This card is audio interface card that outputs line level analog audio signals.

Audio Reference Input Level 10dBu ± 0.5dB
Output impedance Less Than 50Ω
General
Transmission frequency range 20 - 20,000Hz
Sampling frequency (Fs) 48kHz / 96kHz
Dimensions 129(W)x40(H)x152(D)mm
Weight 180g
Connector 25pin D-type connector (female)

■ MIC/LINE IN CARD
This card is audio interface card that inputs microphone level and line level analog audio signals.

Audio Reference Input Level 10dBu ± 0.5dB
Output impedance Less Than 50Ω
General
Transmission frequency range 20 - 20,000Hz
Sampling frequency (Fs) 48kHz / 96kHz
Dimensions 129(W)x40(H)x152(D)mm
Weight 180g
Connector 25pin D-type connector (female)

■ AES3id Card
Audio interface card of 4ch AES3 input / 4ch AES3 output.

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

■ 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 ethercon Connector
Dimensions 129(W)x40(H)x152(D)mm
Weight 150g

■ VCA Card (for NT MATRIX)
Interface card for input of VCA control signals.

LINE OUTPUT (CH1~CH8)
Audio Reference Input Level 10dBu ± 0.5dB
Output impedance Less Than 50Ω
General
Transmission frequency range 20 - 20,000Hz
Sampling frequency (Fs) 48kHz / 96kHz
Dimensions 129(W)x40(H)x152(D)mm
Weight 180g
Connector 37pin D-type connector (male)

■ VCA INPUT (CH1~CH8)
Reference voltage 0V DC
Compatible potentiometer Linear type, TML1
General
Dimensions 129(W)x40(H)x152(D)mm
Weight 150g
Connector 25pin D-type connector (female)

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

General-purpose control
signal inputs (GPIO INPUT)
37-pin D-type connector (male)
Dimensions 125(W)x40(H)x152(D)mm
Weight 185g

■ VCA Card (for NT MATRIX)
Interface card for input of VCA control signals.

VCA INPUT (CH1~CH8)
Reference voltage 0V DC
Compatible potentiometer Linear type, TML1
General
Dimensions 129(W)x40(H)x152(D)mm
Weight 150g
Connector 25pin D-type connector (female)

■ 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

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

MADI INPUT (CH1~CH16)
Reference voltage +5V DC
Compatible potentiometer Linear curve, 10kΩ
General
Dimensions 129(W)x40(H)x152(D)mm
Weight 160g
Connector 25pin D-type connector (male) x2

Audiwave®, the Audiwave logo and Dante are trademarks of Audinate Pty Ltd.
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/
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

Dimensions

NT MATRIX System

- Windows PC
- Dedicated app or custom app
- Custom panel
- GPIO TALLY
- Video system

- NT MATRIX
  - Maximum 160ch AUDIO INPUT
  - Maximum 160ch AUDIO OUTPUT

- Dimensions: 430 x 482 x 133
**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**

<table>
<thead>
<tr>
<th>Items</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>AUDIO ROUTER</td>
<td>160ch x 160ch</td>
</tr>
<tr>
<td>DSP PROCESS</td>
<td>32in x 32out DSP x 6</td>
</tr>
<tr>
<td>DSP FUNCTION</td>
<td>32in x 32out Mix Matrix or Filter/Limiter, AUD, Internal OSC</td>
</tr>
<tr>
<td>CONTROL PORT</td>
<td>LAN/RJ45/RS422/RS232/GPI/GPIO/Power</td>
</tr>
<tr>
<td>SYNCHRONIZED INPUT SIGNAL</td>
<td>WORD CLOCK/VIDEO</td>
</tr>
<tr>
<td>POWER SUPPLY</td>
<td>AC100-240V 50/60Hz</td>
</tr>
<tr>
<td>OPERATION TEMPERATURE</td>
<td>-10 ~ 40 °C</td>
</tr>
<tr>
<td>EXTERNAL DIMENSIONS (WxDxH)</td>
<td>482 x 554 x 133</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Items</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSP CARD</td>
<td>Redundancy DSP CARD</td>
</tr>
<tr>
<td>MIC/LINE INPUT CARD</td>
<td>MIC INPUT 4ch + LINE INPUT 4ch</td>
</tr>
<tr>
<td>LINE OUTPUT CARD</td>
<td>LINE OUTPUT 8ch</td>
</tr>
<tr>
<td>AES3id CARD</td>
<td>AES3id INPUT 4ch + AES3 id OUTPUT 4ch</td>
</tr>
<tr>
<td>MADI CARD</td>
<td>MADI INPUT 1ch + MADI OUTPUT 1ch (OPTICAL &amp; COAXIAL)</td>
</tr>
<tr>
<td>Dante CARD</td>
<td>Dante 1ch (Primary &amp; Secondary)</td>
</tr>
<tr>
<td>GPIO CARD</td>
<td>GPIO INPUT 16ch + GPIO OUTPUT 16ch</td>
</tr>
<tr>
<td>VCA CARD</td>
<td>VCA INPUT 16ch</td>
</tr>
</tbody>
</table>

**Example of application**

**Audio Router (Matrix)**

**N-1 Sending back system**

**Output Matrix**
Example of application

OTC system

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
Illuminated 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.
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Analog Wireless Intercom System
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OFDM Digital Wireless Microphone System
P. 60~64

Wireless Monitoring System
P. 65~67
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.

<table>
<thead>
<tr>
<th>Simultaneous calls only</th>
<th>Combined with dedicated command-receiving devices</th>
</tr>
</thead>
<tbody>
<tr>
<td>Max. 60 personal stations for simultaneous calls connected</td>
<td>Max. 48 personal stations for simultaneous calls connected</td>
</tr>
</tbody>
</table>
| + | +
| Max. 128 personal stations for receiving commands connected | Max. 128 personal stations for receiving commands connected |

“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.
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.
**Personal Station MK-B96**

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

**System Configuration Examples**

**Basic configuration**

A maximum of four units can be connected in cascade.

**Configuration Example Using HUB**

- **Active antenna:** Max. 4 lines
- **Ethernet cable (LAN cable)**

**Main System Specifications**

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

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

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

**Microphone**

- Impedance: 1.6kΩ
- Sensitivity: –73.0dB
- Frequency characteristics: 100Hz~10kHz

**Receiver**

- Impedance: 16Ω
- Rated input: 1mW
- Maximum permissible input: 300mW
- Output sound pressure level: 101.5dB
- Frequency characteristics: 20Hz ~ 9kHz

**Appearance**

- MK-316C (Condenser type)
- HS-316C (Condenser type)
- HS-126D (Dynamic type)

**Main System Specifications**

<table>
<thead>
<tr>
<th>Item</th>
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<tr>
<td>Max. no. of stations connected per system</td>
<td>60 personal stations for calls or 48 personal stations for calls + 128 command-receiving devices</td>
</tr>
<tr>
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</tr>
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</tr>
<tr>
<td>Continuous use time for personal station</td>
<td>Approx. 8 hours (AA alkaline battery x 2)</td>
</tr>
</tbody>
</table>
DECT Based Wireless Intercom System

System Configuration Examples

When using optical cable

- Active antenna: Max. 4 lines
- Optical media converter
- Optical cable (single mode)
- Power

max 2.5 km

Call groups: Max. 4 groups (4W/2W)

Max. Configuration

- Ethernet cable (LAN cable)
- Max 16 units
- Active antenna: Max. 4 lines
- 100m

Use a power supply when connecting 5 or more active

��統配置例

1) 使用光纖時

- 主要天線：最大 4 線路
- 光纖（單模式）
- 功率

max 2.5 km

- 調頻：最大 4 績

最大配置

- 埃塞特電纜 (LAN 電纜)
- 最多 16 單元
- 主要天線：最多 4 線路
- 100m

當連接 5 台或以上運作時，請使用電源供應。
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 BS units can be connected
- Rack-mount type EIA=2U, JIS=2U

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

YPL-1800A  Production on order

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

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 certified
Environment: -10~+50˚C
Weight: Approx. 430g (fittings included)
Dimensions: Width: 153mm; height: 135mm; depth: 45mm
(Excluding the dimensions of the protrusions)

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 certified
Environment: -10~+50˚C
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: DC 24 V (supplied from the main device)
DC 12V (external power supply)
Number of calls: Simultaneous calls are possible in 1: 4
Antenna: Case-integrated (removal prohibited)
Gain: 2.14 dB or less
Channel setting: Multi-channel access system
Standards: Technical standard conformance has been certified
Environment: -10~+50˚C
*HS-316C is exclusive for personal station

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)
Inductance: 0.45mH±10%
DC resistance: 7.7Ω±10%
Maximum permissible input: 500mW
Output sound pressure level: 112dB at 1kHz (dB=2×10^-5 Pa)
Frequency characteristics: 50Hz~5kHz

Specifications (HS-126D)
Microphone part (dynamic type)
Impedance: 200Ω±20% at 1kHz
Inductance: 1.96mH±10%
DC resistance: 190Ω±10%
Sensitivity: –66dB±4dB at 1kHz (dB=1V/1Pa)
Frequency characteristics: 100Hz~5kHz
Receiver part
Impedance: 8Ω±10%
Inductance: 0.04mH±10%
DC resistance: 7.7Ω±10%
Maximum permissible input: 500mW
Output sound pressure level: 112dB at 1kHz (dB=2×10^-5 Pa)
Frequency characteristics: 50Hz~5kHz

Battery pack BH-190

AA alkaline cell×2

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)
Inductance: 0.45mH±10%
DC resistance: 7.7Ω±10%
Maximum permissible input: 500mW
Output sound pressure level: 112dB at 1kHz (dB=2×10^-5 Pa)
Frequency characteristics: 50Hz~5kHz

Specifications (HS-126D)
Microphone part (dynamic type)
Impedance: 200Ω±20% at 1kHz
Inductance: 1.96mH±10%
DC resistance: 190Ω±10%
Sensitivity: –66dB±4dB at 1kHz (dB=1V/1Pa)
Frequency characteristics: 100Hz~5kHz
Receiver part
Impedance: 8Ω±10%
Inductance: 0.04mH±10%
DC resistance: 7.7Ω±10%
Maximum permissible input: 500mW
Output sound pressure level: 112dB at 1kHz (dB=2×10^-5 Pa)
Frequency characteristics: 50Hz~5kHz

HEADSET

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)
Inductance: 0.45mH±10%
DC resistance: 7.7Ω±10%
Maximum permissible input: 500mW
Output sound pressure level: 112dB at 1kHz (dB=2×10^-5 Pa)
Frequency characteristics: 50Hz~5kHz

Specifications (HS-126D)
Microphone part (dynamic type)
Impedance: 200Ω±20% at 1kHz
Inductance: 1.96mH±10%
DC resistance: 190Ω±10%
Sensitivity: –66dB±4dB at 1kHz (dB=1V/1Pa)
Frequency characteristics: 100Hz~5kHz
Receiver part
Impedance: 8Ω±10%
Inductance: 0.04mH±10%
DC resistance: 7.7Ω±10%
Maximum permissible input: 500mW
Output sound pressure level: 112dB at 1kHz (dB=2×10^-5 Pa)
Frequency characteristics: 50Hz~5kHz
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.

System example -1

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

System example -2

BS: One unit  |  Power Unit: One unit  |  CS: 16 units  |  PS: 40 units
# Digital Wireless Intercom System

## Basic System

**Wired Intercom System, Connection with 2 groups is possible!**

- **Cell Station (CS):**
  - B/F part
  - Communication control part
  - AF part
- **Personal Station (PS):**
  - 2/4W

---

## Electrical Characteristics

<table>
<thead>
<tr>
<th></th>
<th>BS YF F-1870B</th>
<th>CS YRW-1870B</th>
<th>PS TWI-P190B</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Common</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>to high frequencies</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Radio wave type</td>
<td>G7D, G7E, G7X, G1D, G1E, G1X</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Antenna type</td>
<td>λ/2 sleeve antenna</td>
<td>Whip antenna</td>
<td></td>
</tr>
<tr>
<td>Antenna impedance</td>
<td>50Ω</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Frequency range</td>
<td>1895.15 ~ 1905.95MHz</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Number of frequencies</td>
<td>42 waves (control carrier 2 waves, communication carrier 40 waves)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Separation</td>
<td>30kHz</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Oscillation system</td>
<td>Quartz control frequency synthesizer system</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Frequency stability</td>
<td>Within ±3×10⁻６</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Modulation accuracy</td>
<td>12.5% or less</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

| **Transmission** |               |              |              |
| Antenna power    | 10mW          |              |              |
| Intensity of spurious radiation | 2.5μW or less (within band) |              |              |
| Modulation system | π/4 shift QPSK |              |              |
| Audio frequency  | 3.4kHz or less |              |              |
| Neighboring channel leak power | 600kHz mistuned 800μW or less, 900kHz mistuned 250μW or less |              |              |
| Occupied frequency band area | Within 288 kHz |              |              |

| **Reception**   |               |              |              |
| Reception system | Double superhetrodyne |              |              |
| Reception sensitivity | 16 dBμV or less (bit error rate 1×10⁻³) |              |              |
| Spurious sensitivity | 47 dB or more |              |              |
| Neighboring channel selectivity | 50 dB or more (600 kHz detuning) |              |              |
| Body radiation  | 4μW or less   |              |              |

| **Common**     |               |              |              |
| **Line frequency characteristic** | 3.4kHz or less |              |              |
| Line input/output | 0dBm balanced | –            |              |
| Microphone input | -60dBm balanced | -60dBm unbalanced |              |
| Speaker output  | Inside 1W Outside 2W at 8Ω | –            |              |
| External input  | 0dBm balanced | –            |              |
| Used power supply/ power consumption | AC100V±15%: 3A AC240V±15% | 130 mA or less at DC 3V |              |
| Use environment | Temperature: -10 ~ +50°C, Humidity: Within 30~90% |              |              |
Digital Wireless Intercom System

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: 0.3 - 4 kHz
- Audio encoding system: 32 kbit/s ADPCM
- Line specification:
  - 4W: IN 0 dBm, OUT 0 dBm
  - AIR: IN -20 dBm
- Microphone input: -50 dBm (unbalanced 600 Ω)
- Speaker output: 15 mW or more (at 8 Ω)
- Power supply: DC 8.0~16.0V (negatively grounded)
- Structure: Portable type
- Standards: Technical standard conformance has been certificated
- Environment: -10~ +50°C
- 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, batteries 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

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.
### Analog Wireless Intercom System

#### 1:4 Simultaneous Communication System

**Base Station**
- **YFF-4530**
  - **Specifications**
    - Structure: Rack mounting type
    - Power supply: AC 100 V
    - Number of calls: 10 simultaneous calls
    - Circuit configuration: U-shaped 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 certified
    - Environment: -10 ~ +50°C
    - 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 alkaline cell,2 Continuous use time is 23 hours
    - Call: Interactive simultaneous call
    - Antenna: Helical antennas or whip antennas for transmission/reception
    - Channel setting: Station selection is easy by quartz control PLL synthesizer system
    - Standards: Technical standard conformance has been certified
    - Environment: -10 ~ +50°C
    - Weight: Approx. 210g (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 alkaline cell,2 Continuous use time is 23 hours
    - Call: Interactive simultaneous call
    - Antenna: Helical antennas or whip antennas for transmission/reception
    - Channel setting: Station selection is easy by quartz control PLL synthesizer system
    - Standards: Technical standard conformance has been certified
    - Environment: -10 ~ +50°C
    - Weight: Approx. 210g (battery pack YBA-4120 included)
    - Dimensions: Width: 85.4mm; height: 82mm; depth: 22.5mm (not including protruding portions)

#### Antenna
- **CAW-4510**
- **Battery pack**
- **YBA-4120**
  - **Specifications**
    - Type: Dipole type
    - Applied frequency: 413~454MHz
    - Type: 0.5/2 half wavelength type
    - Junction type: M type
    - Impedance: 75Ω
    - Weight: Approx. 800g (attachment base included)
    - Dimensions: Width: 325mm; height: 287.5mm; depth: 91.5mm

#### Frequency within license
- **(B6) Downward transmission frequency**
  - | Channel | Frequency (MHz) |
  - |-------|----------------|
  - | 1 | 454.05000 |
  - | 2 | 454.05625 |
  - | 3 | 454.06250 |
  - | 4 | 454.06875 |
  - | 5 | 454.07500 |
  - | 6 | 454.08125 |
  - | 7 | 454.08750 |

- **(P6) Upward transmission frequency**
  - | Channel | Frequency (MHz) |
  - |-------|----------------|
  - | 1 | 413.70625 |
  - | 2 | 413.71250 |
  - | 3 | 413.71875 |
  - | 4 | 413.72500 |
  - | 5 | 413.73125 |
  - | 6 | 413.73750 |
  - | 7 | 413.74375 |

#### Electrical characteristics
- **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 range**
  - 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
- **Oscillation system**
  - Quartz control PLL synthesizer system
- **Frequency stability**
  - With in ±0.5ppm
- **Antenna power**
  - With in 1mW (+20 ~ -50%)
- **Strength intensity of spurious radiation**
  - 2.5 µW or less
- **Modulation scheme**
  - Direct frequency modulation
- **Voice frequency**
  - 3 kHz or less (30kHz ~ 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
- **Radio frequency output**
  - Continuous use time is 23 hours
  - Continuous use time is 20 hours
- **Audio frequency characteristics**
  - Within 300 Hz ~ 3kHz
- **Line input/output**
  - 0 dBm, balanced 600Ω (4W)
  - -20 dBu, unbalanced 220Ω
- **Microphone input**
  - -60 dBm, balanced 600Ω
  - -60 dBm, unbalanced 600Ω
- **Speaker output**
  - Inside: 1W (8Ω), 0 dBm
  - Balanced 600Ω (2W)
  - Unbalanced 600Ω
- **Microphone input sensitivity**
  - 57 dBm
  - 54 dBm
- **Receiver output**
  - 4nW or less
  - 2.5 μW or less
  - Strength intensity of spurious
  - -20 dBm, unbalanced 600Ω
- **Transmission power**
  - 2W
  - 2.5W
- **Reception system**
  - Double superheterodyne
- **Reception sensitivity**
  - 0 dBm or less at SINAD 12 dB
- **Squelch sensitivity**
  - Tone SQ: 0 dBm or less
  - Noise SQ: 0 dBm or less
- **Cabinet radiation**
  - 4nW or less
- **Use environment**
  - Temperature: -10 ~ +50°C
  - Humidity: with in 35 ~ 90%
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 (handheld type)**
- **TWO-T120 (two-piece type)**

**Transmitter**

- 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)

**RF**

- Operating frequency band: 1240.325 MHz to 1259.675 MHz (excluding 1251.700 MHz to 1253.300 MHz)
- Maximum number of channels used: 23 channels (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
- Dynamic range: 120 dB (A-weighted)
- Sampling frequency: 48 kHz
- Low-cut frequency: 60/80/100/125 Hz, 12 dB/oct
- Level setting:
  - Gain (3 dB step): -21 dB to +21 dB
  - LINE setting: +4 dB to -16 dB
  - MIC setting: -24 dB to -75 dB
- Sensitivity (1 dB step):
  - LINE setting: -4 dB to -16 dB
  - MIC setting: -24 dB to -75 dB
- Information compression: Linear PCM/ADPCM

**Indicators**

- Display: LCD
- Power supply state display
- Audio level display

**Power supply**

- Battery: Dedicated battery (lithium rechargeable battery) and AA cell battery (+0) (optional)
- Operating time (10 mW output, 25°C):
  - 6 hours or more (When the dedicated battery is used)
  - 1 hour or more (When AA cell batteries are used)
- Dimensions: 37 mm, height 185 mm (not including antennas)
- Weight: 223 g (Not including microphone capsule / including battery pack)
- Operating temperature/humidity: 0°C to +40°C, 20% to 90% (No condensation)
- Standard: Compliant with ARIB STD-T112

**Recommended Use**

- Digital wireless microphone (two-piece type)
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

**Audio**

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

**General**

- PC applications for the receiver are available
- Incorporating a LAN port for external remote control
- Digital audio output with a built-in SRC, available remote control synchronization and output compatible with 48 kHz/96 kHz external cable losses

**Operating frequency band**

- Digital output reference level -36 dBFS to -18 dBFS (2 dB step)
- Digital audio external synchronization signal WORD CLOCK 48 kHz or 96 kHz

**Digital output reference level**

- 48 dBm or more (35Ω load)

**Headphone output**

- Unbalance (600Ω load)
- powering level 50 mW or more (35Ω load)

**Headphone output power**

- 32-bit (8 ch) or 32-bit (32 ch)

**Digital output**

- AES/EBU (AES3 compliant)
- Sampling frequency: 48 kHz/96 kHz

**RF**

- Output frequency 198 MHz to 219 MHz
- Input frequency 1240 MHz to 1260 MHz

**Antenna mixing/distributing device**

- Antenna mixing/distributing device
- Two-D120
- Two-AY120
- Two-A120
- Two-BC120
- Two-RM120
- Two-BC120

**Antenna**

- Antenna with a built-in down converter
- Antenna (directional)
- Antenna (omnidirectional)
- Antenna input terminal (ANT A/B) BNC-J (50 Ω) × 8
- Antenna input connector (ANT A/B) BNC-J (50 Ω) × 4 (DC 12 V, local 45 MHz superimposition)

**Remote repeater**

- Remote repeater
- Two-RM120

**RF**

- Passing frequency 198 to 219 MHz
- Power supply DC 12 V (supplied by the receiver or the mixing/distributing device)

**Charger**

- Charger
- Two-BC120

**RF**

- Passing frequency 198 to 219 MHz
- Power supply DC +12 V (supplied by an AC adapter)

**Remote repeater**

- Remote repeater
- Two-RM120

**RF**

- Passing frequency 198 to 219 MHz
- Power supply DC 48V (supplied by PoE)
**OFDM Digital Wireless Microphone System**

**Example configuration of a general-purpose system**

- **Transmitter**
  - The maximum number of frequencies used is 23CH
  - Handheld-type transmitter
  - Two-piece-type transmitter

- **Operating terminal for transmitter setting**

- **Antenna with a built-in down converter**
  - Mixing/distributing device

- **2-ch receiver**
  - A1 B1 A2 B2

- **Monitoring PC**

- **BLE communication**

---

**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.
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.

**Example 1**

**Example 2**

**Transmitter**

**WTP-0705**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>RF Power Output</td>
<td>10mW</td>
</tr>
<tr>
<td>The Number of Channels</td>
<td>4 channels PLL type</td>
</tr>
<tr>
<td>Input</td>
<td>-60 ~ -20 / +4 dBm (600Ω)</td>
</tr>
<tr>
<td>Structure</td>
<td>Portable size</td>
</tr>
<tr>
<td>Available Time</td>
<td>More than 10 hours (1 AA alkaline)</td>
</tr>
<tr>
<td>Environment</td>
<td>-10℃~+60℃</td>
</tr>
<tr>
<td>Weight</td>
<td>100g (including battery)</td>
</tr>
<tr>
<td>Dimensions</td>
<td>Width: 60mm; Height: 60mm; Depth: 19mm (not including protruding portions)</td>
</tr>
</tbody>
</table>

**Receiver**

**WRP-0705**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Receive Sensitivity</td>
<td>Less than 2μV (SINAD12)</td>
</tr>
<tr>
<td>Type of Reception</td>
<td>Space Diversity</td>
</tr>
<tr>
<td>The Number of Channels</td>
<td>4 channels PLL type</td>
</tr>
<tr>
<td>Structure</td>
<td>Portable size</td>
</tr>
<tr>
<td>Power Source</td>
<td>DC10V (supplied by WFF-0711A)</td>
</tr>
<tr>
<td>Available Time</td>
<td>More than 18 hours (AA alkaline×2)</td>
</tr>
<tr>
<td>Environment</td>
<td>-10℃~+50℃</td>
</tr>
<tr>
<td>Weight</td>
<td>680g (including batteries)</td>
</tr>
<tr>
<td>Dimensions</td>
<td>Width: 360mm; Height: 77mm; Depth: 250mm (not including protruding portions)</td>
</tr>
</tbody>
</table>

**Transmitter mixer**

**WFF-0711A**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Input</td>
<td>LINE ×2 / MIC ×1 (600Ω)</td>
</tr>
<tr>
<td>Output</td>
<td>LINE ×1 (Transformer Balanced)</td>
</tr>
<tr>
<td>Channel Change</td>
<td>Remote Control, 4 Channels</td>
</tr>
<tr>
<td>Frequency Characteristic</td>
<td>100kHz ~ 10MHz</td>
</tr>
<tr>
<td>Environment</td>
<td>-10℃~+50℃</td>
</tr>
<tr>
<td>Power Source</td>
<td>AC100V / DC12V</td>
</tr>
<tr>
<td>Weight</td>
<td>2.5kg</td>
</tr>
<tr>
<td>Dimensions</td>
<td>Width: 260mm; Height: 77mm; Depth: 250mm (not including protruding portions)</td>
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</tbody>
</table>