English

TAMURA

Professional Studio Audio

Audio Equipment

&

Communication Systems

DIGITAL AUDIO MIXING CONSOLE

NTX800

NTX600

NT880

NT660

NT110

Audio Equipment
Communication Systems

NTX800

NTX600

Overview

The NTX Audio Mixing Console is a digital mixing console suitable for live broadcasting and recording consist mainly of the X CORE that performs audio routing and audio signal processing.
The X CORE has a MEDIA port that supports ST2110-30, AES67, and ST2022-7 standards, being compatible with IP-based next-generation broadcast systems.
Various adjustment knobs and switches arranged on the console, provides stress-free operability. It is a digital mixing console that implements various functions to enhance the efficiency of producing programs.

Features

Original technologies: Latest technologies

(1) X CORE and IO FRAME are connected by the IP-based protocol and are compatible with the AoIP standard.

(2) X CORE adopts high-performance audio signal processing with a unique hybrid system of DSP and FPGA. It provides reduction in rack space and low power consumption through high-density mounting.

(3) A newly developed signal processing algorithm and high-precision 64-bit floating point arithmetic improve the sound quality in the audio processing, such as equalizer and dynamics.

Operational safety: High safety

(1) Redundant power supply for all units in the system

(2) Redundant transmission paths within the system

(3) Redundancy of the IO FRAME interface card and the internal audio system, ensuring a high level of safety.

(4) Microphone input headroom 36 dB, enables resistancy from sudden excessive audio inputs.

(5) The audio signal processing and the audio routing are configured based on firmware, provides high stability.

Usability: Excellent operability

(1) Each channel has ten encoders (NTX800) or six encoders (NTX600). The equipment consists of two encoders, an HA/Trim control encoder and a Pan control encoder, and eight (NTX800) or four (NTX600) function switching encoders, providing intuitive operability like an analog console.

(2) Channel and bus parameters can also be manipulated in the center section. A touchscreen panel and encoders allow all parameters to be manipulated without leaving the center section. By using the Parameters Link Functions you can link the parameters figures between several faders temporarily

(3) It is equipped with a full-color LCD display which can clearly show channel names and corresponding parameter values. Excellent visibility is achieved by using both numerical values and visual graphs for setting the parameters.

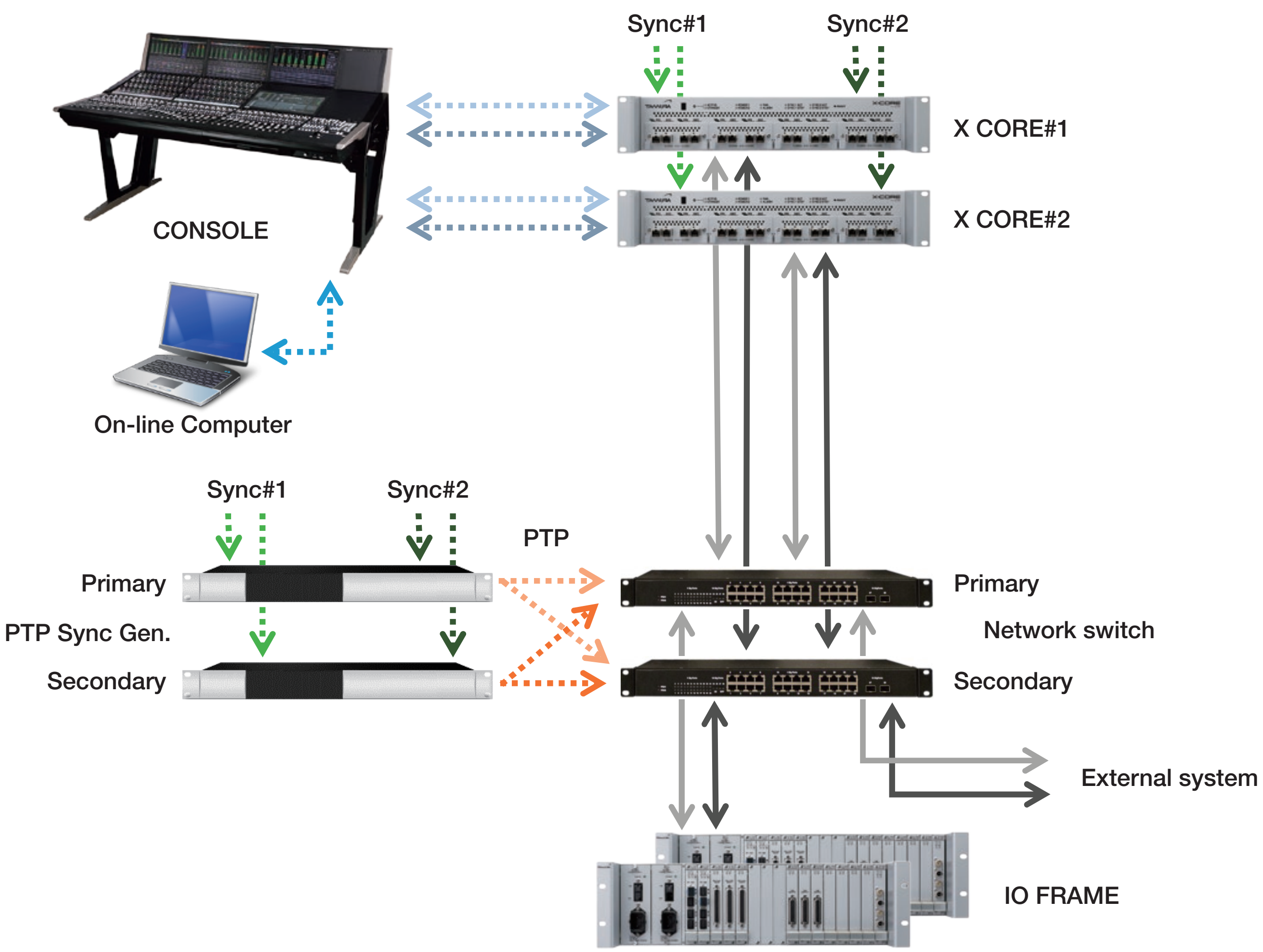
(4) Center section operations, such as input matrix switching, meter settings, and monitor control, are also possible in the input section. When a large-scale system is being built or multiple operators are working, the necessary operations can be performed in every part of the console.

(5) It supports a free layout allowing free assignment of input channels and various master faders. Free layout is possible regardless of channel types.

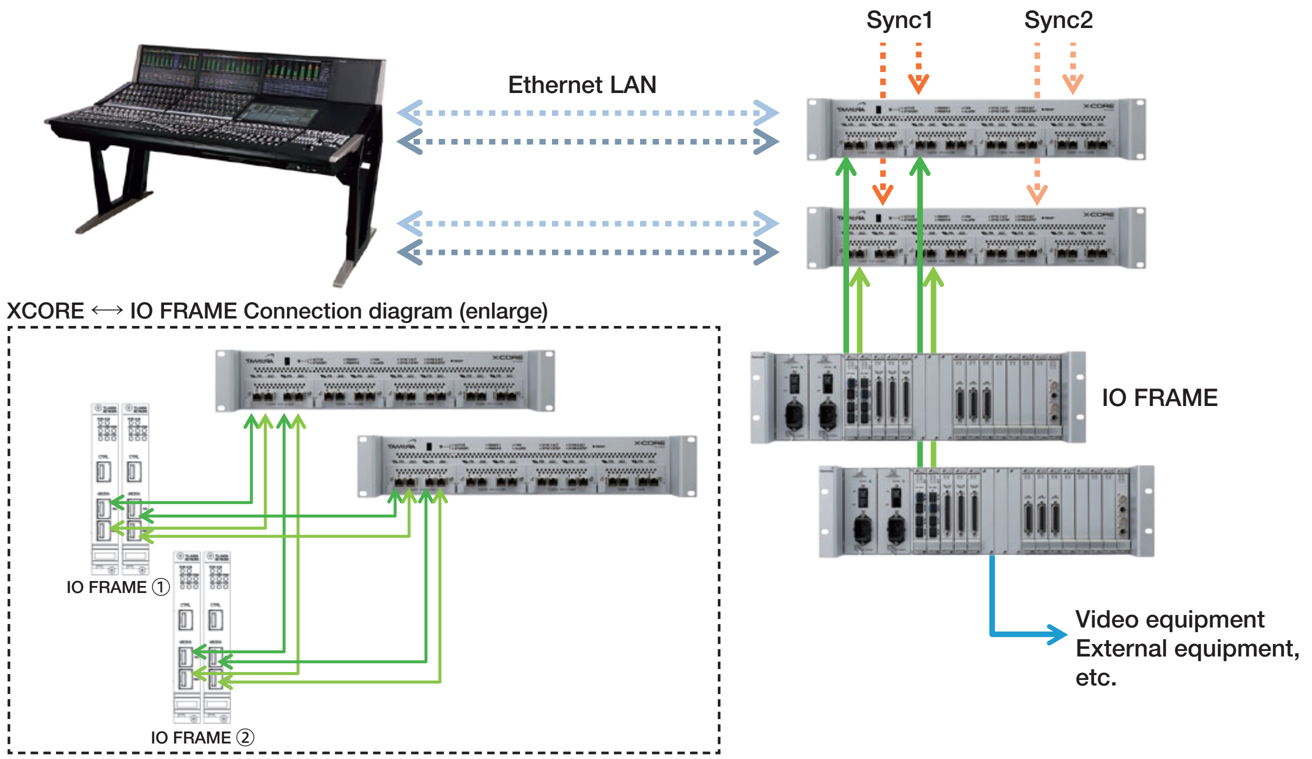
(6) With BH(Bbehind) HA Trim/Fader feature you can check and control HA gain and Trim which is assigned to layer B

CONNECT IO DIAGRAM

PTP connection



Packet PLL connection



SPEC

Supply voltage	AC100 - 240V 50/60Hz
Maximum fader number	120ch
Bank/layer	6 bank / 2 layer
Number of fader groups	64 groups
DSP maximum processing number	1024 ch
DSP maximum processing bit number	64 Bit
Sampling frequency	48 kHz / 96 kHz
Sync signal input	AES3id/WORD VIDEO (NTSC/PAL) PTPv2 (when ST-2110 is connected)
Transmission frequency range	20 - 20kHz (Fs=48kHz) 20 - 40kHz (Fs=96kHz)
X CORE lines	Maximum 4,096 ch (input) Maximum 4,096 ch (output)

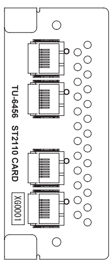
Transmission frequency range	20 - 20 kHz (Fs = 48 kHz) 20 - 40 kHz (Fs = 96 kHz)
Digital audio signal	AES3id compliant 1Vp-p (75Ω unbalanced) Input: 16 to 24 Bit Output: 24 Bit Input/output level: -18/-20 dBFs
MIC input level	-64 to +10 dBu
HA headroom	30/36 dB
LINE input/output level	0/+4 dBu

Audio Processing engine

■ TS-10322 X core

Routing and Audio Processing engine for NTX mixing console
* TU-6469 DSP Board : Audio Processing board which is bulid inside X core

Maximum Input Channels When 4 ST2110 Cards installed	fs48kHz:4,096ch fs96kHz:2,048ch
Maximum Output Channels When 4 ST2110 Cards installed	fs48kHz:4,096ch fs96kHz:2,048ch
Total DSP Channels When 4 DSP Board installed	fs48kHz:1,024ch fs96kHz:512ch
Numbers of Mix Busses	fs48kHz:256ch fs96kHz:128ch
Numbers of DSP Board Slot	4
Numbers of IO Card Slot	4
Numbers of X core Needed in 1 System	2
Accessories	AC cable×2 (2m)
Power Requirements	AC100 - 240V 50/60Hz
Maximum Power Consumption	265VA
Operating temperature range	+5 ～ +40℃
Operating humidity range	20 ～ 80%RH
Dimensions	430 x 460 x 88.1mm
Weight	9.2kg

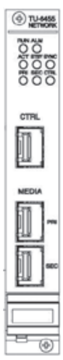


■ TU-6456 ST2110 CARD

Audio Input/Output Cards for X core. Maximum 4 Cards could be installed in each X core.

Number of MEDIA Ports	2 x Primary & Secondary
Number of Audio Channels for Ports	fs48kHz : 512ch fs96kHz : 256ch
Number of Streams for each Ports	64
AoIP Protocol	SMPTE ST 2110-30 SMPTE ST 2022-7 AES67-2018
Synchronization	PTP V2/ Packet Synchronization
MEDIA Port communication speed	1Gbps
Compatible SFP module	1000BASE-T/RJ45 1000BASE-LX/SMF
Accessories	RJ45 SFP module (FCLF8522P2BTL equivalent)×4
Weight	0.2kg

NETWORK Card

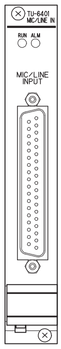


■ TU-6455 NETWORK Card

Network interface card for ST2110

Number of MEDIA Ports	SFP Port 1 x Primary & 1 x Secondary
Number of Control Ports	1 x CTRL Port
Number of MEDIA port channels	48 kHz: 512 ch 96 kHz: 256 ch
Number of MEDIA port streams	7
AoIP Protocol	SMPTE ST 2110-30 SMPTE ST 2022-7 AES67-2018
Synchronization	PTP V2/Packet Synchronization
MEDIA/CTRL Port communication speed	1 Gbps
Compatible SFP module	1000BASE-T/RJ45 1000BASE-LX/SMF

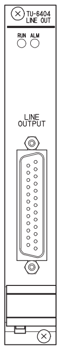
Option card Common to NT and NTX



■ 8ch DSUB MIC/LINE IN Card

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

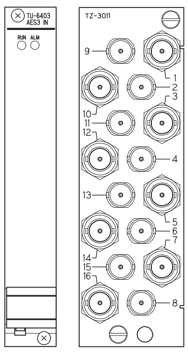
Number of Occupied slots	1 slot
Mic/Line input	balanced type
Number of Channels	8ch
[Mic input] Input level	-64dBu ~ +10dBu
[Mic input] Input inpedance	4kΩ or more
[Line input] Input level	-12~+12dBu (0.1dB step select)
[Line input] Input inpedance	600 / 10kΩ or more



■ 8ch DSUB LINE OUT Card

Audio interface card of analog 8ch output.

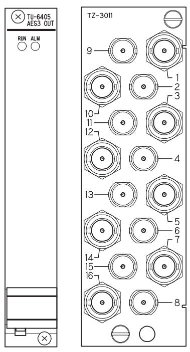
Number of Occupied slots	1 slot
Line output	balanced type
Number of Channels	8ch
Output level	-12 ~ +12dBu (0.1dB step select)
Output inpedance	55Ω



■ 8ch BNC AES3 IN Card

Audio interface card of 8 channel AES3 input. SRC ON/OFF setting 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.

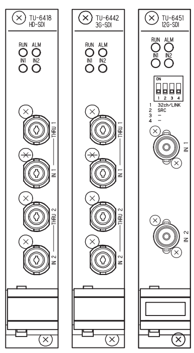
Number of Occupied slots	1 slot
Format	AES-3id
Number of Channels	8ch AES3
Input inpedance	75Ω unbalanced type
Input sampling frequency (SRC ON)	32~100kHz
Input sampling frequency (SRC OFF)	48 / 96kHz (Synchronized with the system clock)
Number of Input bits	16~24bit



■ 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.

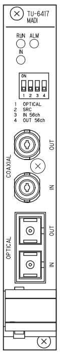
Number of Occupied slots	1 slot
Format	AES-3id
Number of Channels	8ch AES3
Output inpedance	75Ω unbalanced type
Output signal level	1 Vp-p
Output sampling frequency	48 / 96kHz (Synchronized with the system clock)
Number of output bits	24bit



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

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

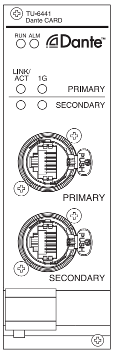
	HD-SDI Card TU-6418	3G-SDI Card TU-6442	12G-SDI Card
Number of Occupied slots	1 slot	1 slot	1 slot
Supported SDI formats	720p 50/59.94/60Hz 1035i 59.94/60Hz 1080i 50/59.94/60Hz 1080p 23.98/24/25/29.97/30Hz 1080psF 23.98/24Hz		2160/59.94p
		1080p 50/59.94/60Hz 1080psF 25/29.97/30Hz	
Embedded audio standard	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, 64ch/56ch IN/OUT setting 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 for Dante network.
Maximum 64 input channels and 64 output channels.

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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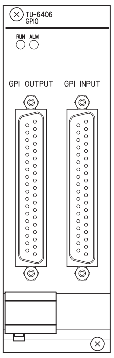
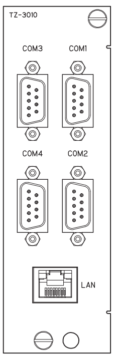




■ GPIO Card

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

[GPI]

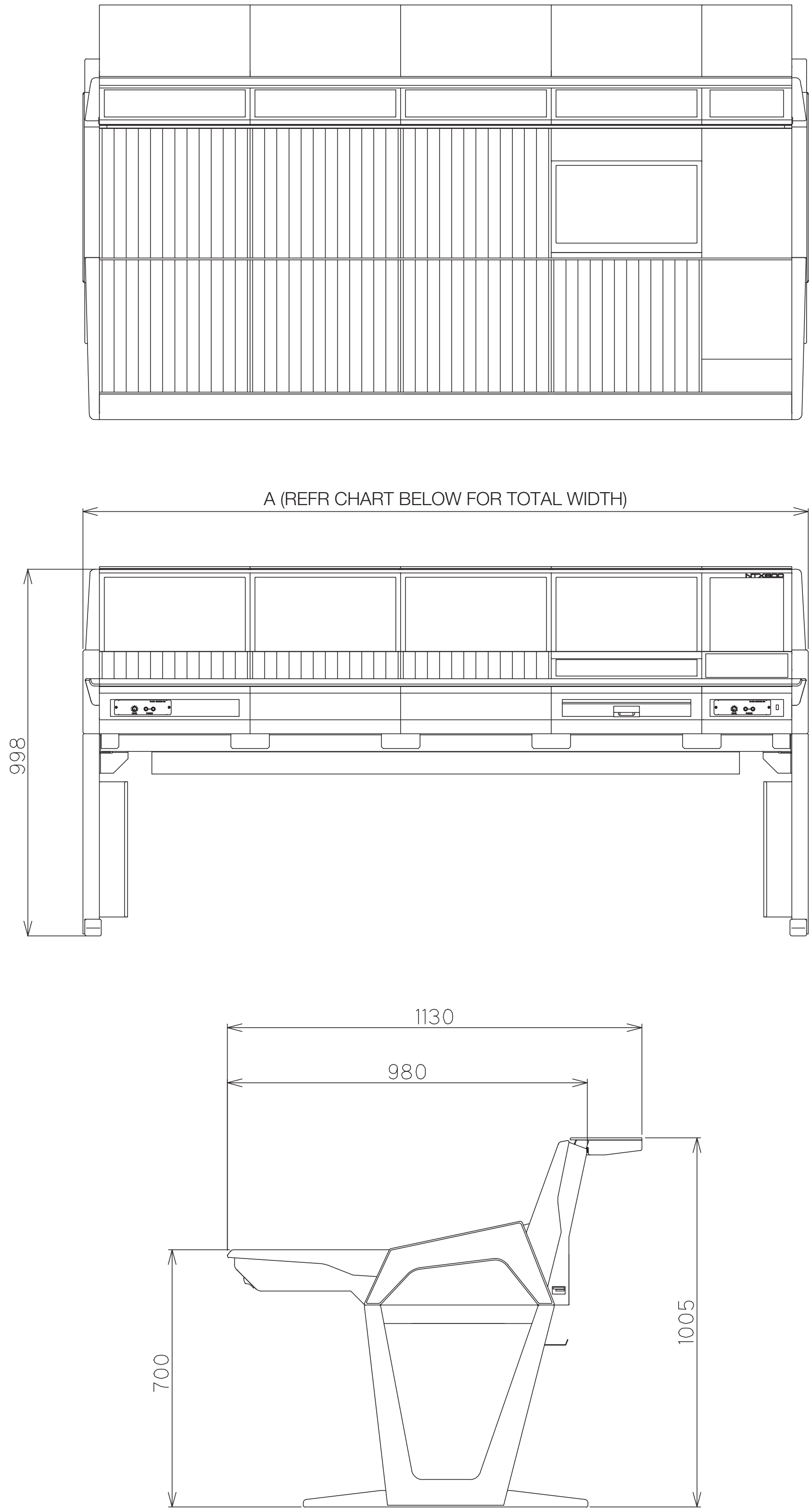
Function	Description
Link Function FU/BT	When On, the specified FU function is set to On or BT is On
Link Function Remote	When On, the specified remote function is set to On
Link Function AVL	When On, AVL function is set to On
System Tally 1	When On, indicator LAMP1 for OSC and TB prohibit control is 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

Dimensions

NTX800 CONSOLE



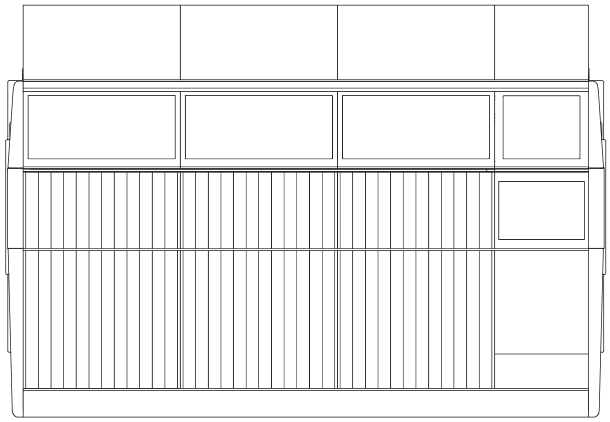
<A: TOTAL WIDTH ACCORDING TO NUMBER OF BAY>

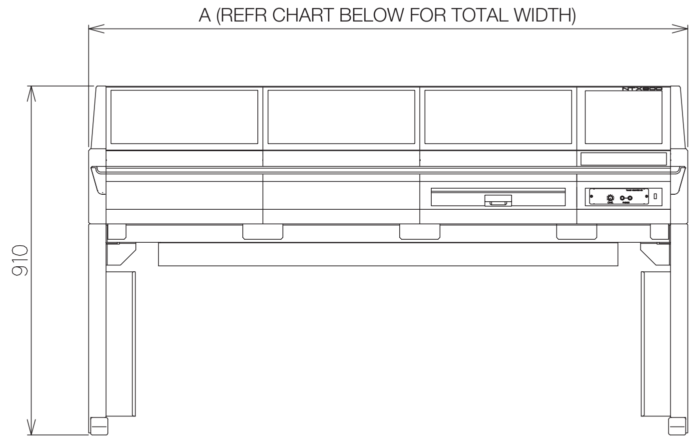
1BAY	2BAY	3BAY	4BAY	5BAY	6BAY	7BAY	8BAY	9BAY	10BAY
745	1155	1565	1975	2385	2795	3205	3615	4025	4435

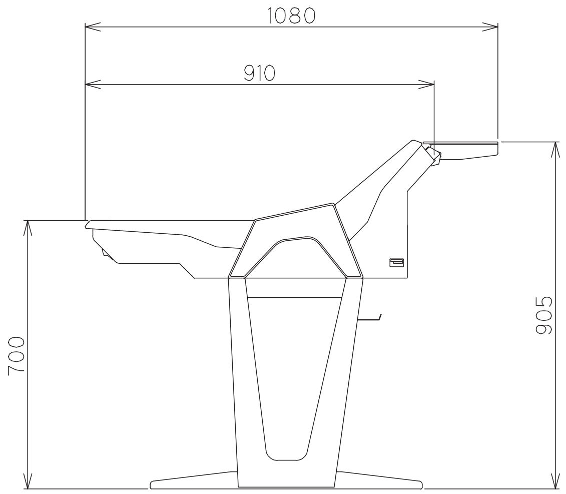
(mm)

Dimensions

NTX600 CONSOLE







<A: TOTAL WIDTH ACCORDING TO NUMBER OF BAY>

1BAY	2BAY	3BAY	4BAY	5BAY	6BAY
745	1155	1565	1975	2385	2795

(mm)

NT880

with Tamura Resource Network Technology

NT660

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

Scalability

› 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)

› Multiple Operation/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

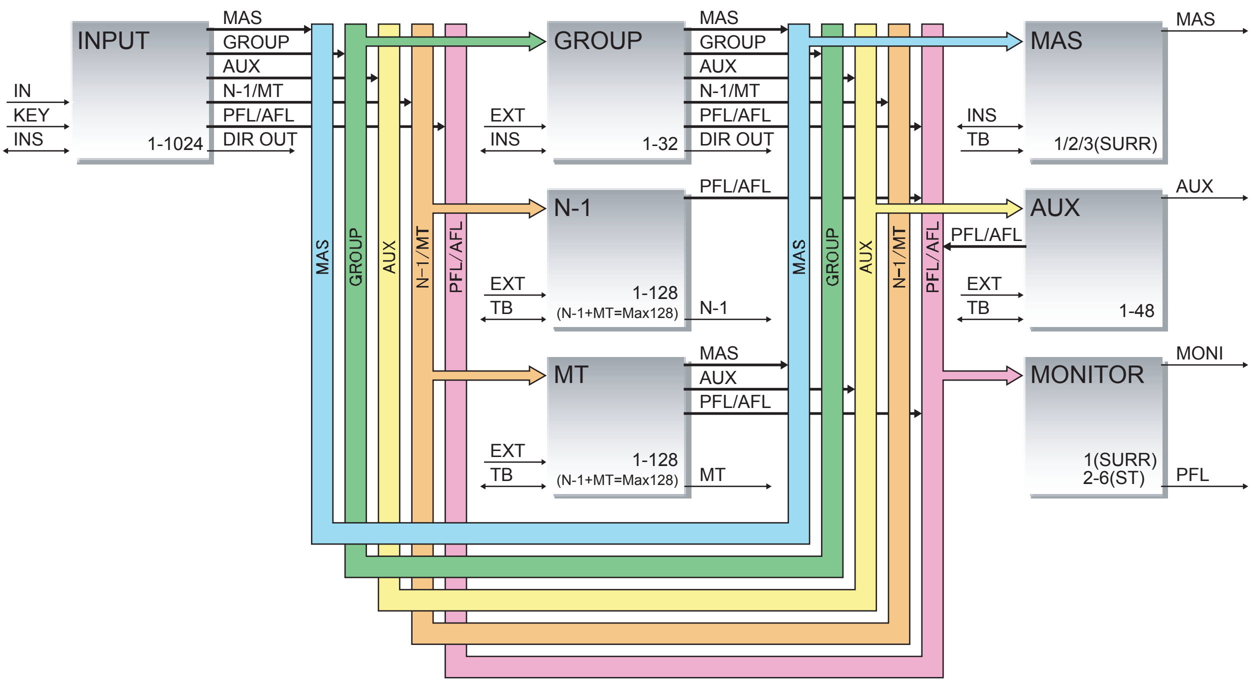
Audio channel (Fs=48kHz)

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

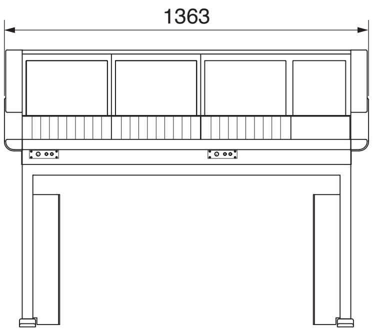
HA Gain	+10dBu~-64dBu
Trim	+24dB~-24dB
Delay	5000ms or more
Filter	Filter1 (HPF/Notch) Filter2 (LPF/Notch)
Equalizer	4Band (Support for all frequency bands)
Dynamics	Compressor 2 channels Gate/Expander 1channel

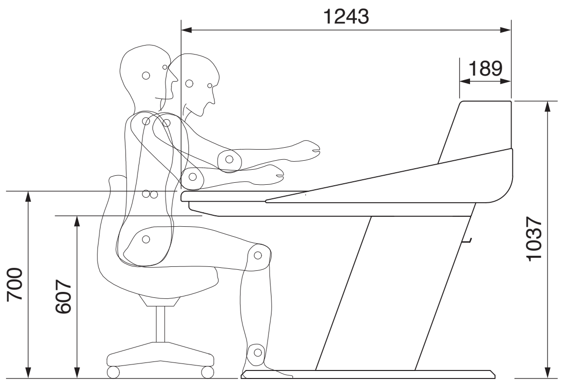
Audio block diagram



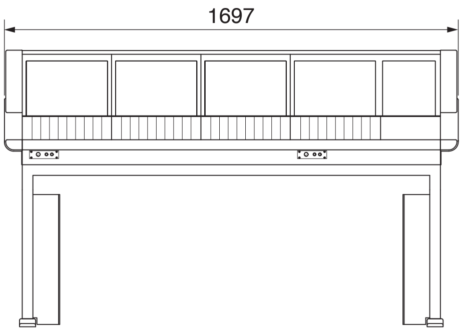
Dimension

30 Fader

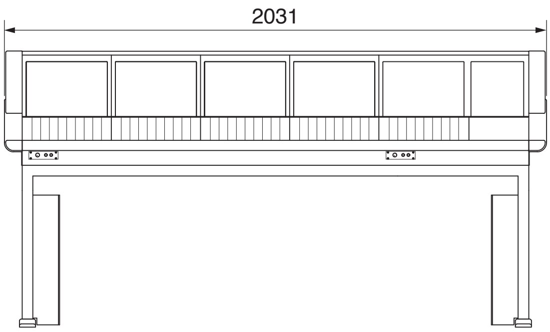




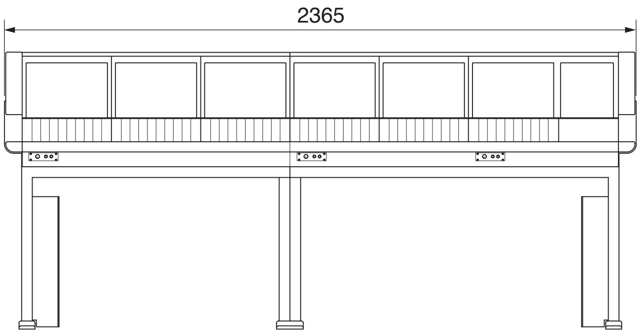
40 Fader



50 Fader



60 Fader



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

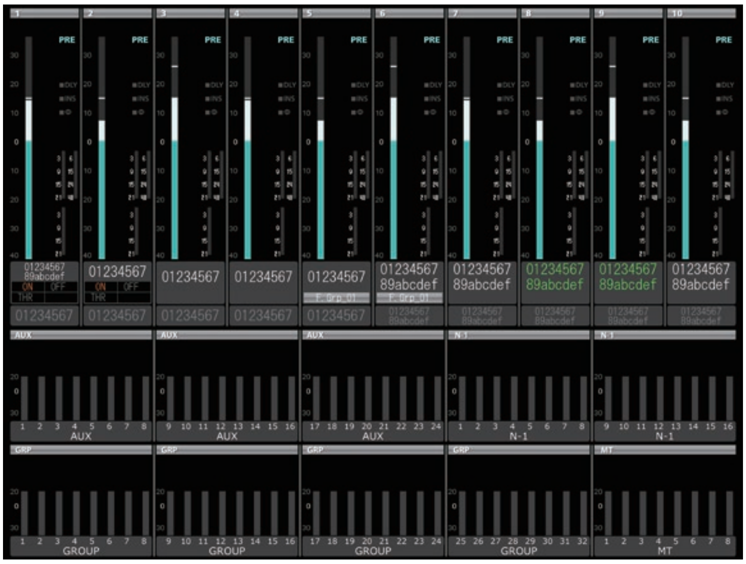
You can use the same sound processing parameters as those of the higher-grade model NT880.
Two compressors are used for each individual channel, and algorithms for full-four-band EQ and the like are exactly the same.
Also, the equipment has a delay compensation function for multistage bus assignment,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

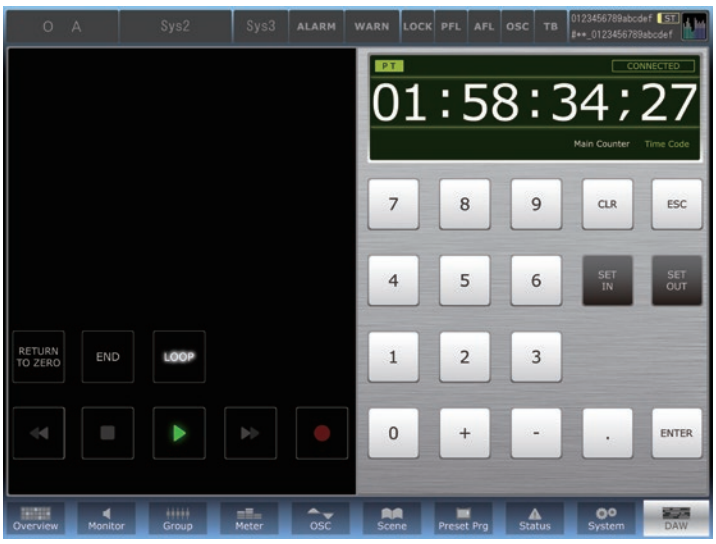
User Level Setting

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 Output Busses

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)

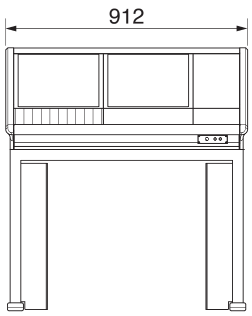
Master Bus	Maximum 24 buses (3 surround)
Group Bus	Maximum 32 buses
Aux Bus	Maximum 48 buses
N-1 / MT Bus	Maximum 128 buses
AFL	1 surround
AFL / PFL	3 stereo
PFL	1 stereo
Main Monitor	1 surround+stereo
Sub Monitor	3 channels (Stereo)

Audio control parameters

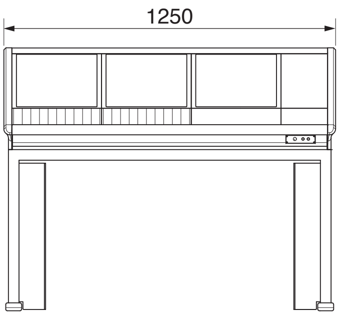
HA Gain	+10dBu~-64dBu
Trim	+24dB~-24dB
Delay	5000ms or more
Filter	Filter1 (HPF/Notch) Filter2 (LPF/Notch)
Equalizer	4Band (Support for all frequency bands)
Dynamics	Compressor 2 channels Gate/Expander 1channel

Dimension

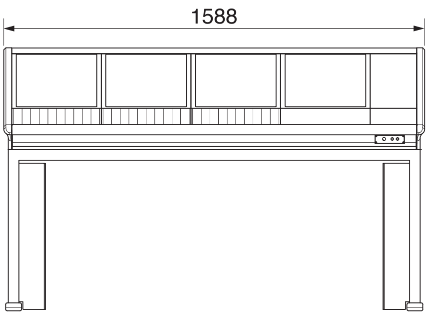
20 Fader



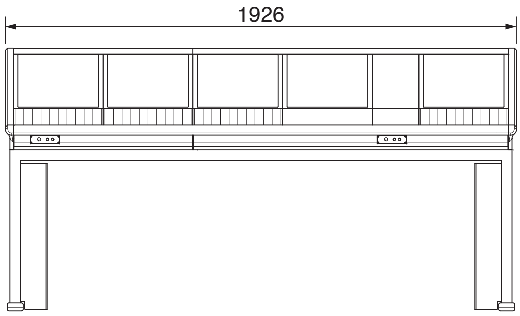
30 Fader

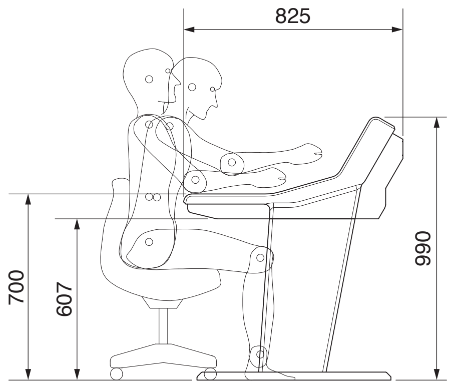


40 Fader



50 Fader





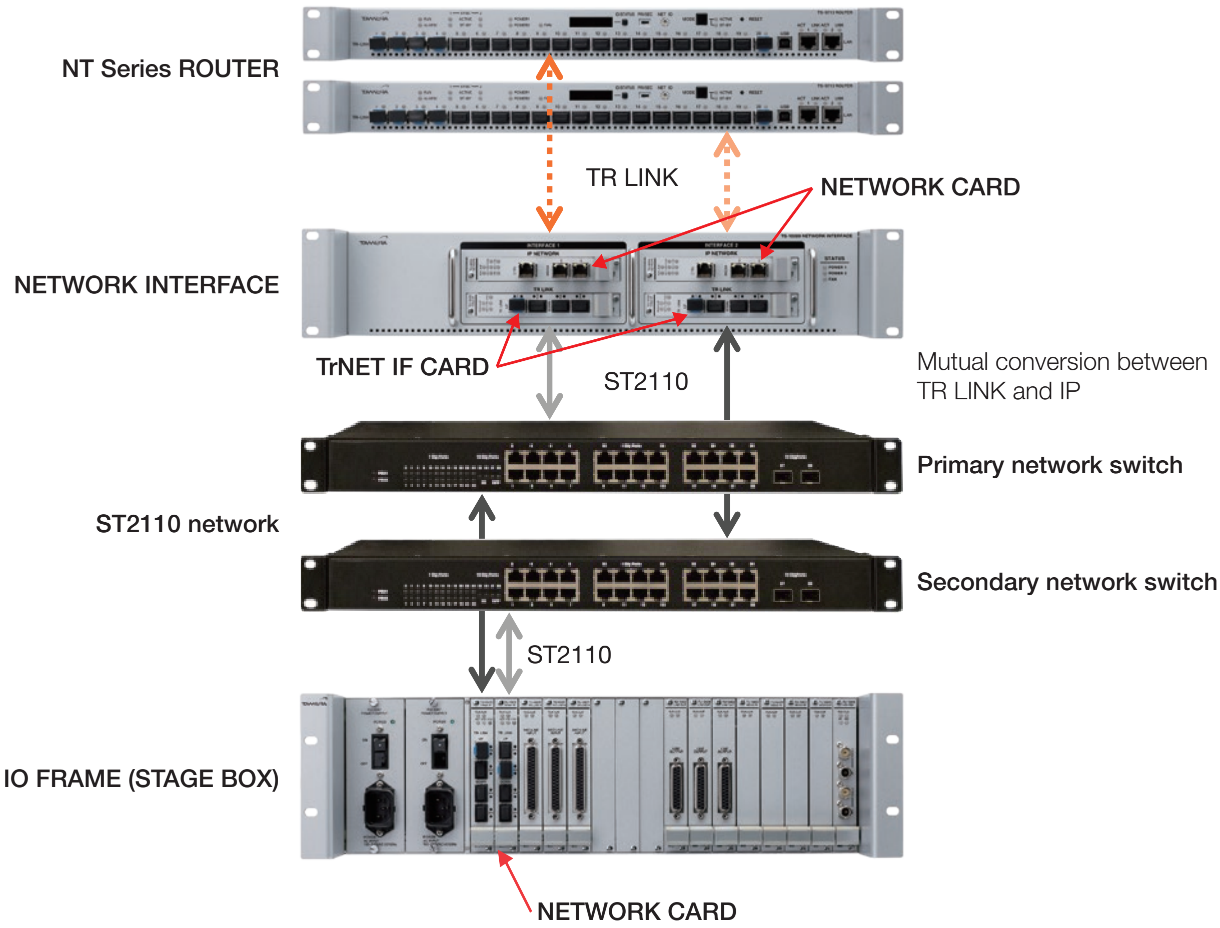
NETWORK INTERFACE

Overview

The device converts TR LINK signals of NT Series (NT880/NT660/NT900/NT900C) into IP signals compatible with SMPTE ST2110.

Conversion system between NT Series TR LINK and ST2110

The system converts NT Series TR LINK signals to ST2110 signals via NETWORK INTERFACE.
The connection of the IO FRAME to the ST2110 network enables transmission of audio signals to and from the ROUTER and control of the HA GAIN/P48 settings of MIC/LINE IN CARD.



* NETWORK INTERFACE converts media independently for INTERFACE1 (left) and INTERFACE2 (right).

Specifications

Item	Specifications	
Power supply	AC100 - 240V 50/60Hz × 2	
Power consumption	200W or less	
Interface	TrNET IF CARD	(maximum 2)
	NETWORK CARD	(maximum 2)
Protocol	TrNET IF CARD	Original protocol of Tamura
	NETWORK CARD	SMPTE ST2110 SMPTE ST2022-7

Specifications

AUDIO signal

(1) AUDIO signal transmission
It outputs 512ch FS48kHz audio signals inputted from the TR LINK to the IP network.
It outputs 512ch FS48kHz audio signals inputted from the IP network to the TR LINK.
* The channel rows are the same for TR LINK and IP (ch1 of TR LINK becomes ch1 of IP).
* There is no audio MTX function inside NETWORK INTERFACE.

(2) Bit number conversion
It performs conversion between TR LINK 32bit floating point and IP 24bit fixed point.

(3) Audio level offset
It is equipped with a voice level offset function used for the purpose of securing the digital reference level (18 dB FS or 20 dB FS) and the microphone headroom (30 dB or 36 dB). (Level setting is performed with WEB UI.)

Control signal

Control signals between TR LINK and IP are either processed within NETWORK INTERFACE or converted into each other before transmission.

Voice synchronization

TrNET IF CARD operates in sync with the audio signal input from TR LINK.
NETWORK CARD operates by synchronizing audio with the PTP signal input from the IP network.

* If TR LINK (ROUTER) and the IP network (PTP) are not synchronized, noise may appear in the audio.
* Each CARD installed in NETWORK INTERFACE does not have the function of built-in sample rate converter. The synchronized state should always be maintained.

The specifications and standards that NETWORK CARD complies with are shown in the table below.

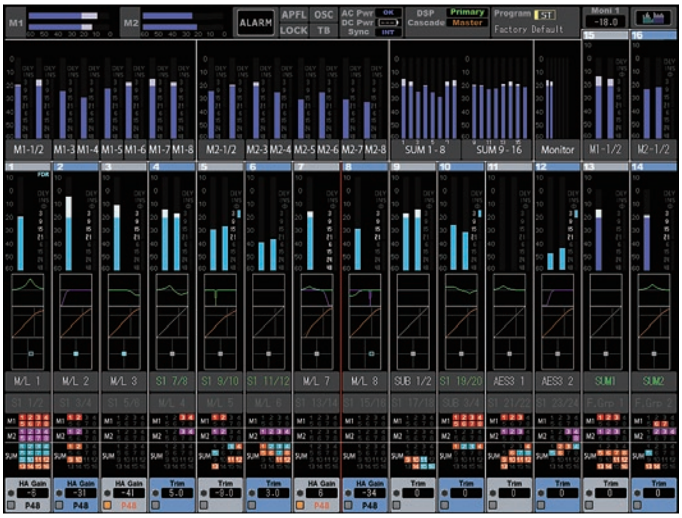
No	Name of standard	Overview
1	AES67 2018	AES standard for audio applications of networks - High performance streaming audio over IP interoperability
2	SMPTE ST 2110 10	Professional Media Over Managed IP Networks : System Timing and Definitions
3	SMPTE ST 2110 30	Professional Media Over Managed IP Networks : PCM Digital Audio
4	SMPTE ST 2022 7	Seamless Protection Switching of SMPTE ST 2022 IP Datagrams
5	NMOS IS 04	Discovery and Registration Specification
6	NMOS IS 05	NMOS Device Connection Management Specification

NT110

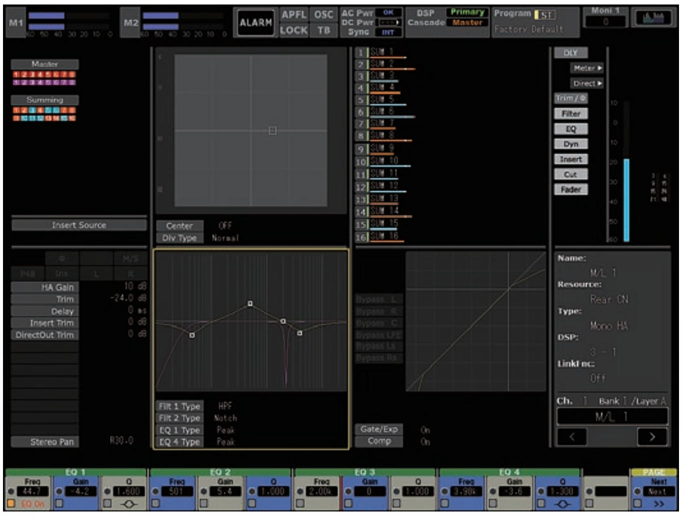
Digital Portable Audio Mixer

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

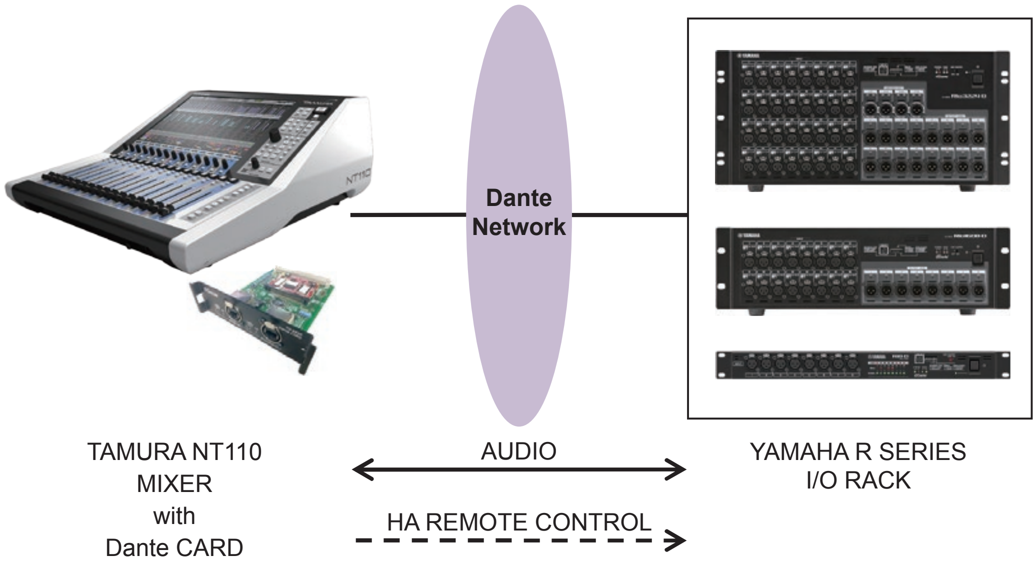
Rio Remote

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 OPTION 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

Overview

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℃
■ 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

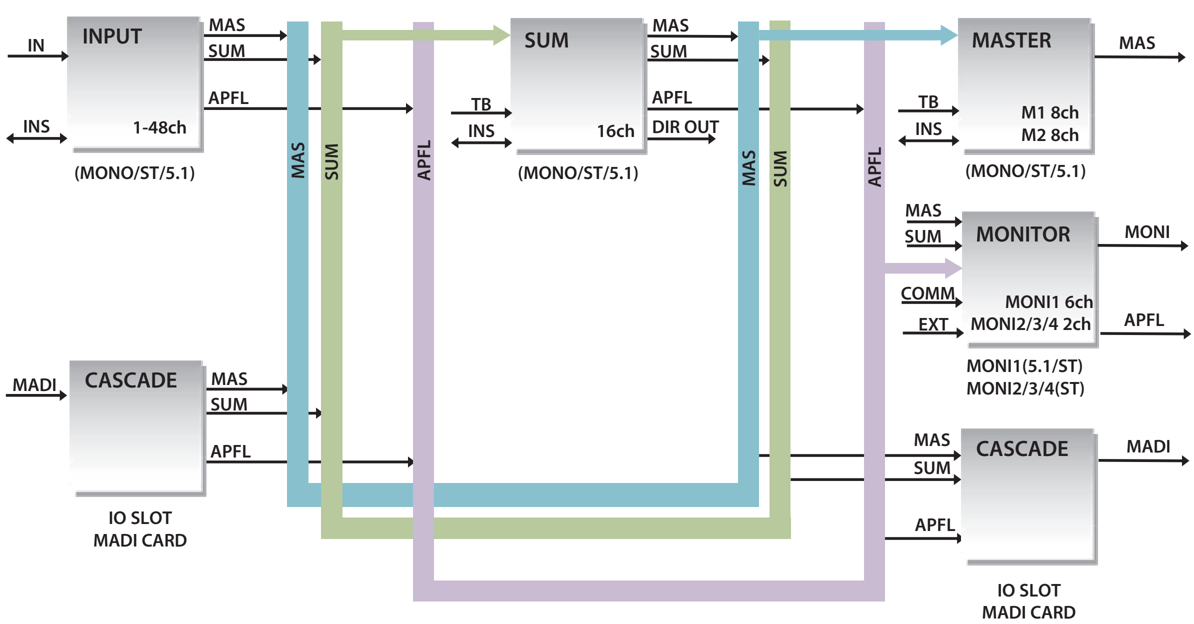
■ DSP CARD (BACK UP REDUNDANT DSP)

■ 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

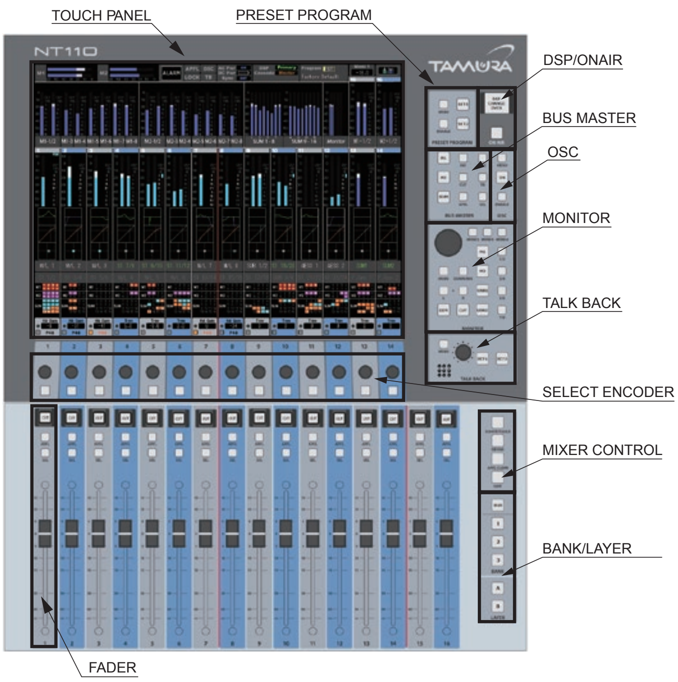
■ Storage case

Audio block diagram

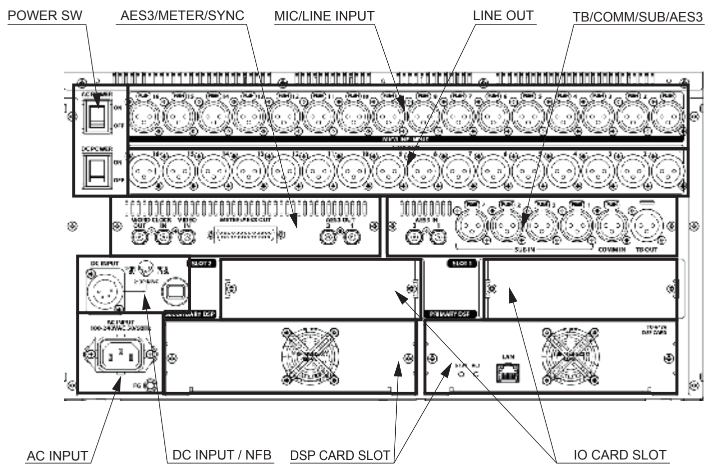


Control Panel Description

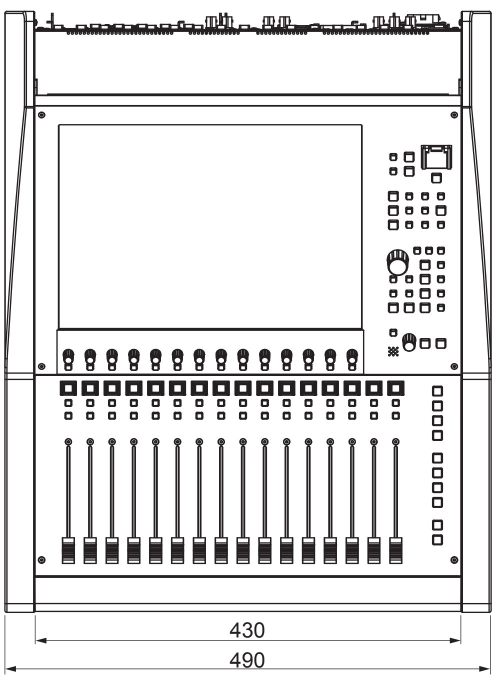
■ Front panel

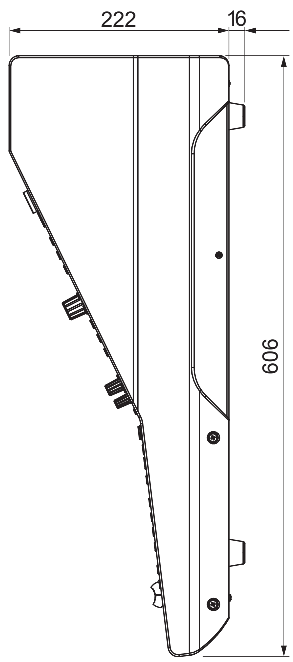


■ Rear panel



Dimensions

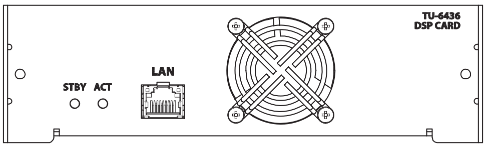




Option card

■ DSP Card

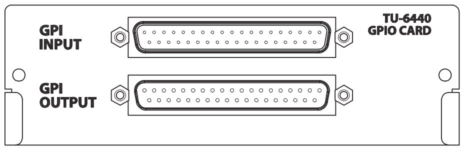
This card is a card with built in audio signal processing, audio routing and control functions.
For redundant system implement 2 DSP card to NT110



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

■ GPIO Card

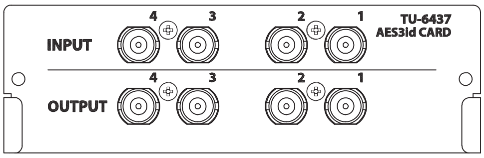
Interface card for 16-input / 16-output of general-purpose control signals.



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

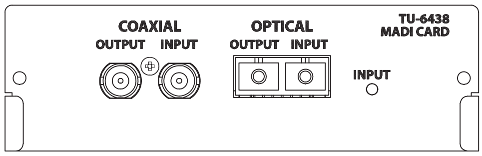
Audio interface card of AES3 input / AES3 output.



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

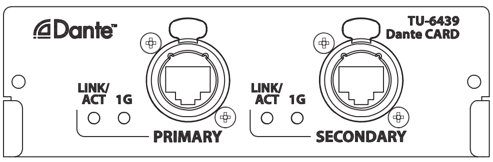
Audio interface card of MADI 64ch input / 64ch output



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 for Dante network.
It is possible to maximum 64 channels input, 64 channels output.

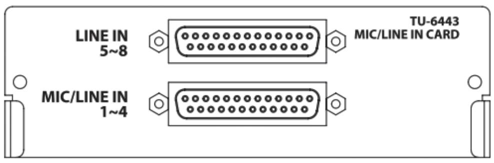


Sampling frequency (Fs)	48kHz / 96kHz
Input / Output (Fs=48kHz)	Max 64 input, 64 output
Input / Output (Fs=96kHz)	Max 32 input, 32 output
Transmission Protocol	Dante
Dante Connector	RJ-45 type /Neutrik etherconConnector
Dimensions	129(W)x40(H)x152(D)mm
Weight	150g

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

■ MIC/LINE IN CARD

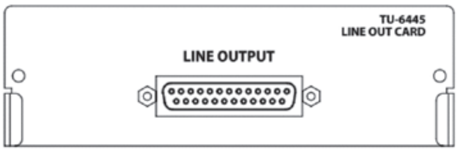
This card is audio interface card that inputs microphone level and line level analog audio signals.



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

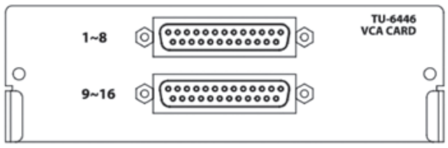
This card is audio interface card that outputs line level analog audio signals.



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)

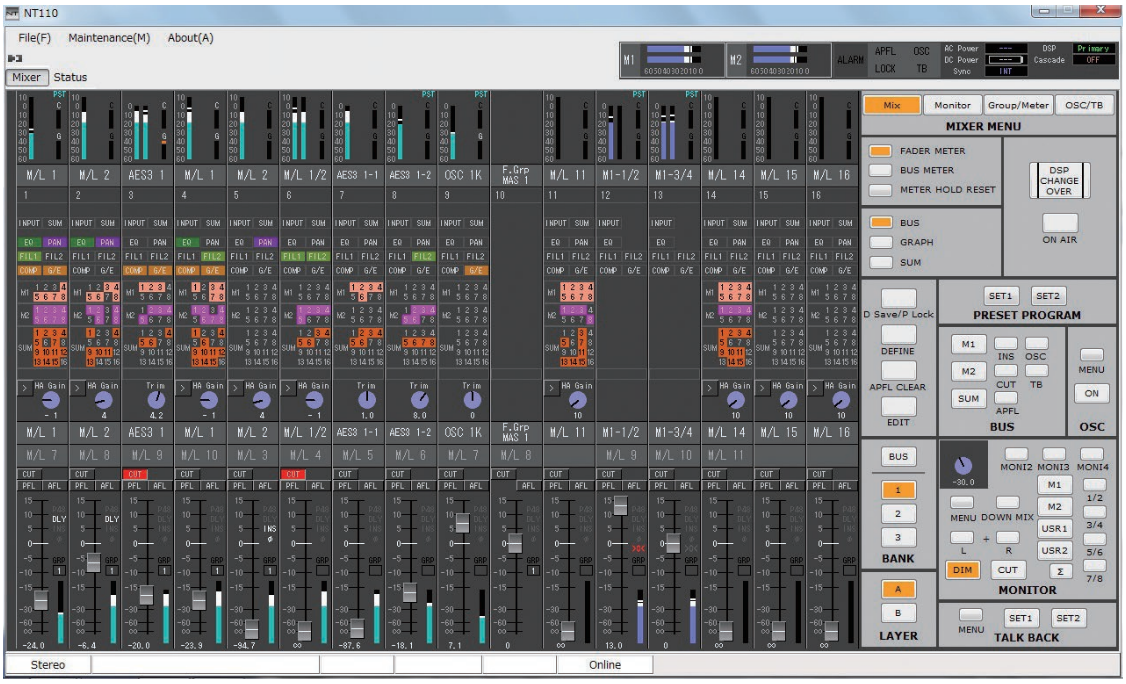
Interface card for input of VCA control signals.



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 Outside broadcasting, live shows, 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

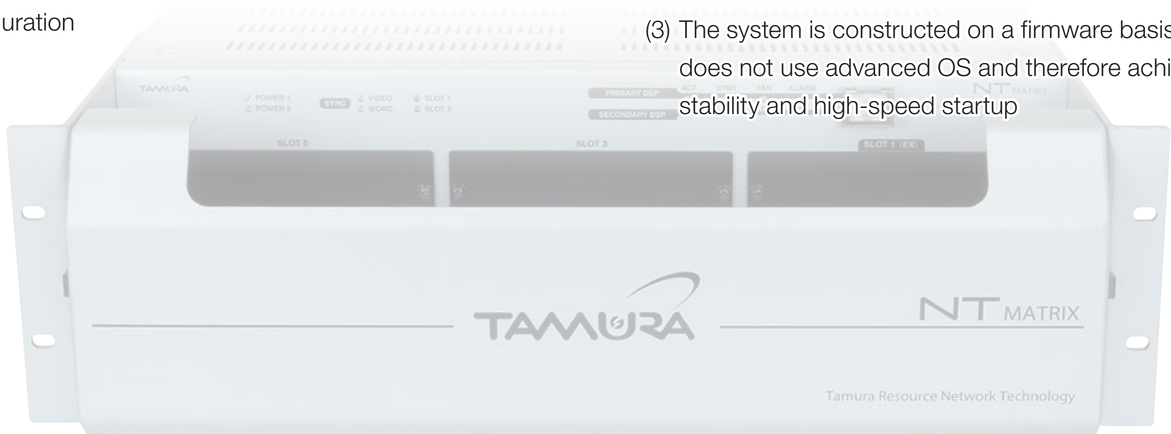
(2) Selection of the signal processing function by the DSP configuration

Operational safety - high safety

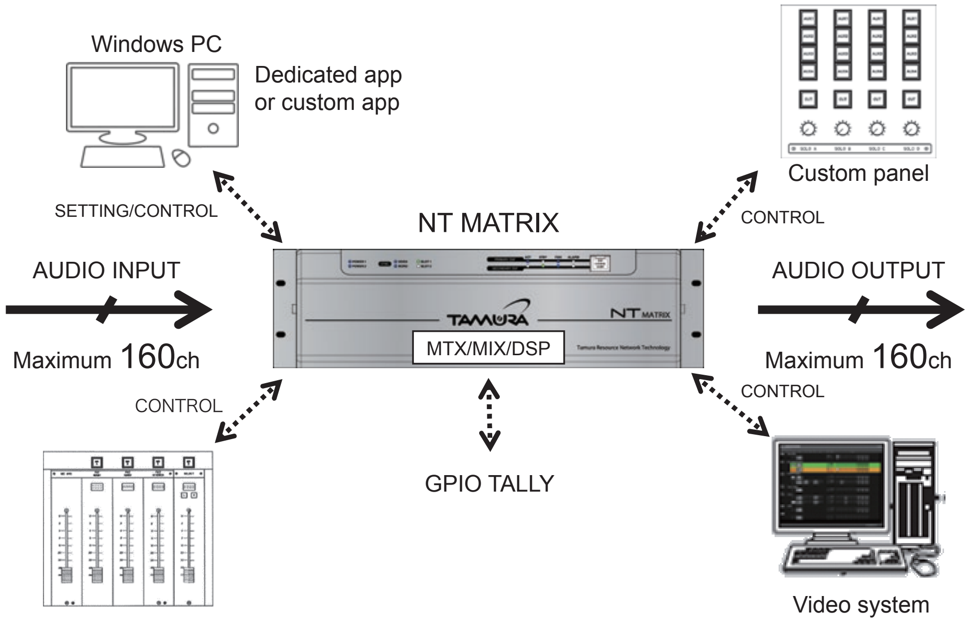
(1) Redundant configuration with two AC inputs for power supply

(2) DSP CARDs equipped with audio signal processor and controller can be redundantly configured to support various applications requiring high reliability (optional)

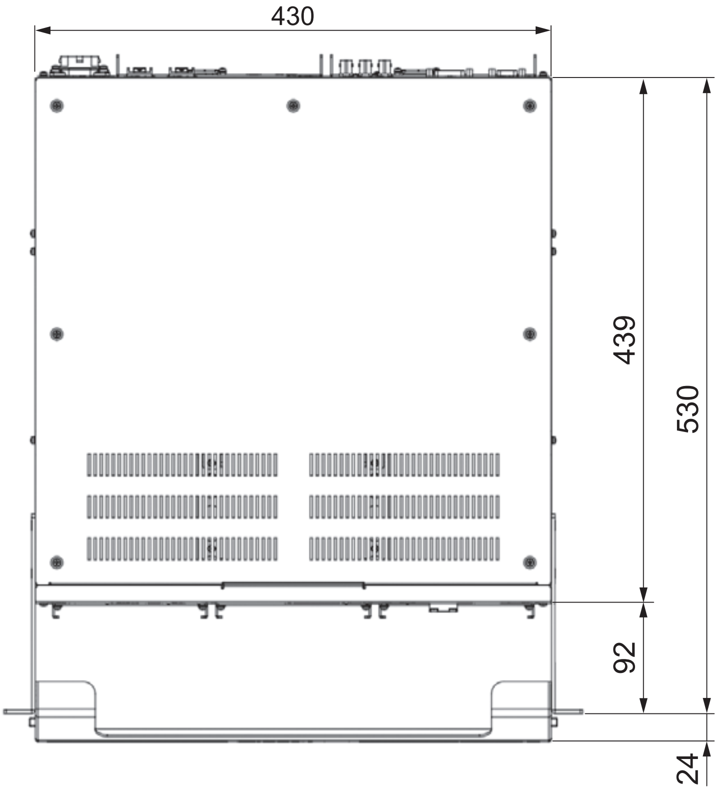
(3) The system is constructed on a firmware basis that does not use advanced OS and therefore achieves high stability and high-speed startup

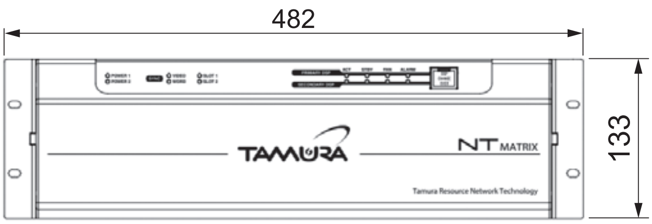


NT MATRIX System



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

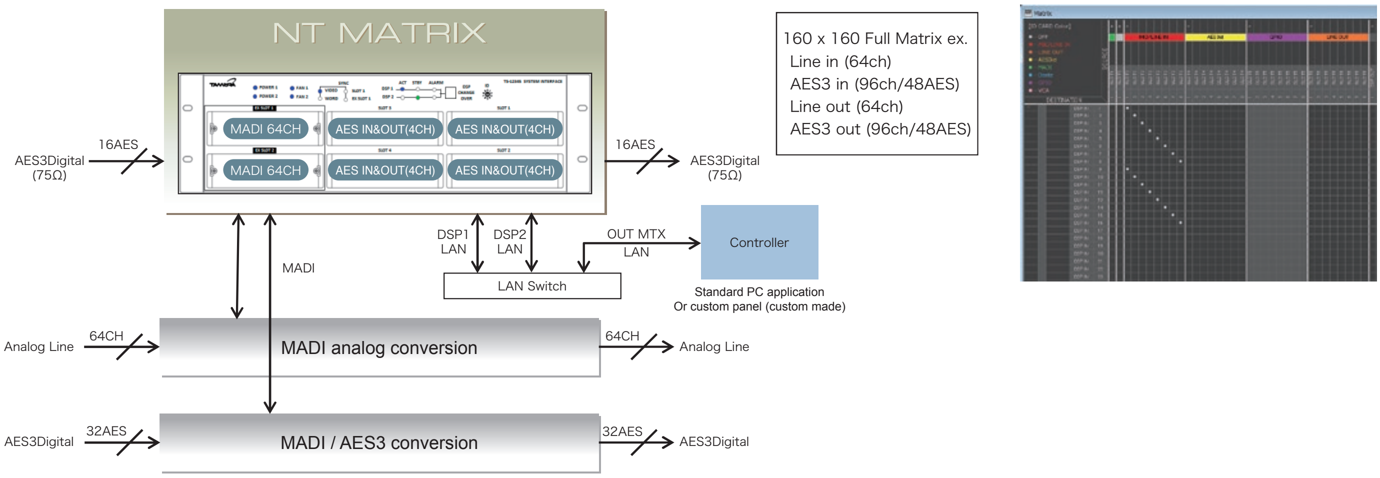
Items	Specification
AUDIO ROUTER	160ch x 160ch
DSP	32in x 32out DSP x 6
DSP FUNCTION	32in x 32out Mix Matrix or Filter/Limitter , AUD , Internal OSC
CONTROL PORT	LAN/RS422SERIAL/GPIO/VCA
SYNC SIGNAL	WORD CLOCK/VIDEO
POWER SUPPLY	AC100-240V 50/60Hz
OPERATION TEMPERATURE	-10 ~ 40 °C
EXTERNAL DIMENSIONS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

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

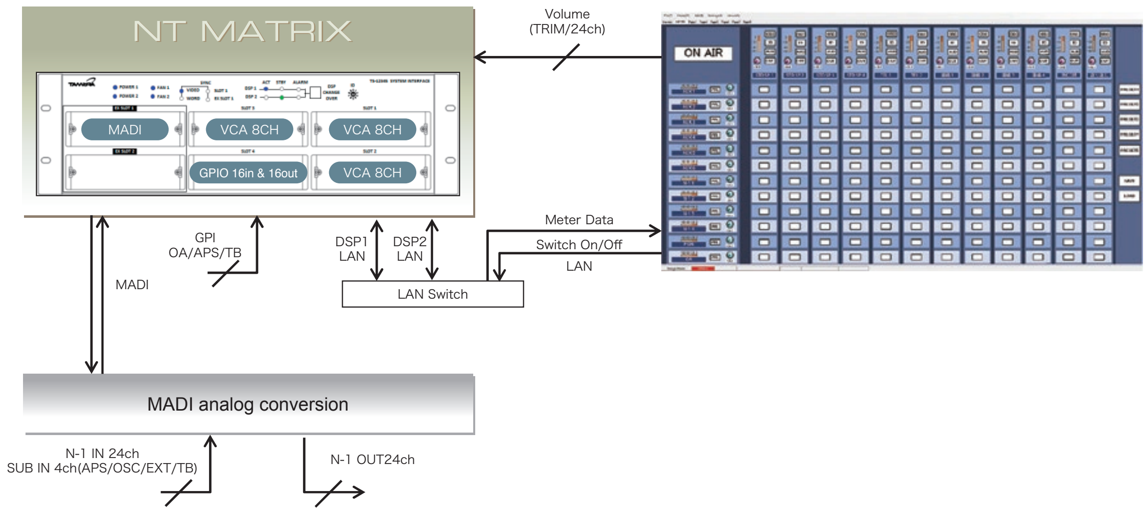
Items	Specification
DSP CARD	Redundancy DSP CARD
MIC/LINE INPUT CARD	MIC INPUT 4ch + LINE INPUT 4ch
LINE OUTPUT CARD	LINE OUTPUT 8ch
AES3id CARD	AES3id INPUT 4ch + AES3 id OUTPUT 4ch
MADI CARD	MADI INPUT 1ch + MADI OUTPUT 1ch(OPTICAL & COAXIAL)
Dante CARD	Dante 1ch (Primary & Secondary)
GPIO CARD	GPI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

Example of application

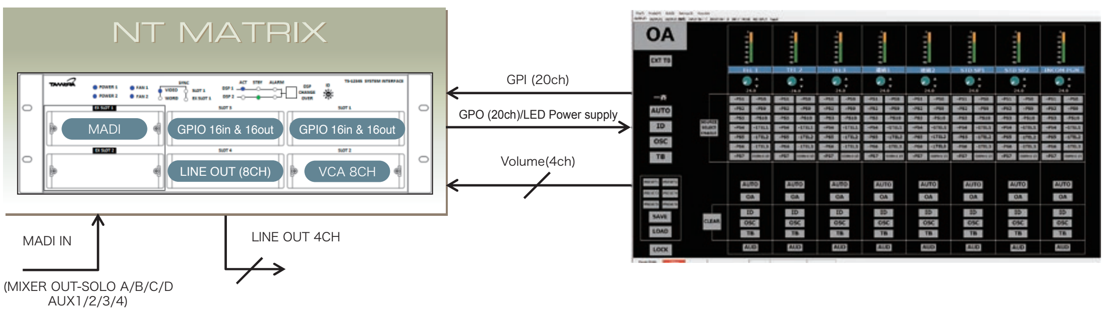
Audio Router (Matrix)



Mix Minus Foldback system



Output Matrix

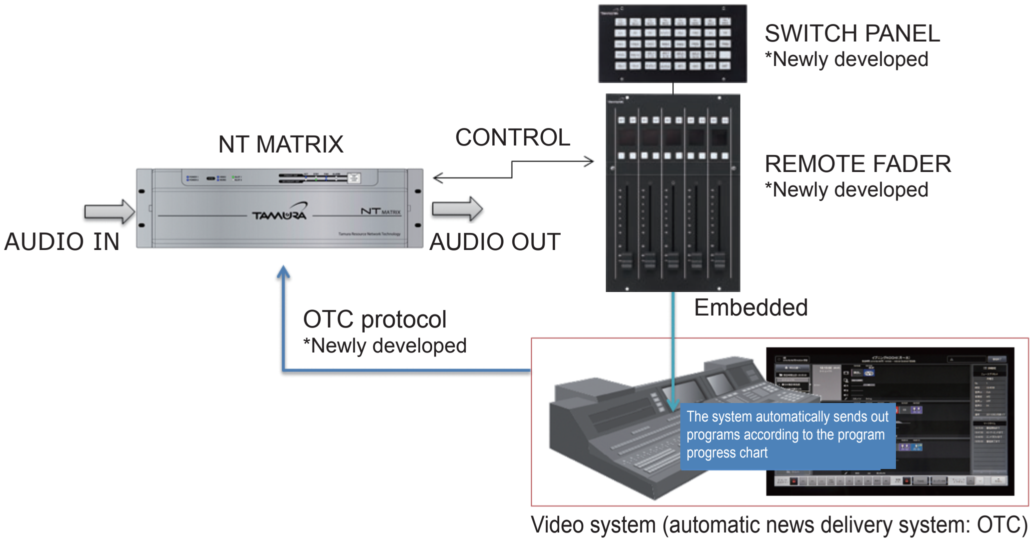


OTC/ONE Touch Control System

Overview

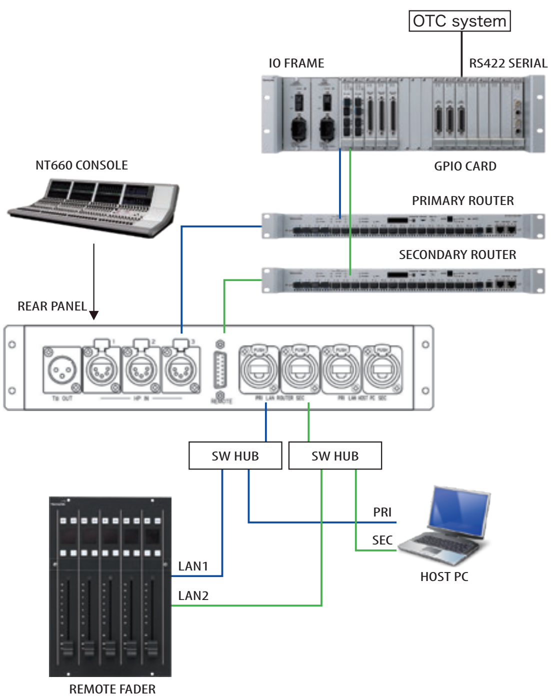
The NT MATRIX OTC system has the function of performing audio processing for an automatic transmission system of news programs. This system receives instructions on the audio part from the OTC system (video equipment) that automatically transmits the program according to the program progress chart (scenario, cue sheet), and performs audio matrix switching and volume control with a fader.

NT MATRIX OTC system



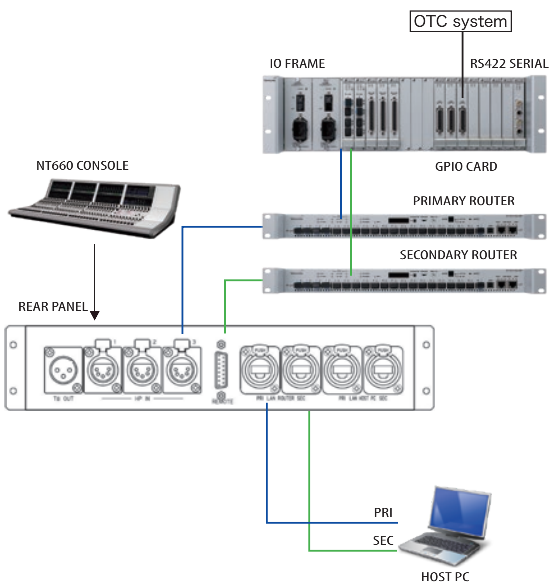
NT660+REMOTE FADER OTC system

Each REMOTE FADER channel is specified as NT660 MTX audio input material by the OTC system. The CONSOLE FADER channel and REMOTE FADER channel, assigned as MTX audio input material, operate being linked together.



NT880/NT660 OTC system

The NT880 Input fader is remotely controlled by the fader level and fade time, specified by serial command control inputted via an OTC terminal.
When the fader is touched during the operation of changing the audio level, the change operation is terminated, keeping the audio level fixed.

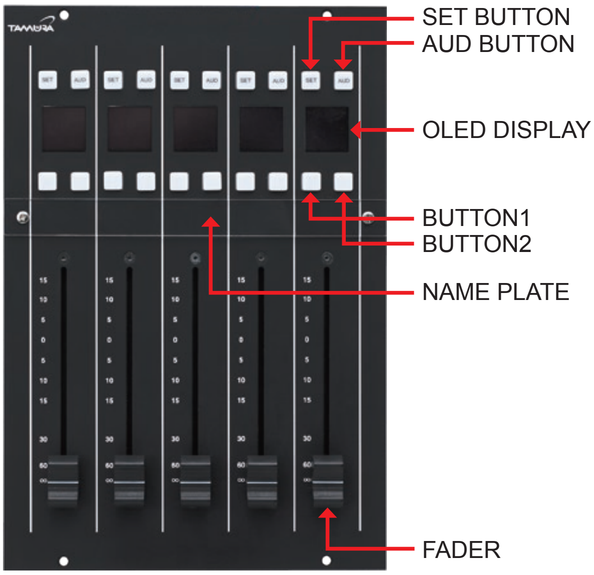


OTC system

Function

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

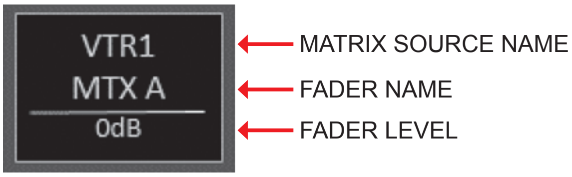
When the SET button is pressed while the TU-6454 SWITCH PANEL button is selected, the source defined for the SWITCH PANEL button is set as the input source for the channel. The set source name is displayed in OLED DISPLAY.

(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]

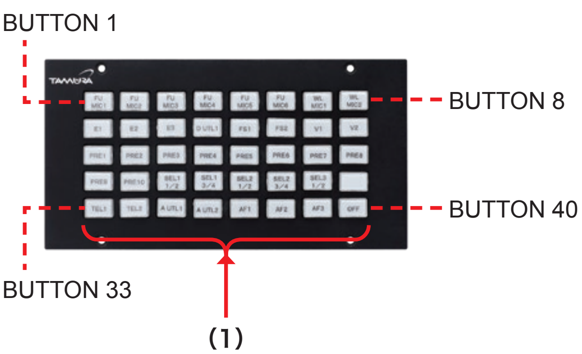
Displays the name of the fader channel.

[FADER LEVEL]

Displays the level value set by the fader.

2 SWITCH PANEL

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

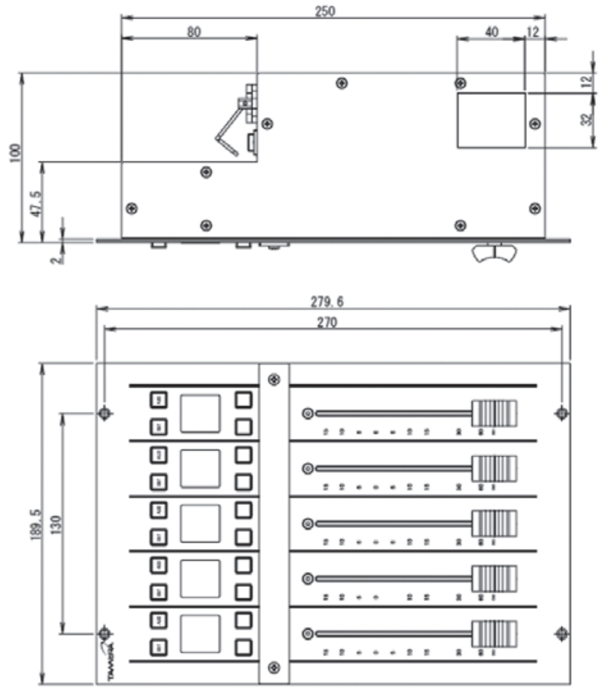


(1) BUTTON 1 to 40

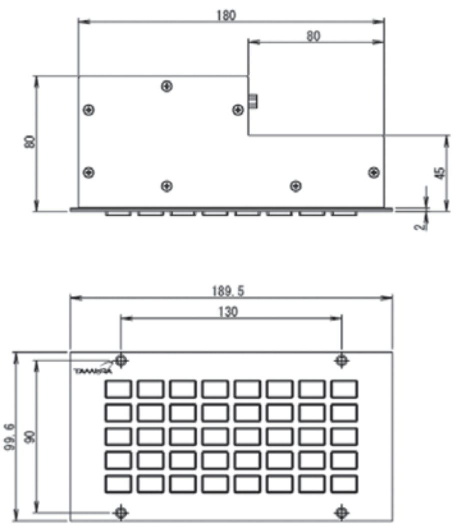
lluminated pushbutton switch. The operation and use of this button are defined in the connection destination device.

Dimensions

■ REMOTE FADER Dimensions



■ SWITCH PANEL Dimensions



DECT Based Wireless Intercom System

DECT Based wireless intercom system is a digital wireless intercom that complies with the new DECT format (ARIB STD-T101).
Audio lines can be organized into 4 groups per system. 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 can be connected so as to enable large-scale system configuration.

Maximum Numbers of personal stations

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

Maximum numbers of personal stations and receivers

Max. **48**
personal stations

+

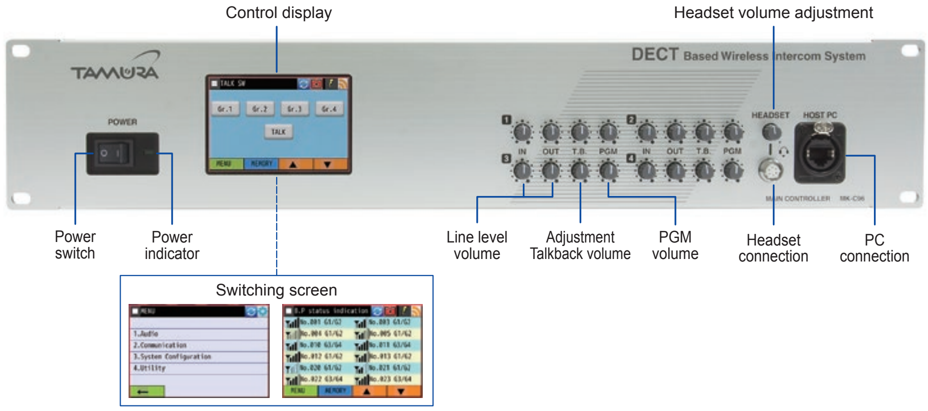
Max. **128**
receivers

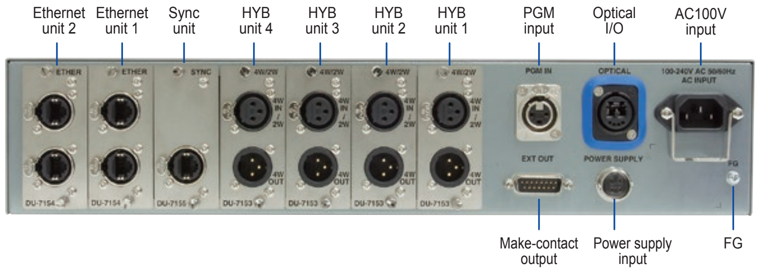
Main System Specifications

Item	Specification
Max. no. of stations connected per system	60 personal stations for calls or 48 personal stations + 128 receivers
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)

Main controller operation•Connection part

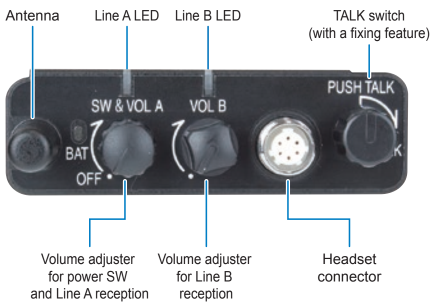
- The main controller can be operated by using the touch panel and adjustable knobs.
- 4W/2W external interface allows for easy integration with wired intercoms.

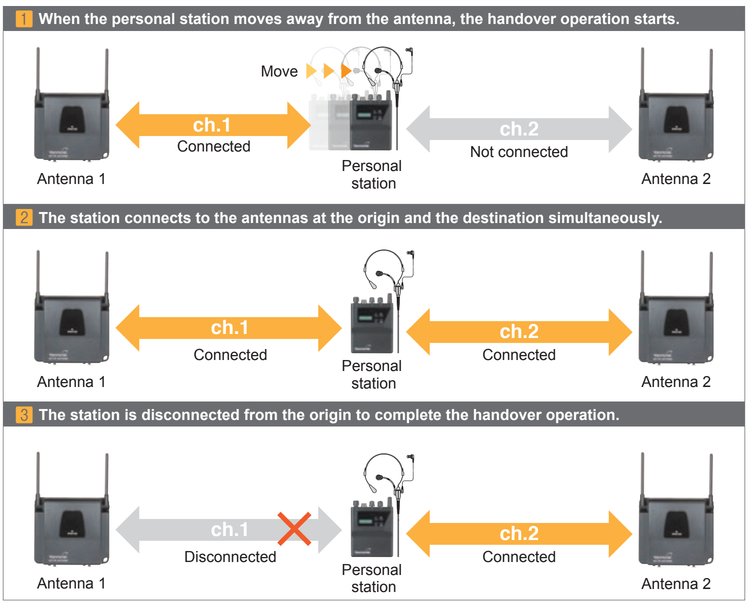




Personal station operation•Active antenna function

- Two audio groups can be assigned to the personal station to allow you to listen to the two groups simultaneously. Each volume adjustment can be intuitively operated using a knob.
- 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.





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	10 personal stations or 8 personal stations and 128 recievers.
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-B96A

- 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

Charger MK-E96

- Charger for battery pack (PBA-4120) AC100 V
- Can be charged while installed on the personal station.

* The product consists of a battery charger alone.

Battery pack

YBA-4120

Build-to-order manufacturing

Build-to-order manufacturing
Batteries used: Two AA alkaline batteries.
* Batteries not included

PBA-4120

Build-to-order manufacturing

Batteries used: Nickel-metal hydride secondary batteries (2.4V)

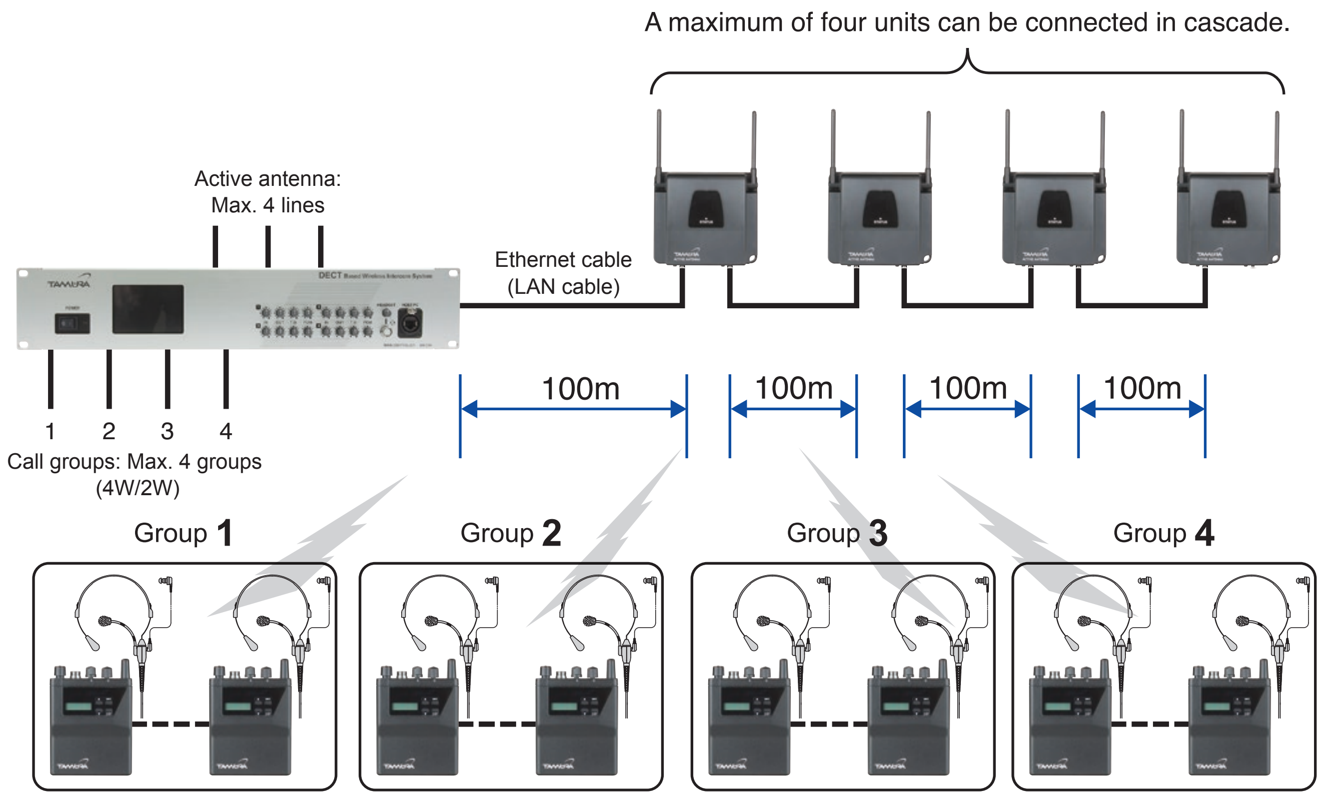
Headset MK-316C/MK-316CTSW / HS-316C / HS-126D

		MK-316C/MK-316CTSW (Condenser type)	HS-316C (Condenser type)	HS-126D (Dynamic type)
Appearance		  Equipped with switch		
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

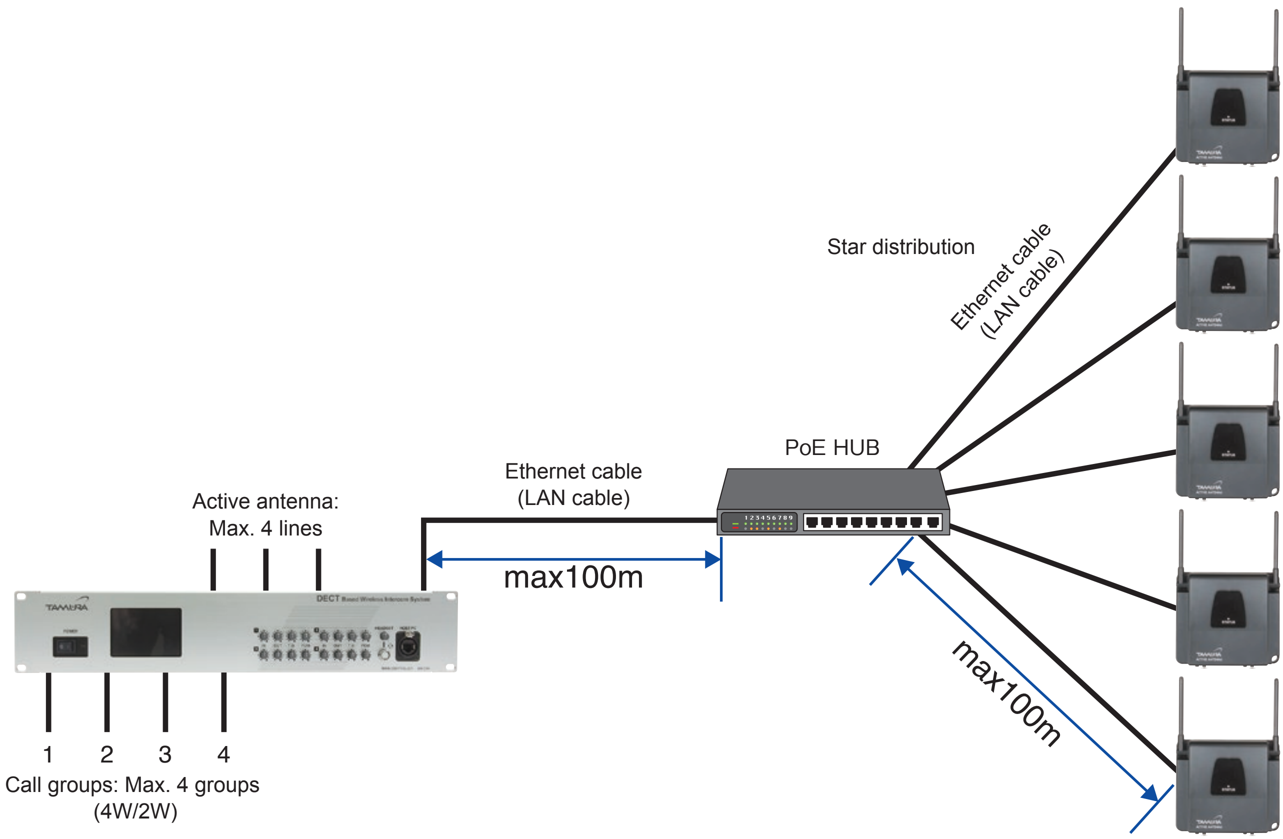
* HS-316C is exclusive for personal station.

System Configuration Examples

Basic configuration

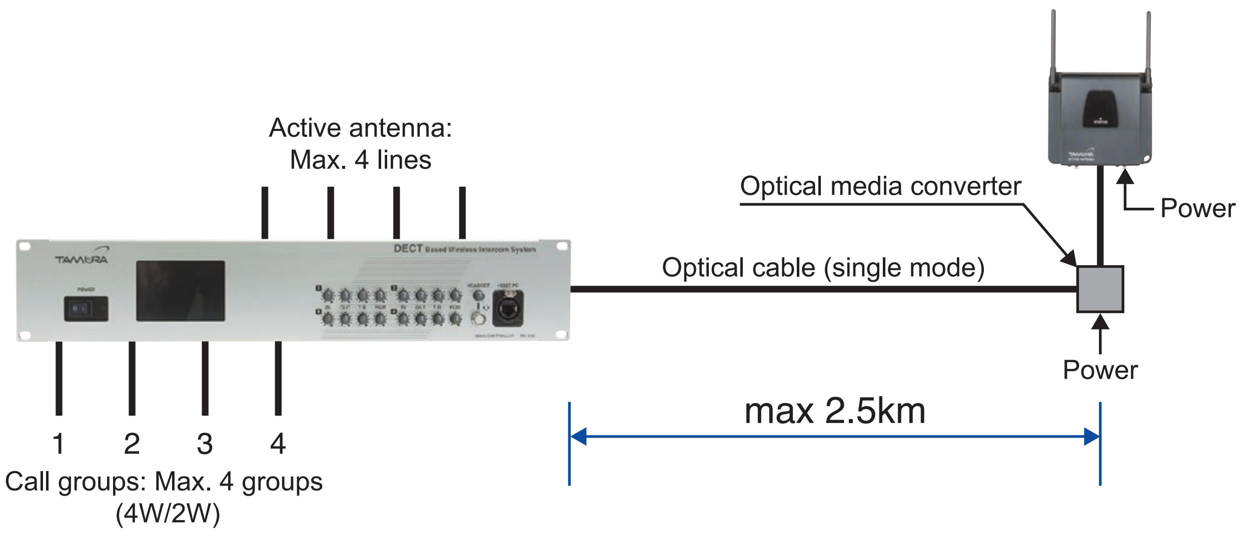


Configuration Example Using HUB

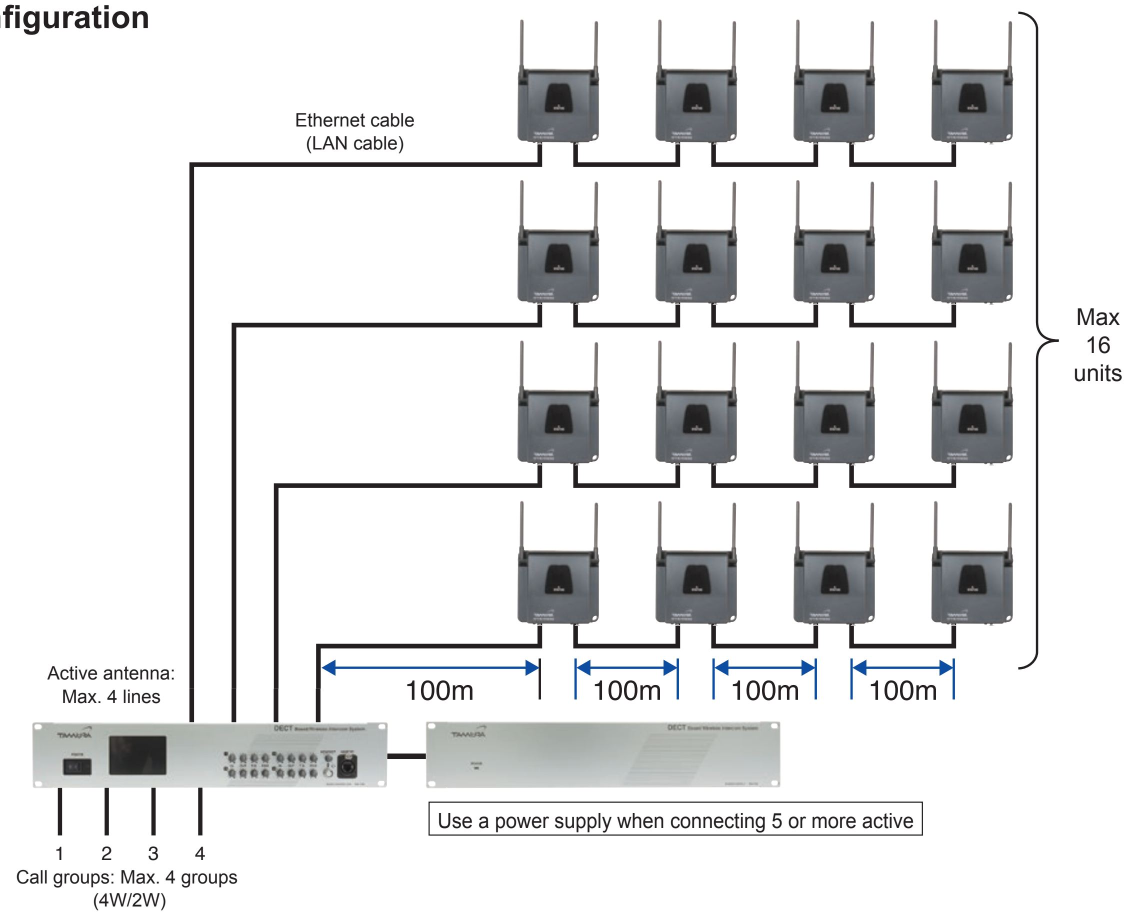


System Configuration Examples

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.

DECT Based Wireless Intercom System

Portable system

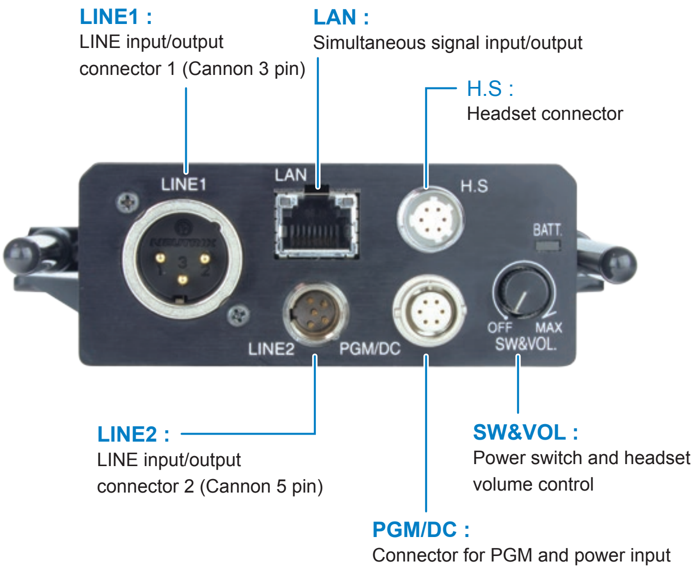
It is a portable system that can fully realize the high functional performance of a DECT type intercom system.
The functions of the main unit and other important units are integrated, and ten handsets can be connected, enabling establishment of two talk groups.
It is battery-powered and can be used outdoors where power supply is not possible.

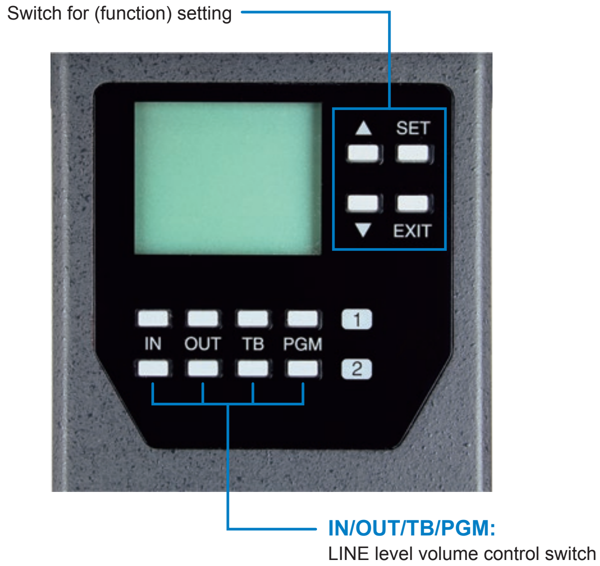
It enables building two talk groups and simultaneous listening.

Ten handsets can be connected.

Main specifications of the system

Item	Specifications
Maximum number of connected handsets per a portable controller	10 units
Number of talk groups per a portable controller	2
Number of simultaneous listening groups for a handset	2 (independent volume adjustment is possible)
Audio frequency characteristics	100Hz ~ 7kHz
Radio system / frequency used	ARIB STD-T101 / 1.9 GHz band
Multipath compatible handsets	Polarized wave diversity
Communication distance (line of sight)	about 300 m
Continuous operating time	about 8 hours (two AA alkaline batteries) about 10 hours (two AA Ni-MH rechargeable batteries)

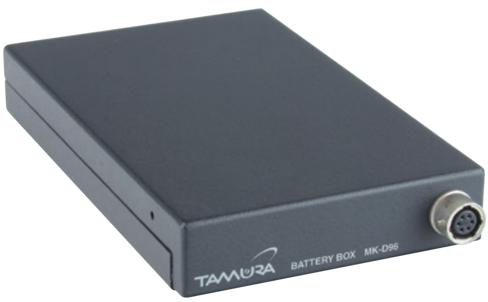




Portable MK-H96

Item	Specifications
Number of handsets connected per a portable controller	10 units
External interface	2 systems (LINE input)
PGM input	1 system
Power supply	DC 8.0 V to 16.0 V
Environment	-10℃ to 50℃
Dimensions	H129×W89×D36(mm) (excluding dimensions of protrusions)
Weight	about 455 g

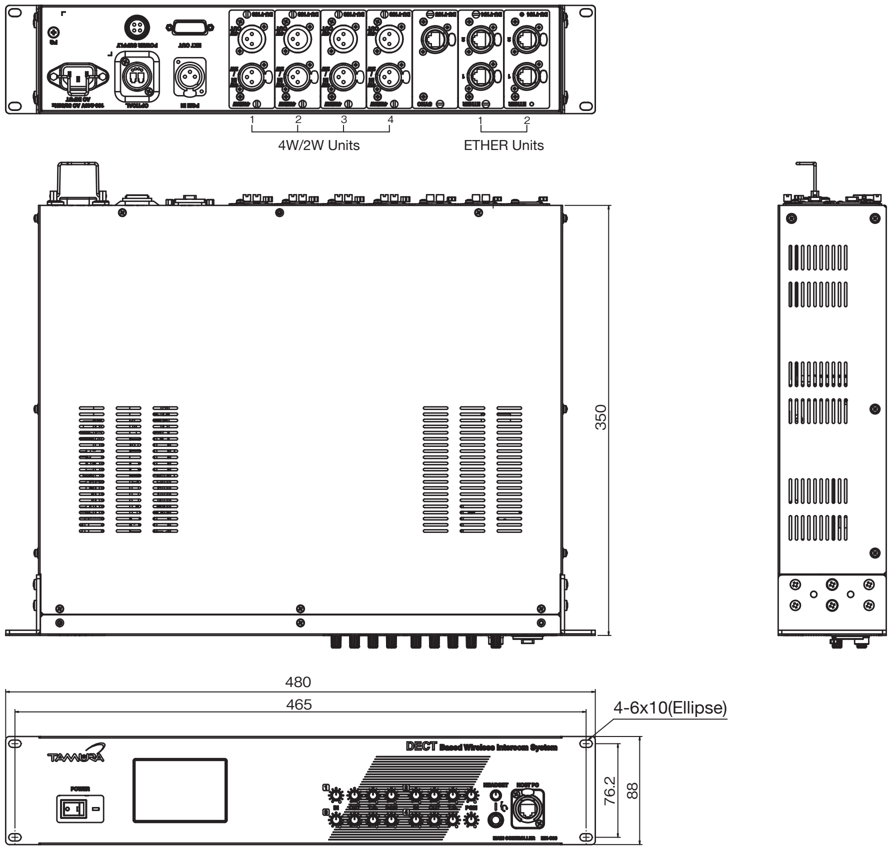
Battery box MK-D96



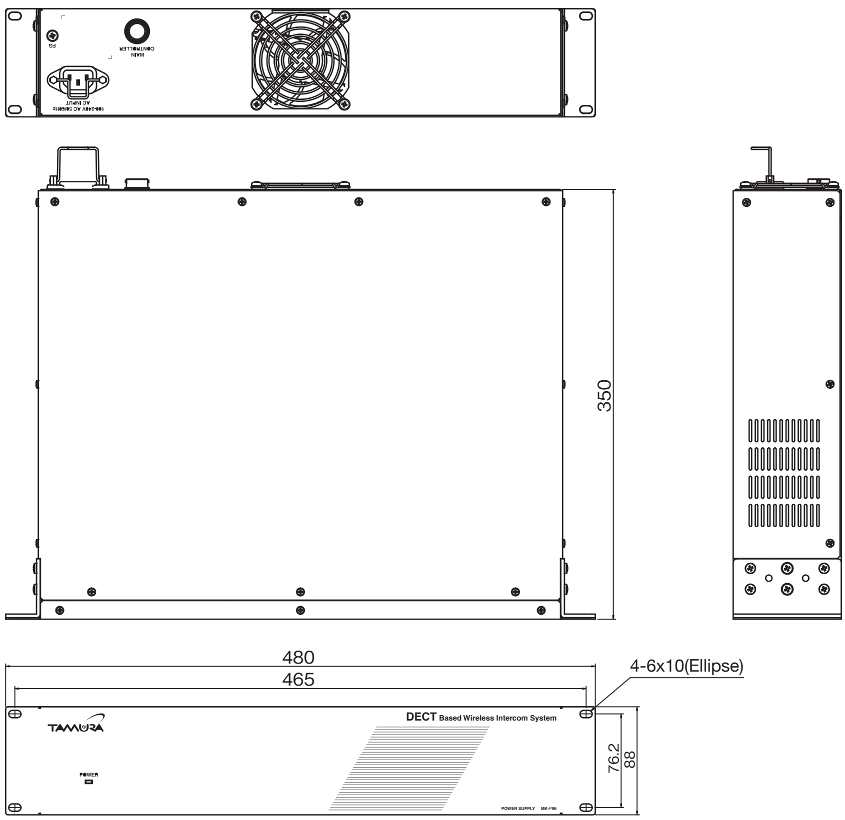
Battery box for eight AA alkaline batteries or Ni-MH secondary batteries

Item	Specifications
Dimensions	H142×W89×D22(mm) (excluding dimensions of protrusions)
Weight	about 160 g (excluding cables and batteries)

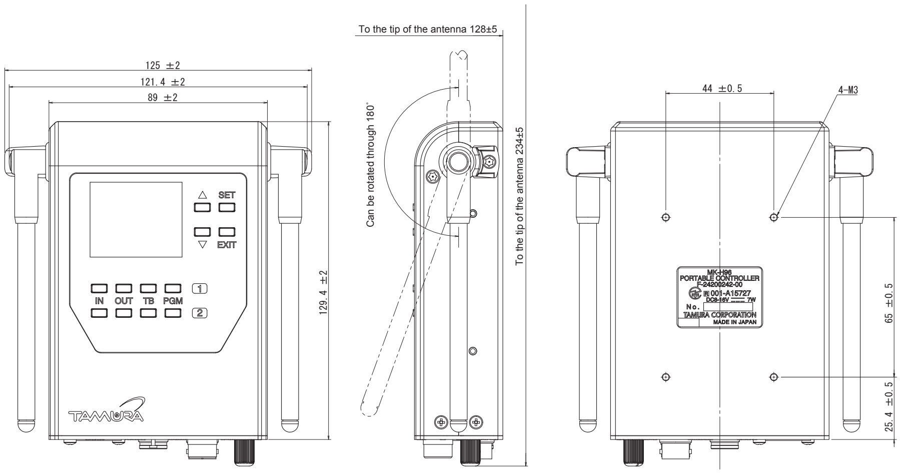
MK-C96
Maincontroller

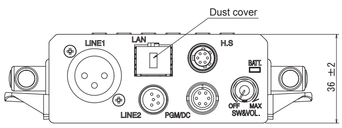


MK-P96
Subcontroller



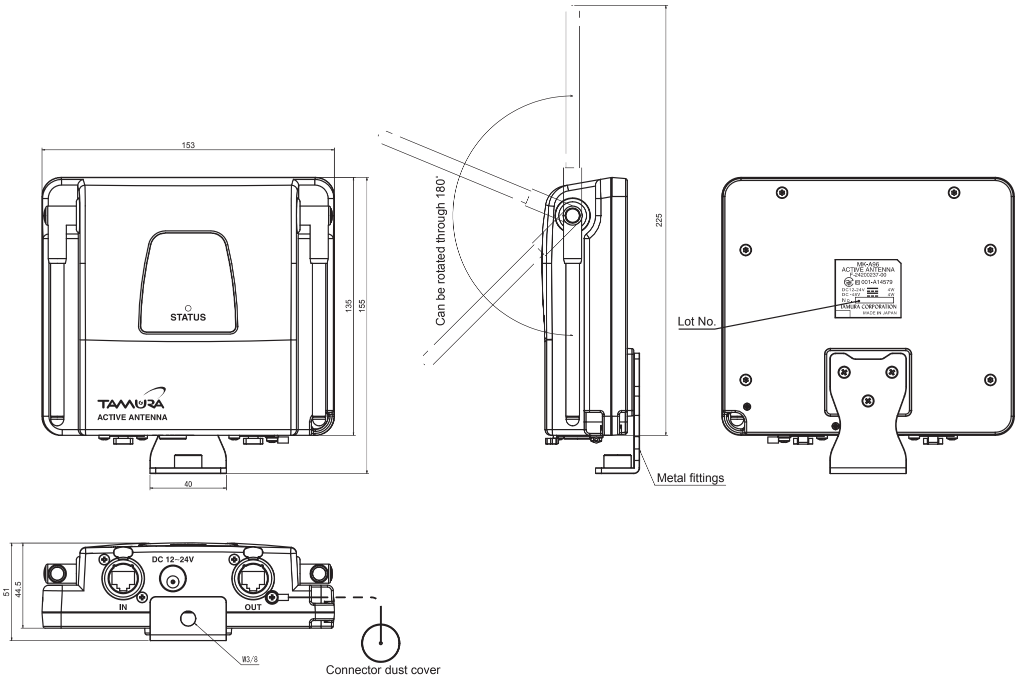
MK-H96
Portable controller



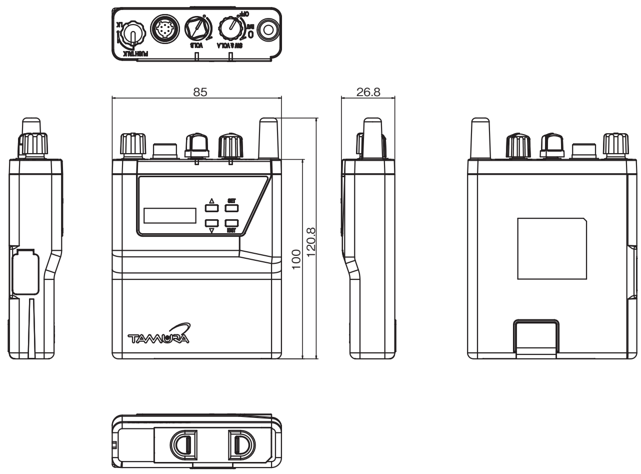


Note: Appearance color: Black leather-tone finish

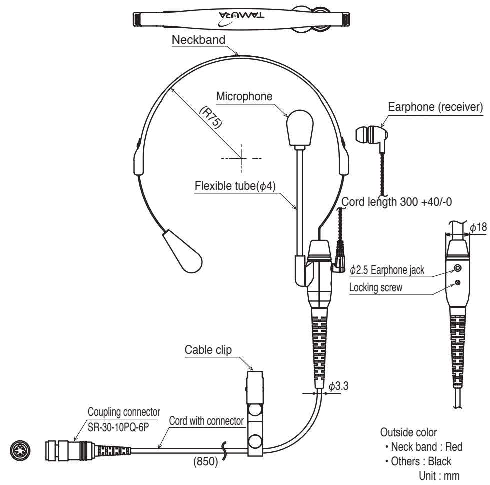
MK-A96
Activeantenna



MK-B96
Personal Station



MK-316C
HEADSET



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