English

TAMURA

Professional Studio Audio

Audio Equipment
&
Communication Systems

Audio Equipment
Communication Systems

DIGITAL AUDIO MIXING CONSOLE

NT Series

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NT880

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NT660

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NT110

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NT

Tamura Resource Network Technology

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

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

High-speed data transmission protocol TR-LINK

Simplified connections between units

A single-mode optical fiber cable is used for TR-LINK.
In addition to 512-channel audio signals, synchronous and control signals are transmitted via a single optical fiber cable.
Therefore, the synchronous signal cable and control signal cable, which were conventionally required for each unit along with the audio cable, are no longer required, and the units are connected to each other using a pair of optical fiber cables only.

Easy maintenance

Maintenance of the router unit and DSP core, which form the heart of the system, is performed by unit-wise replacement instead of the more troublesome replacement of a circuit board.
Because connections between the units are made using optical fiber cables only, a faulty unit can be replaced even when the system is operating. Connecting and disconnecting an optical fiber cable while the system is operating does not affect the system operation.

32-bit floating point data transmission

MADI was previously used for connections between the I/O unit and the audio processing unit.
MADI performs 24-bit fixed point data transmissions, however, and even if the DSP core performs high-precision arithmetic operation, some data loss is unavoidable owing to data transmission using MADI.
When using TR-LINK, on the other hand, all audio data are transmitted in a 32-bit floating point data state.
Thus, as long as the I/O unit is connected to DSP via TR-LINK, regardless of the distance between them, there will be no data loss at all, as if these units are connected in a single cabinet.
Analog audio signals to be input to the I/O unit are converted into 32-bit signals in the I/O unit, whereas analog audio signals to be output from the Line-out card are directly converted into analog audio signals from 32-bit signals in the I/O unit.

Separation of units

A massive volume of data is sent and received between the DSP module and the routing module, and a mutual connection via the backplane inside the same cabinet was the only method used in the past.
When the above method is used, all modules are put in an electrically connected state. Therefore, it was not possible to completely eliminate the risk of a trouble occurring in a single module affecting the other modules.
On the other hand, using TR-LINK, which can transmit 512-channel audio data in a 32-bit floating point data state, the data can now be transmitted between modules using an optical fiber cable. As a result, the DSP module and routing module can be installed as completely separate units.
The units are completely electrically separated from each other and thus it is possible to minimize the risk of a trouble occurring in a single unit affecting the entire system.

Hybrid Audio Processing

Higher integrated processor

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

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

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

Availability and fault tolerance

Hot standby system

The router unit, which is a core component of the system, has a backup system that is always on hot standby. That is, exactly the same unit is in a standby state with the same operation status as that of the active unit.
The standby system always stores a mirror copy of the active system's operation status. Therefore, switching to the standby system can be performed immediately.
This feature minimizes the system downtime.

High-speed startup

The startup time of the entire console system from its power-off state is approximately 30 seconds.
Even when a critical system error occurs and the entire system must be restarted, this feature can minimize the system downtime.

Firmware-based system

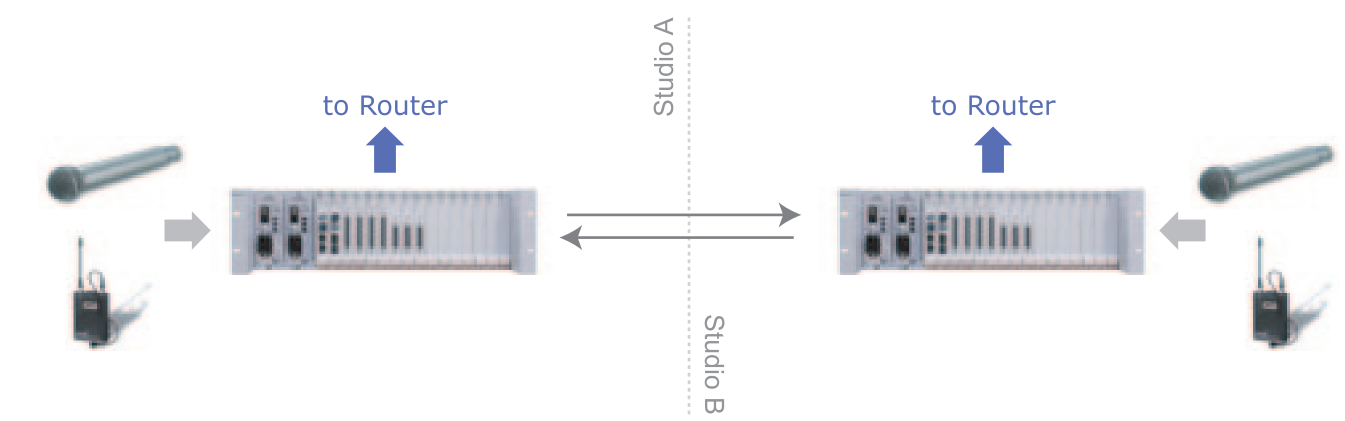
The NT series has been built as a firmware-based system without using general-purpose operating systems such as Windows and Linux.
Because this system does not require a shutdown operation, the system can be restarted promptly at any time.
Furthermore, all operations are always stored in the backup memory; therefore, the status immediately prior to shutdown is restored when the system is restarted.
Even when the system is involuntarily restarted after a power trouble or other unexpected accidents, the operation status will be securely maintained.

IO Sharing

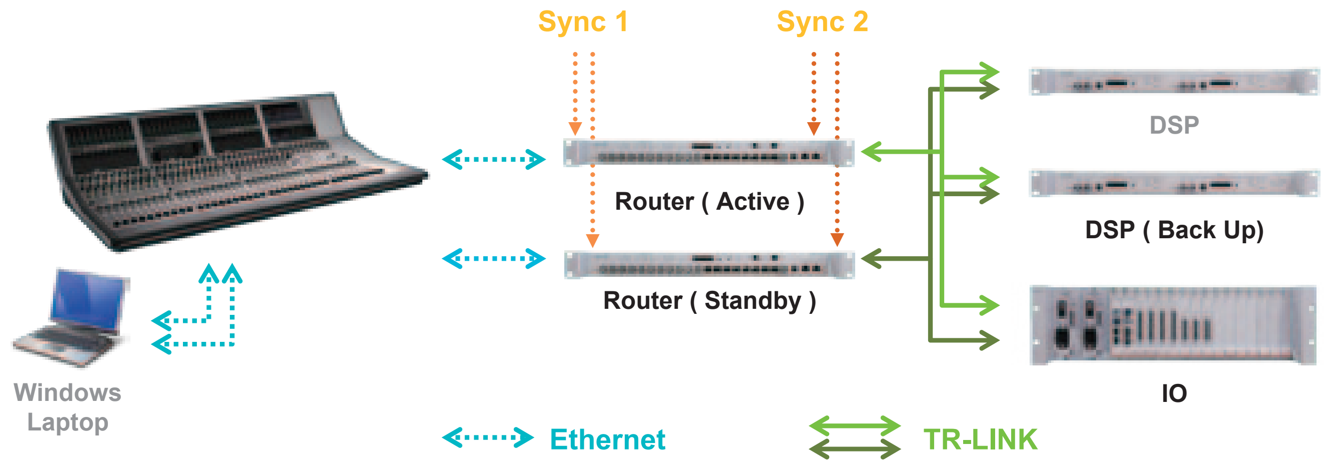
Sharing of input audio

The audio input to a single I/O Frame can be shared between multiple systems.
For example, you can construct a system that allows a microphone to be used in each studio from either of the two studios.
This feature makes it possible to mutually use the systems at two studios for emergency backup or use one of the systems as the Premix Mixer.

Controls, such as the gain control of a microphone input shared by multiple systems, are enabled from any system.
Furthermore, the control protect setting is made at any system, which enables the gain control to be performed from a specific system only.
The input audio can be shared among a maximum of eight systems.



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), Word, AES3 / AES3id
- DSP CORE — Maximum 5 DSP core units (including 1 backup unit)
- Number of TR-Link audio channels — 512ch

ROUTER

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

DSP CORE

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

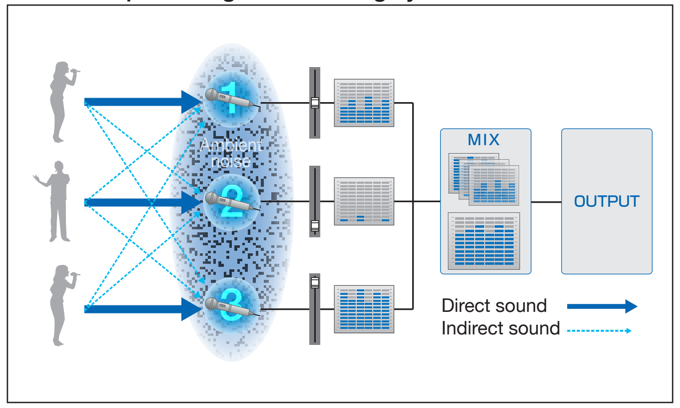
IO FRAME

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

AUTOMIX function

The AUTOMIX function of NT series models automates some of the mixing operations.
In broadcast and content production that use several microphones, an audio mixing engineer must accurately and immediately control the fader for multiple microphone channels depending on the situation.
The AUTOMIX function uses network technology to perform an automatic microphone channel fader operation in an effort to lighten the load of the mixing engineer and provide environment in which the engineer can concentrate on sound quality adjustment and other tasks.

Conceptual diagram of mixing system environment



Main Specifications of AUTOMIX

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

The AUTOMIX function of NT series models uses gainsharing technology to provide the following features.

(1) Makes it possible to gain a natural auditory sensation
- Produces a sound without the obvious noise gate effect.
- Produces sound right from the start of a speech.
- No need for mixing engineer to bother about level fluctuations.
- Causes no ambience imbalances.

(2) No need to set a threshold level
- Ambient noise during a low threshold level does not cause the gate function to activate.
- High threshold levels do not cause the gate to be closed.
- Even if the threshold level is set in a quiet room, it will operate properly when there is audience clapping or a musical performance.

(3) No need to set attack time and hold time

(4) No unnatural muting (no ambient) occurs even immediately after a speech has finished and the subsequent feeling of reverberation is maintained.

(5) The endings of words in a speech are captured properly.

(6) The quality of ambient noise does not change when a new speaker starts talking.

(7) No popping noise occurs in the lower frequency range (caused by gate operation).

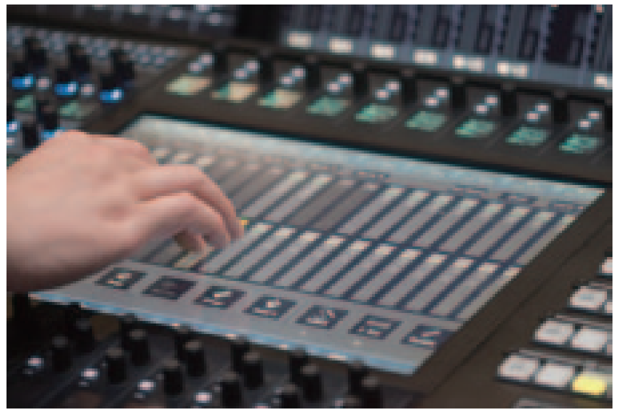
NT880

with Tamura Resource Network Technology

Excellent operability

Two parameter operation methods

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



Channel layout editing functions

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

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

Expandable to large-scale systems

Number of physical faders

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

Operator-specific section concept

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

High-performance processor

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

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

Specifications

Console

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

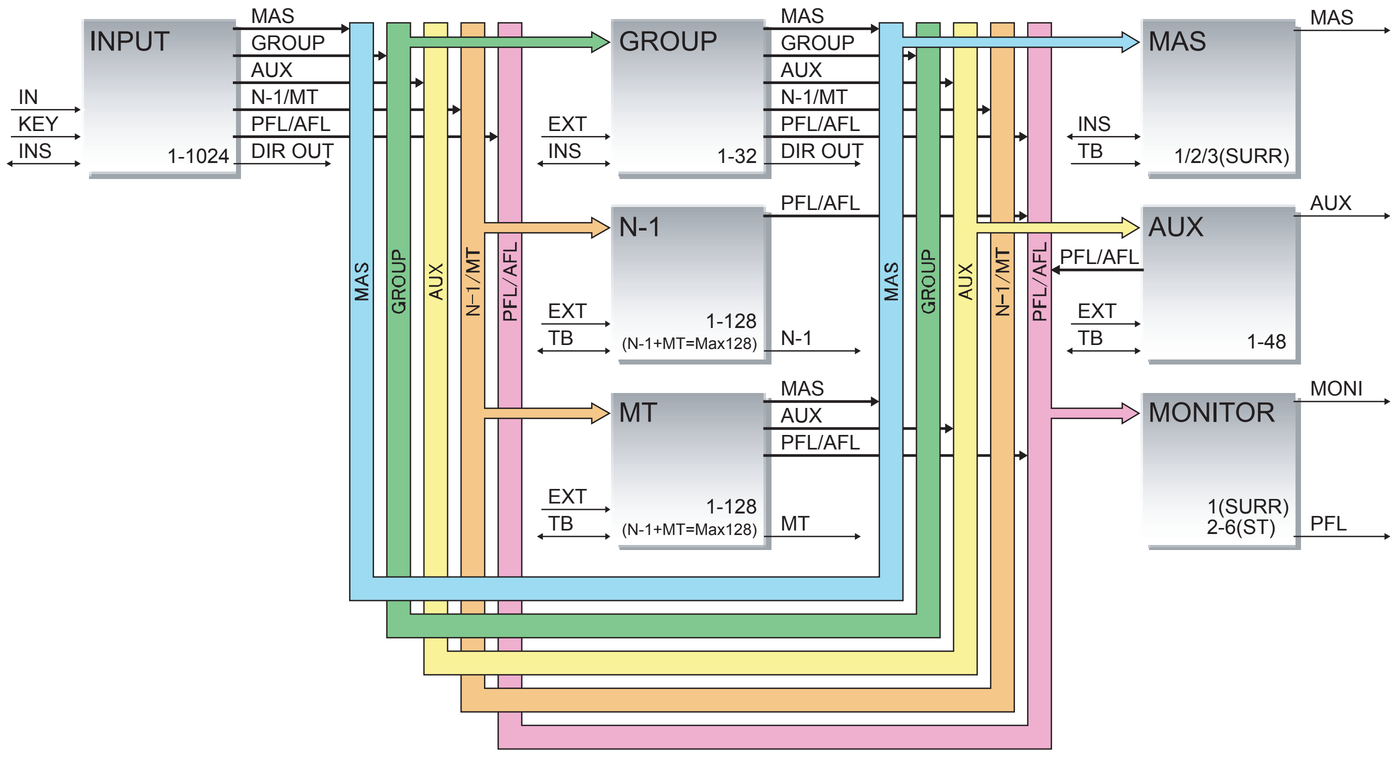
Audio channel (Fs=48kHz)

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

Audio control parameters

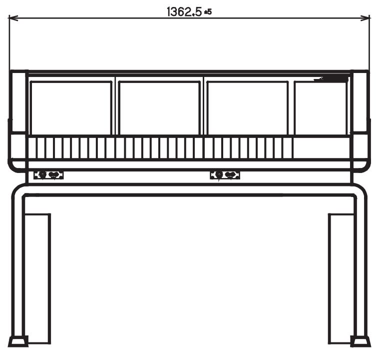
HA Gain	+10dBu~-64dBu
Trim	+24dB~-24dB
Delay	5000ms or more
Filter	Filter1 (HPF/Notch) Filter2 (LPF/Notch)
Equalizer	4Band (Support for all frequency bands)
Dynamics	Compressor 2 channels Gate/Expander 1channel

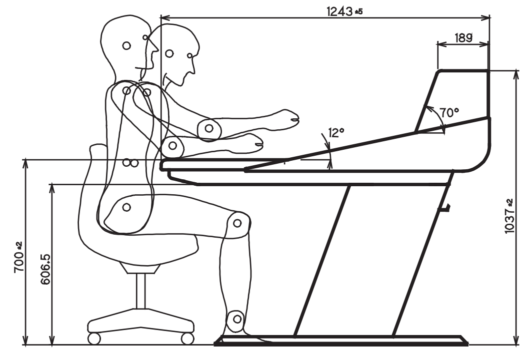
Audio block diagram



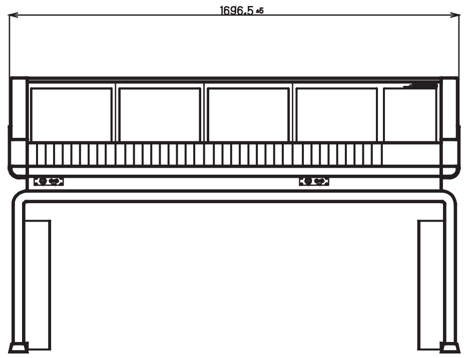
Dimension

30 Fader

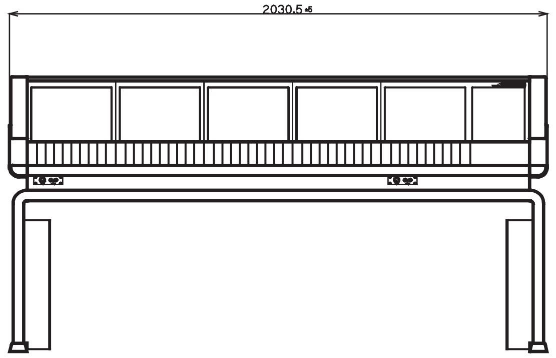




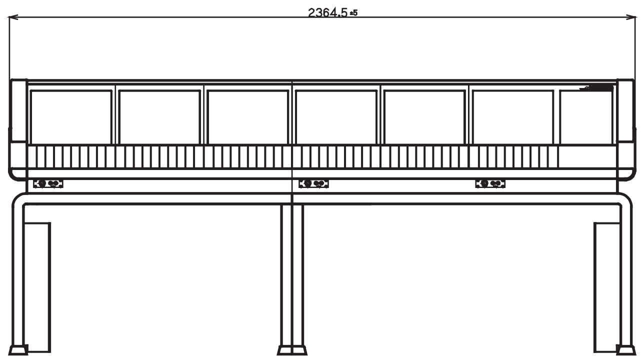
40 Fader



50 Fader



60 Fader



NT660

with Tamura Resource Network Technology

Flexible Operation

New parameter operation method

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

Touch Panel Surround Panner

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

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

Greatly Enhanced Functions

Inheriting Enhanced Functions

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

User Level Setting

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

Consolidated Control of Bus Outputs

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

DAW Control Functions

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

Specifications

Console

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

Audio channels (Fs=48kHz)

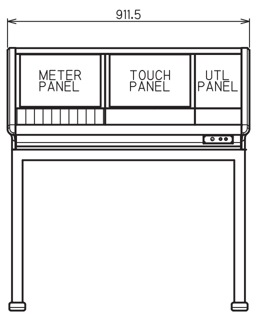
Master Bus	Maximum 24 buses (3 surround)
Group Bus	Maximum 32 buses
Aux Bus	Maximum 48 buses
N-1 / MT Bus	Maximum 128 buses
AFL	1 surround
AFL / PFL	3 stereo
PFL	1 stereo
Main Monitor	1 surround+stereo
Sub Monitor	3 channels (Stereo)

Audio control parameters

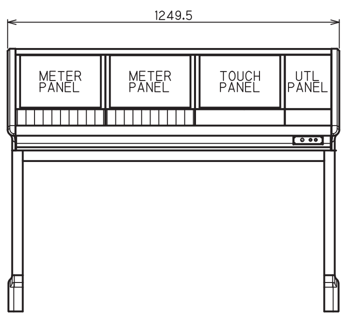
HA Gain	+10dBu~-64dBu
Trim	+24dB~-24dB
Delay	5000ms or more
Filter	Filter1 (HPF/Notch) Filter2 (LPF/Notch)
Equalizer	4Band (Support for all frequency bands)
Dynamics	Compressor 2 channels Gate/Expander 1channel

Dimension

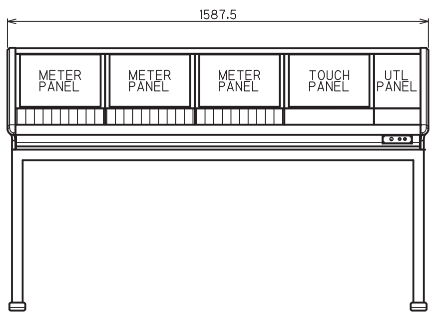
20 Fader



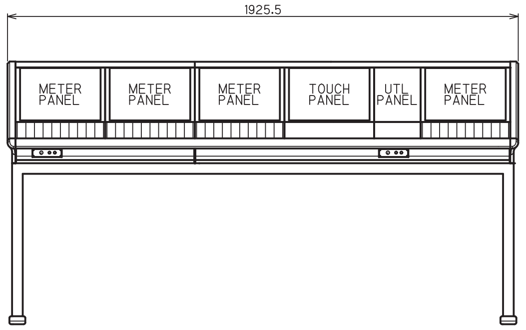
30 Fader

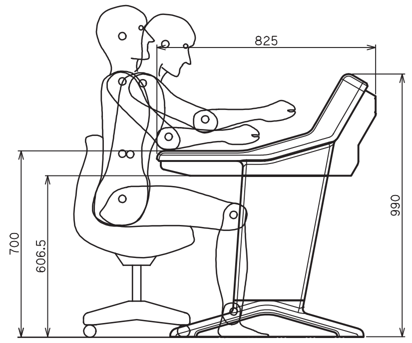


40 Fader



50 Fader





NT110

DIGITAL AUDIO MIXER

Operability of trust

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

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

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

Portable Model with Inherited Functions RE-Liability of NT Series

Specifications

Overall Rating

Dimensions (without Side panel)	490(W)×221.5(H)×610(D)mm (Protruding parts not included) 430(W)×220.5(H)×550(D)mm (FRONT/SIDE PANEL not included)
Weight	19 kg
AC	100 - 240V, 50/60Hz
DC	12V/14.8V
Power Consumption	150W
Operating free-air temperature range	-10~ 40˚C
Number of faders	16 Fader
Bank/Layer	3Bank/2Layer

Audio Channels (Fs=48kHz)

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

Audio control parameters

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

Option (Will be released)

- DSP CARD (BACK UP REDUNDANT DSP)
- OPTION CARD
  AES3id IO CARD(4ch IN + 4ch OUT BNC)
  GPIO CARD
  MADI CARD(1Coax/1Opt)
  DANTE CARD
- Storage case

Audio block diagram