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YAMAHA

PM1D VERSION 2

DIGITAL AUDIO MIXING SYSTEM

PM1D, Reloaded

The PM1D advances with the long-awaited Version 2. A new flagship model is born.

With its launch in 2000, the Yamaha Digital Audio Mixing System PM1D completely and comprehensively changed the world's idea of what an "ideal" large-size console should be. In addition to its groundbreaking inclusion of all-digital-processing, intuitive operation, and total recall functions, it also offered overwhelming programmability, unprecedented sound quality, excellent portability and installation, redundancy for reliability, and superior cost-effectiveness. That's why the PM1D has become an indispensable tool for professional engineers and has made its way into the sound reinforcement scene, world-famous concert halls, theaters and broadcasters.

Now in 2005, we present Version 2 of the PM1D.
We collected feedback from a wide spectrum of fields.
After thorough analysis, we defined the needs of each category, and installed all-new features to give supreme usability for live performances, sound reinforcement, halls, and broadcasting needs.

With the PM1D Version 2, Yamaha presents the best and newest answer to the question: "what an 'ideal' large-size console should be."

Built-in standard Add-On Effects (Rev-X and VCM)

Remote controllability of the PM1D V2 from units such as the DM1000

Additional new features for better usability in sound reinforcement, theaters, facilities and broadcasters

Equipped with a Wide Array of Advanced Functions, the PM1D Version 2 Expands the Horizons of Digital Mixers.

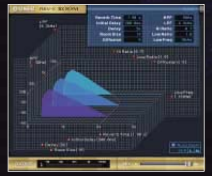
ADD-ON EFFECTS Inside

Many of the latest and highest-grade effect programs have been added to the PM1D Version 2. Choose from three types of effects groups known as Add-On Effects, which are already favorites with professionals thanks to their installation in the PM5D, the PM1D's sister model, and the DM2000 Version 2, a much-praised production console. Add-On Effects include a recording-studio-grade compressor, a vintage EQ, and an effector that offers tape compression derived from open deck recording, as well as the Reverberator REV-X, which uses Yamaha's exclusive new algorithms. With the PM1D Version 2, you'll master the use of these all-new effects in no time.

[AE011 Channel Strip]
Five models that employ VCM (Virtual Circuitry Modeling) technology to recreate the sound and characteristics of several classic compression and EQ units from the '70s.

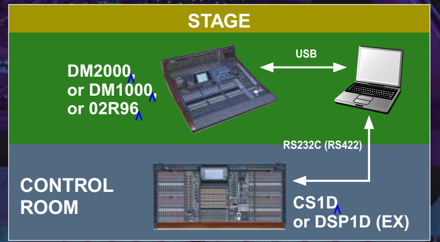
[AE021 Master Strip]
Open Deck VCM technology recreates both the analog circuitry and tape characteristics that shaped the sound of open-reel tape recorders. The choice of "old" and "new" tape types, tape speed, bias, and EQ settings that can vary the "focus" of the sound, distortion and saturation characteristics are also available.

[AE031 Reverb]
"REV-X" algorithms, first introduced in the SPX2000 Professional Effect Processor, deliver the richest reverberation and smoothest decay available. REV-X Hall, REV-X Room, and REV-X Plate programs are provided, with parameters such as room size and decay envelopes offering unprecedented definition and nuance control.



Remote Control Support

This function, demanded by many venues, allows the PM1D to be remotely controlled from near the stage, even when it is placed in relatively distant locations such as a control booth. A computer running the PM1D Manager software, in conjunction with any of the DM2000 V2, 02R96 V2, or DM1000 V2 compact digital mixers, can be used as a remote control device. Using a digital mixer that combines excellent portability with small size offers a new solution that can be flexibly applied to a wide variety of situations. PM1D parameters capable of remote control include all functions of the remote controller-equipped channels as well as MIX, STEREO, MATRIX, 100 scenes, and more.



- The controllable channel functions may be partly traded off.
- Both the CS1D V2 and DSP1D (EX) support connection to a laptop computer.
- Use of the RS422 is recommended for connection of a PC to the CS1D V2 or the DSP1D (EX). Because the PM1D is equipped with a RS232C terminal, a conversion device is required when using RS422.

New Features Offer Better Usability in Sound Reinforcement, Theaters, Facilities, and Broadcasters

Automatic Gain Adjustment
Load Filter / Save Filter
Event List
Fader View
Auto Store Overwrite
Insert/Delete Channel
Direct Out from Pre High Pass Filter
Assignable Functions Added to User Define
GR Meter Display
Enhanced Factory Preset Memory & Additional Libraries
Expanded Recall Safe Functions
*Supports MY8-DA96 Card, MY8-ADDA96 Card**

* PM1D does not operate in 96kHz.

PM1D VERSION 2

DIGITAL AUDIO MIXING SYSTEM

PM1D Version 2 is a digital audio mixing system consisting of a control surface CS1D, power supply PW1D, DSP unit DSP1D (DSP1D-EX), analog input box AI8, analog output box AO8, digital I/O box DIO8, I/O cards, etc.

SOFTWARE UPGRADE KIT **PM1D V2K**
Software kit for smoother upgrade of PM1D Version 1.x to PM1D Version 2.

Control section

CONTROL SURFACE CS1D

- Control section of PM1D. All mixing controls of audio are performed here.
- 48 + 4ST, total of 56ch X 2 layers, up to a maximum of 112ch of input.

D-sub DIGITAL AUDIO CABLE
(FOR MONITOR, 2TR IN TALK BACK, CUE)

BNC CABLE
(FOR CONTROL)

USB, SERIAL

SETTING DATA

DC POWER

PC Card
(Type II)

Personal Computer
(Windows®)

POWER SUPPLY PW1D

- Exclusive power supply unit for the control surface CS1D.
- Compact design of 2U size for optimum space utility.
- 2 PW1D units can be connected to a single CS1D unit to serve as backup.

PM1D Version 2 SYSTEM STRUCTURE

Sound systems used in tours/concerts, halls, theaters and broadcasting are rapidly growing in size, diversification and sophistication.

In order to design a mixer that can respond to the various needs of these scenes, the development of PM1D started by taking a step back and treating the consoles as a "single distributed processing system consisting of multiple devices".

By adopting console functions as components and storing functions in their ideal locations, an extremely efficient and highly flexible system structuring became possible. At the same time, complete fail-safe operation was realized by utilizing redundancy in the power supply section, DSP section, control section, etc.

Furthermore, advanced operational environment including features such as enormous number of allowed I/O, unbeatable space efficiency, incorporation of effects into the main unit, storing/recalling of mixing data and offline editing using a PC was achieved.

Processing section

DSP UNIT DSP1D
(DSP1D-EX)

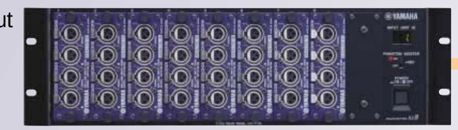
(Maximum digital patch input of 320ch)
(Digital patch output of 192ch)

- Processing engine that processes all digital audio of the PM1D system.
- Digital patch input: compatible up to 320ch, digital patch output: compatible with 192ch.
- Processing input: 48 channels + 4 stereo inputs (96 mono + 8 stereo input for DSP1D-EX). Output: 24 channels matrix in addition to 48 mix + 2 stereo buses.
- 2 units of DSP1D (-EX) can be connected to a single CD1D unit for fail-safe use by mirroring.
- Built-in serial and USB ports for connection with PC. Even if CS1D cannot be used, mixing can be performed by directly connecting the DSP1D and PC.

- 9U size rack mount type for easy installation/transportation. 6 independent internal boards (7 for DSP1D-EX) by function allow easy board replacement for superior maintainability.

I / O section

Analog input



analog input card × 8

LMY2-MLAB
(with head amplifier)

LMY4-MLF
(with head amplifier)

MIC/LINE INPUT CARD LMY2-MLAB (with head amplifier)

- Card equipped with a head amplifier for analog input of mic/line level. 2 channels (A/B input switching) available per card.
- Built-in 28-bit equivalent, high precision AD converter using AD floating control.
- Remote control of A/B input switching, phantom power supply on/off, head amp gain control by 1dB step from CS1D.

MIC/LINE INPUT CARD LMY4-MLF (with head amplifier)

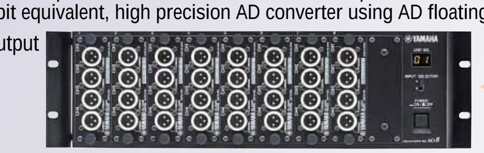
- Card equipped with a head amplifier for analog input of mic/line level.4 channels available per card.
- Remote control of ~~phantom power supply~~ on/off, head amp gain control by 1dB step from CS1D.

AD CARD LMY4-AD

- Compatible with input of line level. 4 channels available per card.
- Built-in 28-bit equivalent, high precision AD converter using AD floating control.

LMY4-AD

Analog output



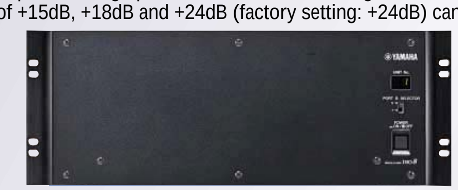
analog output card × 8

LMY4-DA

DA CARD LMY4-DA

- Card that converts digital signals from DSP1D to analog signal output. 4 channels available per card.
- Built-in 27-bit equivalent, high precision DA converter using DA floating control.
- Output levels of +15dB, +18dB and +24dB (factory setting: +24dB) can be selected for each channel via switches.

Digital I/O



digital I/O card × 8

MY8-AE **MY8-AT** **MY8-TD** **MY4-AD** **MY8-AD24** **MY4-DA** **MY8-ADDA96**

DIGITAL I/O CARD
MY8-AE(AES/EBU) MY8-AT(ADAT) MY8-TD(TASCAM)

- Digital I/O card compatible with digital formats of AES/EBU, ADAT and TASCAM.
- 8 IN/8 OUT available per card.

AD CARD MY4-AD

- AD card compatible with 24-bit/4-channel analog input.

AD CARD MY8-AD24

- AD card compatible with 24-bit/8-channel analog input.

DA CARD MY4-DA

- DA card capable of 20-bit/4-channel analog output.

ANALOG INPUT BOX AI8

- Analog input box capable of mounting 8 input cards per unit.
- Lineup of mic/line-input compatible, head amplifier-equipped 2-channel LMY2-MLAB (with A/B input select switch), 4-channel LMY4-MLF and line-input compatible 4-channel LMY4-AD cards available.
- Up to 10 units of AI8 can be connected to a single DSP1D (DSP1D-EX).
- 3U-size rack mount type.
- 3 digital output jacks to DSP engine with parallel output capability.

ANALOG OUTPUT BOX AO8

- Analog output box capable of mounting 8 output cards per unit.
- 4-channel output LMY4-DA card available.
- Up to 6 units of AO8 can be connected to a single DSP1D (-EX).
- 3U-size rack mount type.
- 2 digital input jacks from DSP engine with switching capability.

DIGITAL IO BOX DIO8

- Digital IO box capable of mounting 8 optional I/O cards per unit.
- Compatible with MY8-AE (for AES/EBU), MY8-AT (for ADAT), MY8-TD (for TASCAM) and other common digital formats. Also compatible with AD/DA cards.
- Analog input/output capability by mounting AD cards MY4-AD and MY8-AD24, ~~and~~ DA card MY4-DA.
- Up to 8 IN/8 OUT per card.
- 4U-size rack mount type.
- 2 digital I/O jacks for use with DSP1D. Compatible with multiple DSP1D (-EX) connections using port select switch.

AD DA CARD MY8-ADDA96

- AD DA card supports 24-bit/8-channel analog inputs and outputs.

INPUT SECTION

The input section of CS1D is equipped with controllers for 48 channels of mono input and 4 channels of stereo input. Each input is built with a 100mm full-stroke motor drive fader for visible confirmation and controlling of all channels at all times without having to enter different layers*. Key controllers such as level, pan and selected channel mix bus send are built into each channel module. Furthermore, all input channels are equipped with a variety of functions including delay, compressor, noise gate and 4-band full PEQ + HPF.

The controllers of each function and the send level to the 48 mix buses are reflected in all of the selected input channels for analog-like feel without the need of an LCD screen.

*For DSP1D-EX, layering is made to mono inputs 1 ~ 48/49 ~ 96 and stereo inputs 1 ~ 4/5 ~ 8 by the global layer switch.

INPUT PATCH

PM1D Version 2 is compatible with digital patch input up to an enormous 320 channels in total. With processing input of 56 channels (112 channels for DSP1D-EX) available to be freely assigned, a flexible system structuring is possible. Patches can be set by simply clicking the grids displayed on the LCD screen and patching of a single input to multiple channels is also possible. Advanced tasks such as EQ processing by separating the vocals to the front main and fold back are also easy. The patch screen utilizing an easy-to-view list structure allows intuitive grasping of the patch setting status at a glance by displaying the unit name for each port, etc. Other than saving/loading patch settings to patch libraries and recalling various patch settings instantaneously, they can also be linked to scene data. Patch screens such as INSERT IN are also available for selecting the insert point of any channel by viewing the block diagram on the screen.

Direct Out from Pre High-Pass Filter

VERSION 2

The PM1D allows you to select ~~points from~~ insert in, insert out and direct out while looking at the block diagram on the patch screen. From Version 2, pre high-pass is added to the direct out point. This supports usage that sends the input signal through a direct out to a recorder or another device without feeding it through an HPF, delivering more power for needs such as live recording. Plus, another function has been added to allow adjustment of the direct out level.

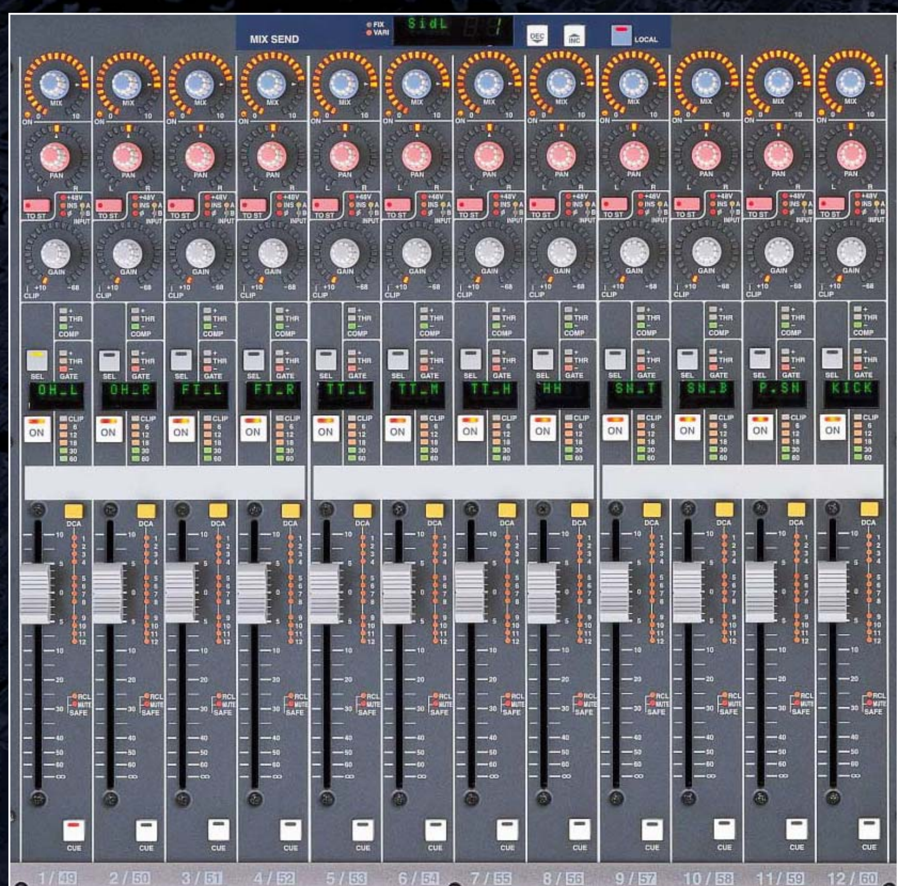
INPUT CHANNEL SECTION

MIX SEND

A controller for setting the level to the selected mix bus (1 ~ 48) is available for each input channel. The mix bus is selected using MIX SEND at the upper portion of the panel and the number and short name are displayed on the indicator. The LOCAL switch is used to select whether to synchronize the selected mix bus using the 4 input blocks or to set it independently. When LOCAL is selected, mix bus selection can be made for the corresponding module alone.

PAN

Pan for sending to STEREO bus or GROUP bus. Assigning to the STEREO bus is set on/off using TO ST. In addition to the normal pan that raises the level by 3dB when panning completely to L or R, "PAN NOMINAL position" that keeps the level at 0dB even when panning completely is also provided. In the LCR PAN mode, CSR (Center-Side Ratio) setting is also available.



+48dB/PHASE/INSERT

Displays the supply of phantom power, insert on/off, phase inversion of input signals, and A/B input during use of mic/line card.

GAIN

Adjusts the head amp gain and displays CLIP.

COMP/GATE

Displays the operation status of compressor and noise gate built to each channel.

SEL

Selects the channel to be used as the selected input channel during startup. Setting of stereo pair is also possible. All parameters with the exception of pan and delay are synchronized for channels set as pair.

NAME

Displays the short name of the input channel for grasping of the input source at a glance.

METER LED

Displays 6 points of CLIP, -6, -12, -18, -30 and -60. The level detection position can be selected from PRE ATT., PRE GATE, PRE FADER and POST FADER.

ON

Sets each input channel on/off.

INPUT FADER

Equipped with a 100mm full-stroke motor drive fader. Using the fader flip function convenient for when creating a monitor mix, any mix bus send can be started on the fader side.

DCA ASSIGN

Assigning to DCA groups 1 ~ 12 can be set for each channel.

RECALL/MUTE SAFE

Setting status of recall safe and mute safe can also be checked for each channel.

CUE

PFL and AFL can be selected for CUE and PRE PAN and POST PAN can be selected for AFL. CUE on/off can be checked using an LED.

STEREO INPUT CHANNEL SECTION

STEREO IN STATUS

Displays which channel of L and R is selected.

MIX

Sets the send level to the mix bus. On/off can be checked using an LED.

ST IN PAN

The orientation of signals sent from the stereo input to the STEREO bus can be independently set for the left and right. PAN NOMINAL position for keeping the level at 0dB when panning completely to the L or R is also available.

+48V/PHASE/INSERT

Displays the supply of phantom power, insert on/off, phase inversion of input signals, and A/B input during use of mic/line card.

GAIN

Adjusts the head amp gain and displays CLIP.

COMP/GATE

Displays the operation and on/off statuses of compressor and noise gate. LED.

SEL

Selects the channel to be used as the selected input channel during startup.

NAME

Displays the short name of the stereo input channel.

METER LED

Displays 6 points of CLIP, -6, -12, -18, -30 and -60. The level detection position can be selected from PRE ATT., PRE GATE, PRE FADER, POST FADER and POST ON.

ON

Sets each stereo input channel on/off.

STEREO INPUT FADER

Equipped with a 100mm full-stroke motor drive fader. Using the fader flip function convenient for when creating a monitor mix, any mix bus send can be started on the fader side.

DCA ASSIGN

Assigning to DCA groups 1 ~ 12 of each stereo input channel can also be performed from this module.

* Phantom power supply, head amp gain control are available for input channels from mic/line card LMY2-MLAB, and LMY4-MLF. A/B input switching is only available for LMY2-MLAB.

SELECTED INPUT CHANNEL SECTION

Parameters of all functions built into the input channels and stereo input channels are reflected in the selected input channel module by the SELECT key of each channel.

+48V/PHASE/INSERT

Sets phantom power supply, phase inversion and channel insert on/off.

INPUT

Selects the input of the mic/line input card A/B terminal.

DELAY

Input delay is available for all input channels and can be applied to various uses such as distance compensation between the audio source and microphone and time difference compensation between microphones placed apart. The delay time can be quickly input using the encoder. Delay time covers range of 0ms ~ 250ms.

MIX SEND

Sending to mix buses are made to 2 layers of MIX 1 ~ 24 and MIX 25 ~ 48. Flipping is performed using the MIX SEND LAYER switch. Send level (-∞dB ~ +10dB) to a VARI type mix bus can be set. Short name of each mix bus is also displayed. 2 types of VARI (mix bus send of variable send level) and FIX (mix bus send of fixed send level) can be selected for every 2 channels for each mix bus. In the case of VARI type mix bus, the SEND point can be selected using the POST FADER/PRE EQ and PRE FADER.

COMPRESSOR

Compressor is available for all input channels. When COMP/EXPANDER type is selected, the ratio is 1:1 ~ ∞:1 and when COMPANDER is selected, the ratio is 1:1 ~ 20:1. Settings such as ATTACK (0 ~ 120msec), RELEASE (5 ~ 42.3msec/during 48kHz operation) and WIDTH/KNEE are available. Linking of compressors for channels available as stereo pairs can be performed. GAIN is set within the range of 0 ~ +18dB and THRESHOLD is set within the range of -54dB ~ 0dB. Input filter for HPF/LPF switching is also built in preceding the compressor.

NOISE GATE

Noise gate is equipped on all input channels. For the noise gate, HPF and LPF capable of simultaneous use for key-in signals are available. Level check and cueing of key-in signals can be performed. The noise gate can be set with ATTACK (0 ~ 120msec), DECAY (5 ~ 42.3msec), HOLD TIME (0.02 ~ 1.96msec*), RANGE (-70dB ~ 0dB) and THRESHOLD (-54dB ~ 0dB). HPL and LPF that can be set with the cut-off frequency between 20Hz and 20kHz for the key-in source are also available. In addition, noise gates for channels available as stereo pairs can be linked.

* During 48kHz operation

INPUT SECTION

EQ
4-band full parametric EQ of HI, HI MID, LOW MID and LOW is available for the EQ. Each band can perform EQ processing with a wide variable frequency point range of 20Hz ~ 20kHz and selection of peaking/shelving settings are available for the HI and LOW bands. HI can also be used as LPF. Q can be set within the range of 10.0 ~ 0.10 and GAIN within the range of -18db ~ +18dB. Other than a 4-band P-EQ, HPF is also available for selection of the cut-off frequency between 20 ~ 600Hz and the slope from 6dB/OCT, 12dB/OCT and 18dB/OCT. Furthermore, channel names are displayed on the LCD screen for high usability.

STEREO/GAIN/ATTENUATION
Equipped with a fixed mix pan switch for selecting PRE PAN/POST PAN of signals to a fixed-type mix bus. Using the TO ST switch, sending to the stereo bus can be turned on/off as well as panning. Also available are GAIN setting of the head amplifier of the mic/line card (-68dB ~ +10dB/ with CLIP indicator) and attenuation in the digital area after AD conversion (-96dB ~ +24dB).

FADER
Equipped with a 100mm full-stroke motor drive fader. CLIP indicator, 6-point LED meter and CUE (PFL, AFL/PRE PAN and AFL/POST PAN can be selected) are also available.

DCA
The selected channel can be assigned to a DCA group. A channel can also be assigned to multiple DCA groups as well.

SAFE
Recall safe (state independent from recalling of scene) and mute safe (state independent from mute group) can be set/cancelled.

CHANNEL SELECT
Channel selection can be made from a selected input selection. Channels can be quickly selected using SHIFT key, INC and DEC key. INC/DEC key can be pressed while holding down the SHIFT key to shift 12 channels at a time. CHANNEL COPY allows the parameters of any channel to be copied to another channel instantaneously for increase in work efficiency.

UTILITY FUNCTIONS FOR INPUT SECTION

SURROUND MODE
Compatible with surround mode of 3-1ch, 5.1ch and 6.1ch. For the surround pan, parameter settings of L - F, FDIV, R - F, RDIV (6.1 only) and LFE (5.1/6.1 only) are possible using the 5 mix send encoders of the selected input. Furthermore, surround pan settings can also be performed using the track pad, mouse, cursor and MIDI control change message.

AUTOMATIC GAIN ADJUSTMENT VERSION 2
When using PM1D for FOH (Front Of House) and Monitor in conjunction with an input device, even if the master PM1D, which has the head amp control authority, changes the gain, the HA slave PM1D ~~instantly and automatically adjusts the attenuator and maintains~~ the level.

FREE ASSIGN FOR 4 INPUT BLOCKS
Layouts of input channel assigned to INPUT blocks 1 ~ 4 of CS1D can be freely specified. Furthermore, layouts can also be freely specified using combination of ST INPUT blocks 1-2, 3-4, 5-6, and 7-8.

GLOBAL PRE/POST, ON/OFF
Global PRE/POST and global on/off to the mix bus are possible from the input channel. In addition, the mix send level and pan/balance of any channel can be applied to other channels or mix send.

EFFECT SECTION

PM1D restructured as a network system with all of the elements necessary for live mixing employs onboard processors and effectors that were outboards up to now for unparalleled space efficiency.
Since all tasks are processed in the digital area, deterioration of sound from repetition of AD/DA conversion and increase in latency are eradicated.
In addition to the 8 cutting-edge 32-bit multi-effect processors built with numerous programs to respond to any application, PM1D is equipped with 24 high-precision 31-band graphic EQs indispensable for live mixing.
The 4 notch filters built to each EQ can be inserted freely at any point.

MULTI-EFFECT PROCESSOR
- 8 cutting-edge 32-bit stereo I/O multi-effect processors. Input and output to effects can be freely patched and functions such as channel insert and send return can be used for various applications.
- Effect types and parameters are displayed using illustrations and knobs to realize a GUI for intuitive operations. Input, output and effect library can be selected from the popup menu for easy operation.
- The effect library is preset with practical effect programs that respond to various uses. Edited programs can be saved in the program library.
- Built-in TAP for parameters. Delay time settings and parameter settings of modulations can be performed quickly using GPI.
- High-precision HPF (thru ~ 8kHz)/LPF (50Hz ~ thru) preceding effects.
- 32-bit full-digital processing with absolutely no effect on the audio quality is realized.
- Built-in multi-band dynamics processor.

31BAND GRAPHIC EQ
- 24 groups of 31-band (mono-in/mono-out) G-EQ are available. Each G-EQ is built with 4 notch filters and use of notch filter only is also possible. Processes that were used on outboards in the past can now be freely used in the digital area of the PM1D.
- Free insertion to any location for responding to various applications from use in input to output such as fold back.
- 31-band faders and EQ graphs are graphically displayed on the LCD screen. Displaying of levels by band using the spectrum analyzer is also possible.
- LIMIT for setting the maximum variable width and variable direction to ±15dB, ±12dB, ±6dB and -24dB and EQ FLAT for spontaneously resetting all faders 0 are also available.
- EQ settings can be saved to and loaded from the library. Original settings can be called at an instant.
- In addition to cursor and mouse operations on the LCD screen for the EQ, adjustment using the DCA fader is also available. Assigning to the fader can be made in unit of 12 bands.

Built-In Standard Add-On Effects (VCM and REV-X) VERSION 2

From the Add-On Effects series compatible with its ~~sister~~ model PM5D and the ~~DM2000 Version 2 Production Console~~, the PM1D Version 2 comes equipped with the advanced reverb REV-X using new algorithms and the VCM (Virtual Circuit Modeling) effects COMP260 (S)*, COMP276 (S)*, EQ601, and Open Deck. Just like the effects installed in the main unit, these Add-On Effects offer totally seamless operability.

*The COMP260 (S) and COMP276 (S) are stereo versions of the COMP260 and COMP276, respectively.

For each Add-On Effect added, 2 of the 24 GEQ included in the PM1D V2 main unit stop functioning.

[VCM (Virtual Circuit Modeling) Effects]
VCM Effects provides simulation of analog equipment using Yamaha's original VCM Technology, which uses modeling of analog equipment circuit and magnetic characteristics on a component level to precisely recreate the characteristics of analog equipment, right down to a sense of saturation. The COMP260 (S), COMP276 (S), EQ601, and Open Deck effects thoroughly recreate vintage model compressors, equalizers, and tape decks that were created in the '70s and are still in demand today. Open Deck offers four models: Swiss '70, Swiss '78, Swiss '85, and America '70, and users can freely switch between recording and playback decks. Even the sound changes originating from the aging of actual tape can be recreated.


COMP276


OPEN DECK

[REV-X]
REV-X, which has already been included in the SPX2000 Digital Multi-Effects Processor, is an all-new algorithm reverberator receiving high recognition. It features rich reverberation with impressive density and smooth decay.

OUTPUT SECTION

The output section is equipped with 48 mix bus outputs, 24 matrix outputs and 2 stereo outputs.
A total output of 76 channels is possible for responding to any need in the live mixing scene. All output channels are equipped with a compressor and output delay and the EQ is built with a powerful 6-band full P-EQ. These variety of functions are reflected in the selected output channel for analog-like control without the need of an LCD screen.
Also equipped are 12 DCA group master faders that inherit the conventional VCA controls. In addition to grouping input, 4 faders of 9 ~ 12 are capable of grouping output as well.

OUTPUT PATCH

Digital patch output of an enormous 192 channels can be freely patched on the screen for flexible system structuring. Patches can be set by simply clicking the grids displayed on the LCD screen and parameter output can also be easily set. The display of unit names allows intuitive grasping of the patch setting status. Other than saving/loading of patch setting to the patch library, various patch settings can be recalled at an instant. Furthermore, link setting to scene data is also possible.

MATRIX OUTPUT SECTION

The master volume of the 24 matrix outputs are expanded to 2 layers, with 12 matrix outputs to each layer. For the matrix output, mix buses 1 ~ 48, stereo A/B and SUB IN can be assigned. For all outputs, separate mixing using the matrix is possible.

ON/RCL/MUTE SAFE

Displays the channel on/off setting and recall safe/mute safe status.

PAIR

Displays the pairing status of channels arranged even-odd.

LEVEL/BAL

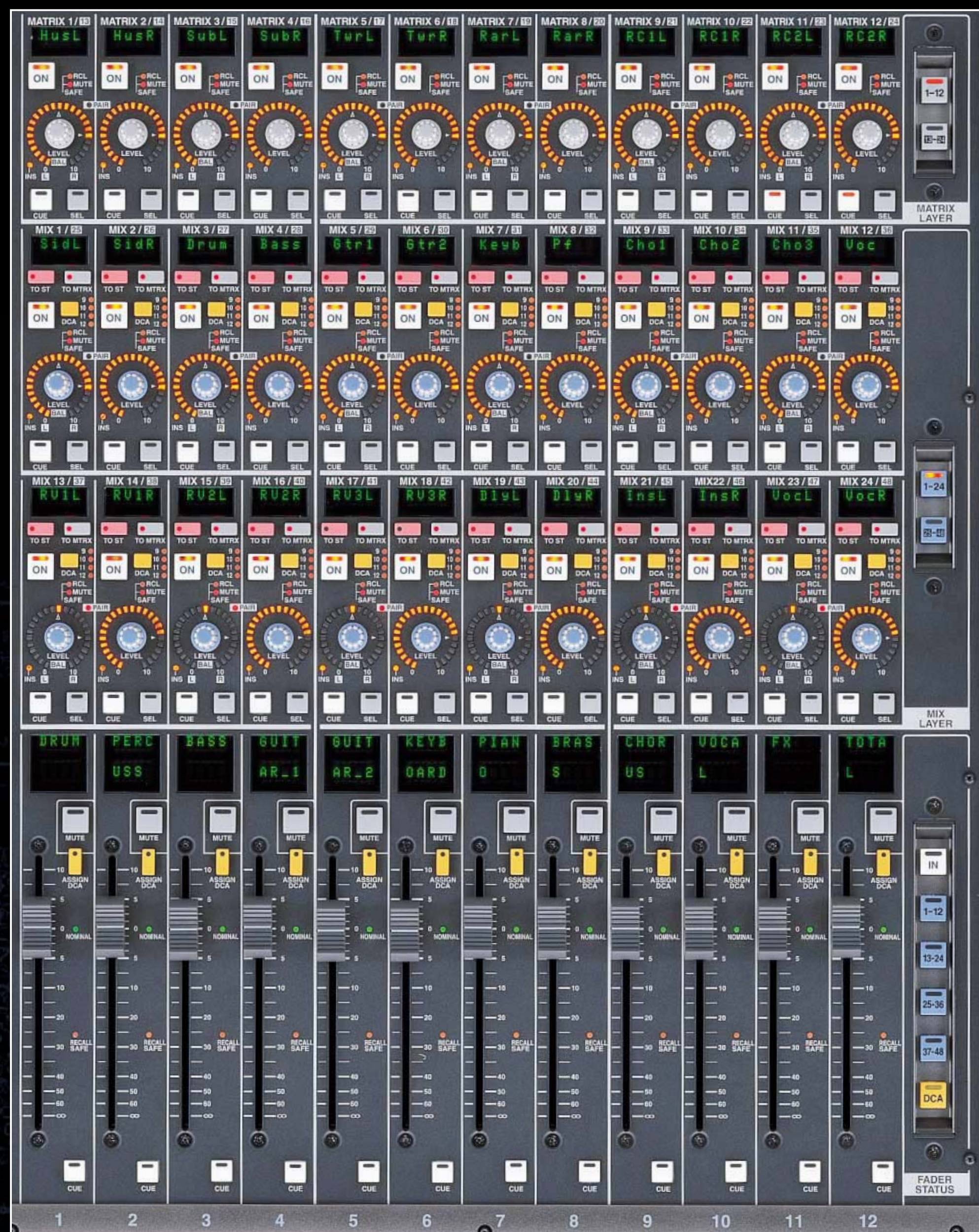
Sets the output level of the matrix master. Setting range is -∞dB ~ +10dB.

INS/CUE/SEL

Matrix also allows insert and cue. SEL is for the selection to the selected output channel module. Pair setting is also possible using the SEL key.

MATRIX LAYER

Switches matrix layers between 1 ~ 12 and 13 ~ 24. Matrix also allows insert and cue. SEL is for the selection to the selected output channel module. Pair setting is also possible using the SEL



MIX BUS OUTPUT SECTION

48 mix bus out masters are expanded to 2 layers. Signal sending, levels and DCA group assign can be controlled directly.

TO ST/TO MTRX/ON/DCA

Performs setting of the stereo bus from the mix bus, on/off to matrix outputs, mix bus output master and assigning to the output DCA group. DCA group assign is a switch that turns on/off DCA groups 9 ~ 12 with the DCA group master switch turned on.

RCL/MUTE SAFE

Displays the recall safe/mute safe setting status.

PAIR

Displays the pairing status of 2 mix bus out channels arranged even-odd.

LEVEL/BAL

Sets the output level of the mix bus out master. Setting range is -∞dB ~ +10dB. Shortcut for controlling the send level from the mix channel and stereo A/B channel to the matrix bus using the level encoder is also available.

INS/CUE/SEL

Displays the insert status of the mix bus out channel. Also equipped with cue of the mix bus. SEL is used for the selection to the selected output channel module. Pair setting can also be performed using the SEL key.

MIX LAYER

Switches mix layers between 1 ~ 24 and 25 ~ 48.

DCA GROUP SECTION

The DCA group section inherits the usability of the conventional VCA. 12-channel grouping can be performed flexibly and grouping of output can be made for the 4 channels of 9 ~ 12 for responding to new applications such as grouping monitor. The DCA fader can be used as the controller for any input channel/mix bus channel using the FADER STATUS switch as well as the controller of the built-in 24 groups of 31-band GEQ.

MUTE/ASSIGN DCA

Selects DCA mute or the input channel/output channel to be assigned to DCA.

Mute indicator function of DCA

When a certain DCA group is muted, [ASSIGN DCA] LED of the channel belonging to that DCA flashes.

DCA FADER

Equipped with twelve 100mm full-stroke motor drive faders.

1.	Input channel of INPUT block selected with the [SEL] switch
2.	Mix channels 1 ~ 12
3.	Mix channels 13 ~ 24
4.	Mix channels 25 ~ 36
5.	Mix channels 37 ~ 48
6.	DCA groups 1 ~ 12
7.	Boost/cut of each band of the graphic EQ (31-band EQ can be split in units of 12 bands for control)

NOMINAL/RECALL SAFE/CUE

Displays the setting status of nominal level and recall safe. Cue can also be performed.

STEREO OUTPUT CHANNEL SECTION

Equipped with 2 stereo outputs with each built with a 100mm full-stroke motor drive fader.
*Only 1 stereo bus available.

ON/TO MTRX/MONO/SEL/INS

Performs channel on/off, sending to matrix, mono/stereo switching of output (STEREO B only), selection to selected output channel module and insert status display.

RCL/MUTE SAFE

Displays the setting status of recall safe/mute safe using indicators.

STEREO OUTPUT FADER

Equipped with two 100mm full-stroke motor drive faders. The variable range is -∞dB~ +10dB. Cue can also be performed.

SELECTED OUTPUT CHANNEL SECTION

All mix bus out channels and functions built in the matrix out channels are reflected in the selected output channel by the select key of each channel.

DELAY

Output delay is available for all channels and can be used for various applications such as distance compensation of speakers and orientation compensation. The delay time covers range of 0ms ~ 1000ms.

COMPRESSOR

Compressor is available for all channels. When COMP/EXPANDER type is selected, the ratio is 1:1 ~ ∞:1 and when COMPANDER is selected, the ratio is 1:1 ~ 20:1. Settings such as ATTACK (0 ~ 120msec), RELEASE (5 ~ 42.3msec/during 48kHz operation), WIDTH/KNEE are available. Linking of compressors for channels available as stereo pairs can be performed. GAIN is set within the range of 0 ~ +18dB (-18 ~ 0dB when COMPANDER is selected) and THRESHOLD is set within the range of -54dB ~ 0dB. Filter for HPF/LPF switching is also built in preceding the compressor.

EQ

6-band full parametric EQ of HI, HI MID, MID, LOW MID, LOW and SUB LOW is available for the EQ. Each band can be set within a wide range of 20Hz ~ 20kHz and bypassing by band is available. Switching of peaking/shelving is available for the HI and SUB LOW bands and HI can be used as LPF and SUB LOW can be used as HPF. Q can be set within the range of 10.0 ~ 0.10, F within the range of 20Hz ~ 20kHz and GAIN within the range of -18dB ~ +18dB.

OUTPUT SECTION

Performs insert on/off, sending to stereo bus*/matrix output**, panning and level setting. The level can be set within the range of -∞dB ~ +10dB. In addition, assigning of DCA groups (9 ~ 12) for output and setting of recall safe/mute safe can be performed.
* When stereo bus is selected, [TO ST] switch will not function.
** When matrix output is selected, TO MATRIX switch will not function.

CHANNEL SELECT

Selects the channel to startup in the selected output channel module. Channels can be quickly selected using SHIFT key, INC and DEC key. INC/DEC key can be pressed while holding down the SHIFT key to shift 12 channels at a time. CHANNEL COPY allows the parameters of any channel to be copied to another channel for increase in work efficiency.

SUPPORTS MY8-DA96 CARD AND MY8-ADDA96 CARD

VERSION 2

Supports the MY8-DA96, which offers DA conversion of 8 channels of audio ~~at 96kHz~~, and MY8-ADDA96, which offers AD/DA conversion of 8 channels of audio ~~at 96kHz~~.
* PM1D does not operate in 98-kHz mode.

CONTROL MEMORY SECTION

TALKBACK
Equipped with 2 talkback functions. Other than the TB mic input jack on the panel, additional TB mic input jack is available on the rear panel. Supplying of phantom power supply is also possible.

OSCILLATOR
2 oscillators are built internally for generation of 100Hz ~ 10kHz sine waves, pink noise and burst noise. Output can be made to any bus/output terminal.

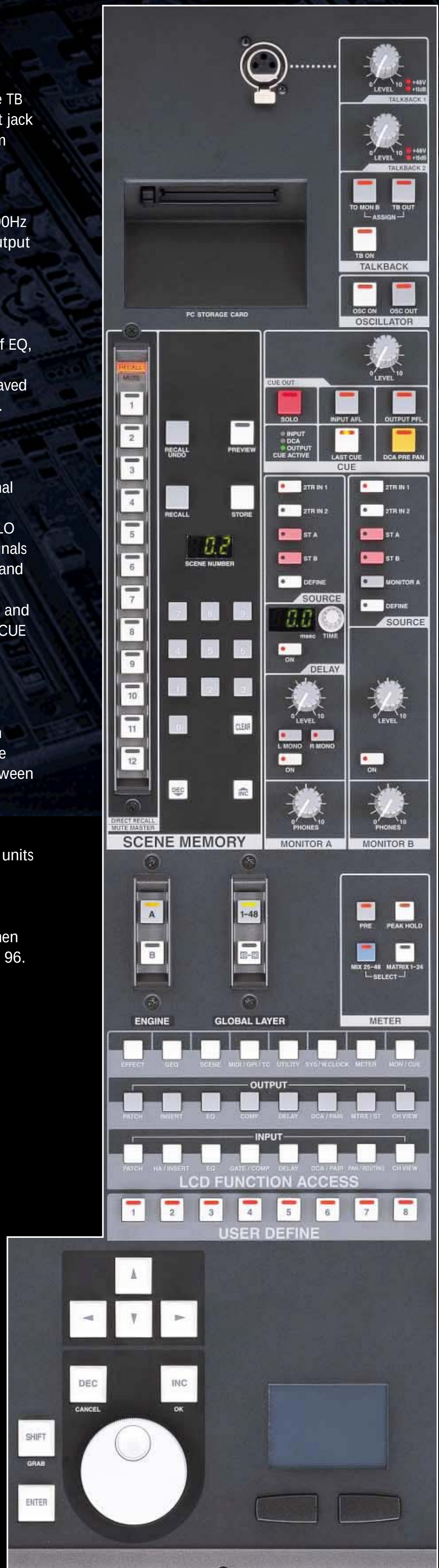
PC STORAGE CARD
2 PC card (Type II) slots are available. Various PM1D settings including scene memory and each library of EQ, dynamics, patches and effects can be saved to and loaded from PC cards. Status settings can also be saved to PC cards and loaded in other CS1D units for use. Work efficiency is drastically increased.

CUE
Sets which position to monitor signals for each signal route of input channels, output channels and DCA groups. Monitor status is displayed using LED. "SOLO mute" for muting other channels when sending signals of a specific channel to each output of mix, matrix and stereo A/B is also available. In addition, "LAST CUE mode" that monitors only the channel selected last and "MIX CUE mode" that mixes all channels with the CUE switch currently set to ON can be selected.

MONITOR
Equipped with 2 monitors of A and B. Monitor A performs distance compensation between the main speakers and the nearby monitor speakers using the built-in delay (0 ~ 750ms). Switching of output between stereo and mono is also possible.

ENGINE
Switch for selecting the engine when connecting 2 units of DSP1D (-EX) using the mirroring function.

GLOBAL LAYER
Layer button when connecting DSP1D-EX. Used when switching input channels between 1 ~ 48 and 49 ~ 96.



METER
Selects the functions built into the level meter such as peak hold. For output, selection of PRE/POST is available. Furthermore, channel (MIX OUT/MATRIX OUT) for display in the 24 meters of the meter bridge can be selected. Meter display of input channels can also be made.

SCENE MEMORY
PM1D can store and recall a total of 1,000 scenes, of which 12 scenes can be assigned to the direct recall keys (1 ~ 12). Other scenes can be selected using the numeric keypad or INC/DEC key. RECALL UNDO is also available for spontaneously restoring the previous settings. PREVIEW mode allows checking of set values without actually switching scenes. For the currently selected scene, number, name and mode (preview) are displayed on the meter bridge. The direct recall key (1 ~ 12) switches can also be substituted as mute master switches of mute groups 1 ~ 12.

LCD FUNCTION ACCESS
Calls any function or screen on the display. The upper section is equipped with keys to call system functions that affect the entire system, the mid-section is equipped with keys to call output functions and the lower section is equipped with keys to call input functions.

USER DEFINE
Using MMC and GPI port, users can select and set as desired such as assigning recorder start/stop.

ADDITIONAL FUNCTIONS FOR USER DEFINED KEYS
VERSION 2

GEQ Select, Tap~~, and~~ Tempo have been added to assignable components for user defined keys. ~~You can access these functions directly with the push of a button.~~

DATA ENTRY
In addition to the cursor keys, INC/DEC key, Shift/GRAB key and Enter key, data wheel for instantly increasing/decreasing parameter values is equipped for speedy data entry. Track pads and left/right pad switches are available as pointing device. Efficient operation is realized by using the track pad to copy channel equalizers using drag & drop, etc. Mouse and keyboard terminals are also available on the front and rear. The keyboard can be used for entering channel names, etc.

The front panel is equipped with mouse and keyboard terminals.

UTILITY

Not only does PM1D inherit the operational system of analog consoles cultivated in the PM series, it also realizes an advanced operational environment. The large color LCD GUI (Graphical User Interface) drastically improves usability. Furthermore, PM1D control software "PM1D Manager" allows offline setting/editing of virtually all PM1D functions on a PC. Data can be sent and received via RS-232C, USB or PC card and preparations for mixing can be performed in advance. In the event that CS1D cannot be used, mixing can be performed by directly connecting a PC to DSP1D (-EX).

COLOR LCD

Equipped with a large color LCD for clear display of operational status. LCD brightness can be adjusted variably for superior visibility even outdoors during daylight. SVGA output terminal is also available for connection of external monitors, etc. The LCD can not only display any screen but can automatically switch screens according to the changed parameters using the auto display function. Confirmation message display can also be turned on/off as desired.

* The auto display function is available for the EQ, delay, gate/compressor, input unit, routing, fader and SOLO/CUE.

GR METER DISPLAY
VERSION 2

The Gain Reduction Meter is displayed for the Gate/Comp of all input channels and the Comp of all output channels.

FADER VIEW
VERSION 2

A new all-inclusive fader display has been installed. Now 48 channel+4 stereo+all DCA can be viewed on one screen, and the display can be switched between CH1-48, ST IN 1-4 and CH49-96, ST IN 5-8.

OPERATION LOCK
2 types of lock functions are available for preventing operational errors. Depending on the usage environment, security settings of parameter lock that prohibits any operation of setup data and console lock that locks all operations using CS1D in the case the operator is away are available.

JOB SELECT
When the [SHIFT] key and [ENTER] key are pressed with the cursor pointing at a parameter on the LCD screen, a list of jobs available for that parameter is displayed in a popup window. Operation superior in usability is realized by allowing jobs to be selected from the popup window.

CASCADE CONNECTION
When using two PM1D systems, sharing of any mix buses and stereo buses is possible by cascading the DSP1D units. Audio and control signals can be transmitted bidirectionally for extremely flexible system structuring and compatibility with large-scale sound systems. When cascading PM1D units, equivalent operations can be performed regardless of master or slave for a remarkable operational environment.

DUAL CONSOLE MODE
Mode for controlling two CS1D units using a single PM1D system. Flexible operations depending on the type of application is possible including the operation of 2 consoles by separating them for main mix and monitor mix, simultaneous operation and sharing of PM1D from both consoles of fixed hall equipment and mobile equipment, and control of 96 channels using 2 consoles on a single layer.

INSERT / DELETE CHANNEL
VERSION 2

The newly added lineup-changing function allows you to change the orders of the input and output channels as you like. Channels in the inserted or deleted range can be picked up and shifted to the end. This also works with any of the Mix, Matrix, and Stereo channels.

GLOBAL CHANNEL COPY
Any parameter of the currently selected channel can be copied to multiple channels at once. Efficient operational environment that can only be realized by digital technology is provided.

CONTROL CHANGE / NRPN
When performing operation other than scene recall, parameters can be sent to an external source as continuous variable data of MIDI. Remote control of internal parameters can also be performed using continuous variable data received from an external source. The same motion can be reproduced any number of times when recording in realtime the changes in parameters using an external MIDI sequencer or sequencing software. 2 types of control change messages ("CONTROL CHANGE" and "NRPN (Non Registered Parameter Number)") can be sent/received.

ADDITIONS TO LIBRARIES & FACTORY PRESET MEMORY
VERSION 2

The previous model offered 100 settings for the Input EQ and Input CH Libraries, but Version 2 offers 200. Also, even more practical additions have been made to the Factory Preset Memory.

UTILITY

GPI

DSP1D (-EX) and CS1D are equipped with a D-sub 9-pin GPI (General Purpose Interface) terminal (serving as GPI IN/GPI OUT) for input and output of control signals up to 8 channels between external devices.

TAP TEMPO

Equipped with a tap-tempo function to specify manually the tempo parameter of the internal effect by tapping the external switch connected to the GPI terminal. Tempo parameters of effectors such as DELAY LCR, ECHO and CHORUS can be edited in realtime via the GPI terminal.

SCENE MEMORY

Advanced scene memory function equipped with various utility functions provides an operational environment with high efficiency and flexibility for any scene.

RECALL SAFE

Recall safe function for excluding specific channels and parameters (ATT, EQ, GATE, COMP, FADER and SEND) when recalling scenes. Recall safe affect all scenes for a global setting.

SELECTIVE RECALL

Selective recall function that allows specification of parameters/channels to recall for each scene. Combination of parameters/channels to recall can be stored for each scene. In addition, "BYPASS" function for batch enabling/disabling of the selective recall function is available.

EVENT LIST VERSION 2

Previously, the scene memory only allowed conversion of scenes in descending order. But new advancements support creation of a scene memory event list. Changing the order of scenes is now much easier than ever.

AUTO STORE OVERWRITE VERSION 2

Library Overwrite Mode can now be selected during Auto Store of scenes.

GLOBAL PASTE

The setting contents of any channel/parameter of the current scene can be copied and pasted to multiple scenes within the scene memory. Changed scene contents can be updated with stored scenes at once.

TRACKING RECALL

When a scene is recalled, offset values set in advance are assigned to each fader value. When a certain fader is offset by -3dB, for example, that fader can be constantly recalled with the value decreased by 3dB for all scenes recalled thereafter.

SCENE LINK EVENT

When a specific scene is recalled, MIDI events can be output from the MIDI OUT terminal. External MIDI devices can be controlled according to the scene recall.

MANUAL FADING

The fade time can be set when switching scenes. When using the manual fading function, the fade time can be operated manually using [DATA] encoder. Convenient for when changing the volume according to the stage progression.

LOAD FILTER / SAVE FILTER VERSION 2

This function manages the PM1D's internal memory more effectively. It supports loading and saving of determined PM1D parameters, as well as loading of selected parameters to determined memory areas and the saving of selected parameters as specified memory data.

BROADCAST FEATURES

The highly acclaimed PM1D in the PA scene with its numerous advanced functions, sound quality, high usability and reliability provides flexibility for responding to various uses. Especially for the field of broadcasting where digitalization is quickly progressing, functions that are indispensable in the broadcasting scene such as mix minus and vertical pair and functions such as P2 REMOTE for heightening the compatibility with video devices are available.

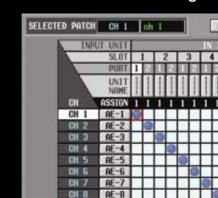
PM1D is already operating in broadcasting stations around the globe with proven results.

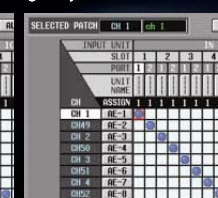
MIX MINUS

Mix minus function for extracting the signals of a specific channel from the signals sent to a VARI type mix bus. Monitors extracted with only the voice necessary for announcements during broadcasting can be created.

VERTICAL PAIR MODE

In addition to the conventional mode (horizontal pairing) that pairs adjacent channels (1+2, 3+4…), vertical pair mode that pairs channels in layers (channel 1+49, 2+50…) is available for pairing input channels. Up to 52 stereo sources (48 mono x 2 + 4 stereo input) can be controlled using a single layer.


Horizontal pair mode


Vertical pair mode

REMOTE terminal

Recorders and locators compatible with protocols of P2, DENON, etc. can be remote controlled via the REMOTE terminal (RS-422, D-sub 9-pin) of DSP1D (-EX) and CS1D.

GPI

DSP1D, DSP1D-EX and CS1D are equipped with a D-sub 9-pin GPI (General Purpose Interface) terminal (serving as GPI IN/GPI OUT) for sending/receiving of control signals up to 8 channels between external devices.

FADER START

When the fader of a channel is raised to -60dB or higher or lowered to -∞, various commands (control signals) can be output from the terminal (GPI/MMS/RS422) assigned in advance. External devices such as GPI-compatible CD players, RS422-compatible video players and MIDI sequencers can be remote controlled using fader operation.

DIGITAL INPUT GAIN, STEREO BALANCE CONTROL

The sensitivity of signals input in the six 2TR IN can be separated to L and R in the digital area for adjustment. Balance parameter is also available for input channels arranged in order of odd/even numbers as well as for the ST IN channels.

OSCILLATOR

The internal oscillator is equipped with LEVEL (-∞db ~ +10db), WAVE FREQ (100Hz ~ 10kHz) and PINK NOISE/BURST NOISE. The burst noise allows time setting at the noise interval. It is also compatible with sending of 2 sine waves. By setting each frequency and level, individual output can be made for odd/even channels.

POST ON / POST TO ST

When sent to 2 mix buses arranged in order of odd/even using the post fader, the sending location of signals can be selected from (POST ON) immediately after the [ON] switch and (POST TO ST) immediately after the [TO ST] switch of the input channel.

PAIR, STEREO TO MONO MIX

nput channels set as pairs and ST IN channel signals can be converted to mono signals. L-MONO, R-MONO and L,R-MONO are available.

CONTROL SURFACE CS1D   MONAURAL INPUT MODULE MIM1D

- "Analog-like" operational environment that allows mixing without viewing the LCD screen.
- Compact size capable of controlling an enormous number of input/output using the layer function and digital patching. Up to 2 layers for a total input of 112 channels can be controlled.
- Intelligent and efficient usability with use of a large color LCD screen, track pads, etc. Mouse and keyboard can also be connected.
- Storing/recalling of mix status using scene memories and libraries.
- Offline setting/mixing using PM1D Manager (for Windows®).
- Superior flexibility with DCA (Digital Controlled Amplifier) capable of I/O section flipping, multiple grouping, etc.
- Variety of functions including cutting-edge multi-effectors, GEQ, etc.

CS1D is structured by a total of 10 sections including 4 types and 7 pieces of modules.

1. **MEMORY / CONTROL section**
2. **METER / DISPLAY section**
3. **INPUT section**
4. **STEREO INPUT section**
5. **SELECTED INPUT CHANNEL section**
6. **MIX BUS OUT section**
7. **MATRIX OUT section**
8. **STEREO OUT section**
9. **SELECTED OUTPUT CHANNEL section**
10. **DCA GROUP section**

POWER SUPPLY PW1D

DC OUTPUT
Supplies power to CS1D.

CONSOLE IN/OUT
BNC terminal for sending/receiving control signals in between two connected CS1D units. For fail-safe operation, 2 I/O terminals are provided. In the case of communication failure for one of the cables, the communication route is automatically switched for continued communication.

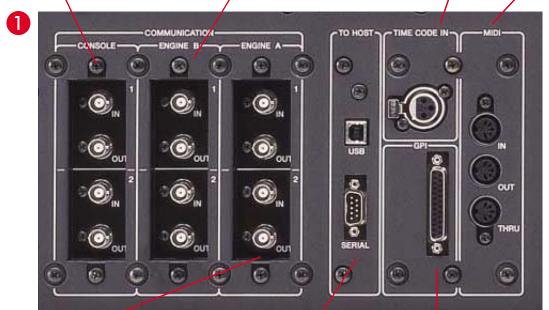
ENGINE B
BNC terminal for sending/receiving control signals between DSP1D (-EX) used as engine B and CS1D when connecting two DSP1D (-EX) units to a single CS1D by using the mirroring function. For fail-safe operation, 2 I/O terminals are provided.

TIME CODE IN
XLR-3-31 terminal for inputting the time code from an external device.

MIDI IN-OUT-THRU
MIDI terminal for connecting MIDI devices.

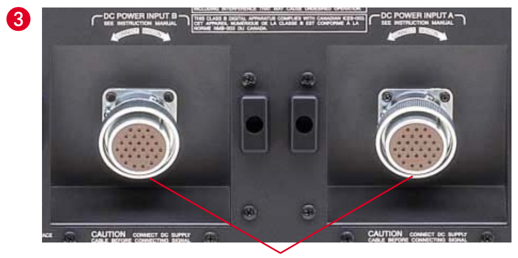
REMOTE
D-sub 9-pin terminal for controlling tape recorders and HD recorders. Connected recorder can be played back and stopped using serial commands.

SVGA OUT
External monitor connection terminal. The same screen displayed on CS1D can be output to an external monitor.



NUMKEY, MOUSE, KEYBOARD
PS/2 compatible numeric keypad, mouse and keyboard can be connected for input/control using the screen.

GPI terminal
8-channel send/receive of control signals with GPI-compatible models. 25-pin D-sub 25-pin terminal.



DC POWER INPUT A/B
Power supply PW1D is connected. For fail-safe operation, up to 2 units of PW1D can be connected, with each supplying 50% of the power when both are in use. If one of the PW1D units fails to supply power, the remaining PW1D unit will supply 100% of the power.

ENGINE A
BNC terminal for sending and receiving control signals between CS1D and DSP1D (-EX). For fail-safe operation, 2 I/O terminals are provided.

TO HOST[SERIAL]
D-sub 9-pin (RS-232C) for communication with PC. PM1D can be controlled using PM1D Manager (for Windows®).

TO HOST[USB]
USB terminal for communication with PC.
PM1D can be controlled using PM1D Manager (for Windows®).

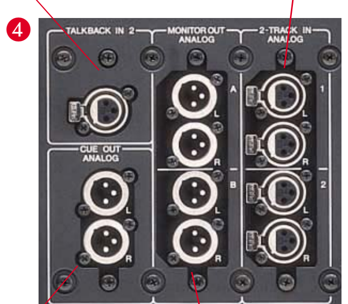
WORD CLOCK IN-OUT
BNC terminal for sending/receiving word clock to an external device. 75Ω ON/OFF switch is used to terminate the word clock connection.

STEREO OUT DIGITAL A/B
Digitally outputs STEREO OUT A/B signals. Servers as both XLR-3-32 terminal for output in AES/EBU format and RCA pin terminal for output in COAXIAL format.

2TR IN DIGITAL
Six 2TR IN terminals for input of digital source from CD players and DATs. Independent SRC is built in for all 6 terminals and XLR-3-31 terminal is also provided. Terminals 1 and 2 also serve as RCA pin and COAXIAL terminals and can be used by switching between 2TR IN ANALOG 1 and 2.

TALKBACK IN 2
In addition to the front panel of CS1D, talkback terminal is also provided on the rear section. Talkback on both front and rear are mixed within CS1D. Phantom power can also be supplied.

2TR IN ANALOG1/2 L/R
Terminal for inputting analog stereo signals from an external source. Signals input from this terminal can be patched to any input channel/ST IN channel using the input patch.



CUE OUT ANALOG L/R
XLR-3-31 terminal for outputting cue signals.

MONITOR OUT ANALOG A/B L/R
XLR-3-31 terminal for outputting signals of monitor A/B.

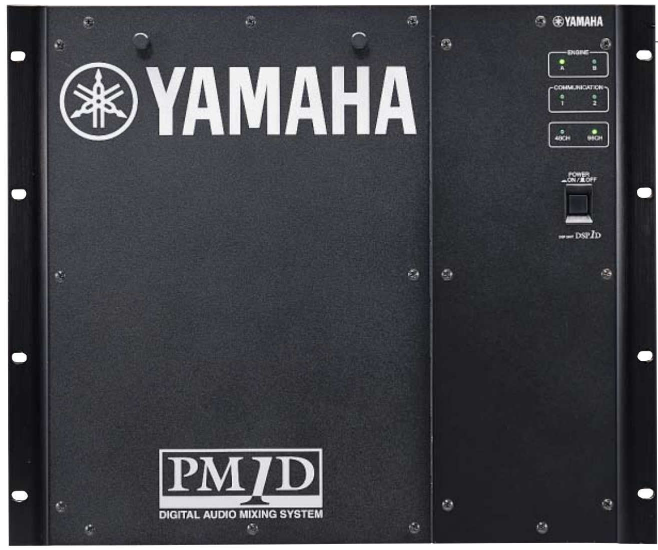
6 LAMP
Accompanied lamps can be connected for working in environment with low visibility.
The lamps can be connected to 4 locations.

DIGITAL I/O

CONSOLE 1/2
68-pin D-sub terminal for sending/receiving digital audio signals such as for monitor between multiple CS1D units. For fail-safe operation, 2 terminals are provided. In the case of communication failure for one of the cables, the communication route is automatically switched for continued communication.

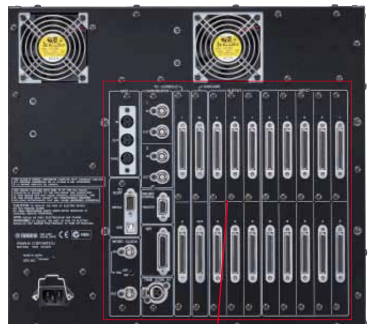
ENGINE A 1/2
68-pin D-sub terminal for sending/receiving digital audio signals such as monitor, talkback, 2TR IN and cue between CS1D and DSP1D (-EX). For fail-safe operation, 2 terminals are provided.

ENGINE B 1/2
68-pin D-sub terminal for sending/receiving digital audio signals between CS1D and DSP1D (-EX) when connecting two DSP1D (-EX) units to a single CS1D using the mirroring function. For fail-safe operation, 2 terminals are provided.

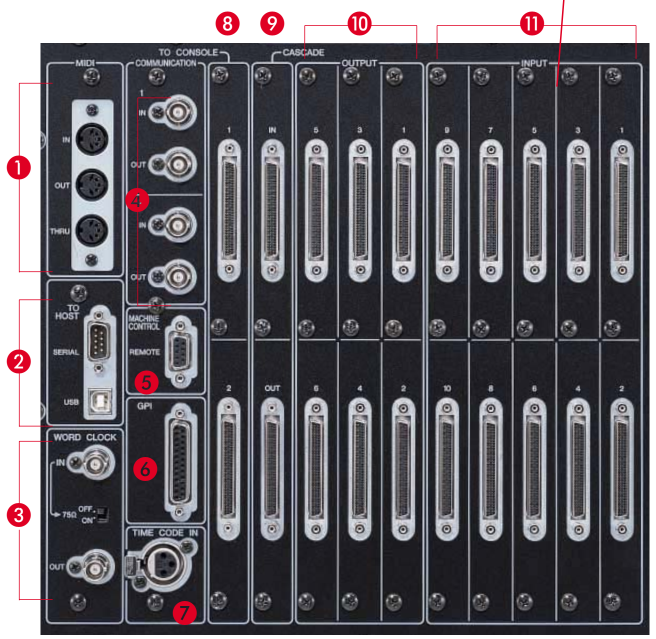


DSP UNIT DSP1D (DSP1D-EX)

INPUT DSP BOARD	EFFECT DSP BOARD	GROUP DSP BOARD
IDB1D	EDB1D	GDB1D
CONSOLE INTERFACE BOARD	ENGINE MANAGEMENT BOARD	PATCH DSP BOARD
CIB1D	EMB1D	PDB1D



DSP1D (-EX) is equipped with a power supply cable lock on the power supply connector section to prevent problems such as disconnection of the power cable, etc.



❶ MIDI IN-OUT-THRU
MIDI terminal for connecting MIDI devices. MIDI messages such as program change can be sent/received between DSP1D (-EX) and an external MIDI device.

❷ PC CONTROL
RS232C: D-sub 9-pin terminal for communication with PC. PM1D can be controlled using PM1D Manager (for Windows®). In the case CS1D cannot be used, mixing can be performed by directly connecting a PC to DSP1D (-EX).
USB: USB terminal for communication with PC. PM1D system can be controlled using a PC.

❸ WORD CLOCK IN-OUT
BNC terminal for sending/receiving word clock to an external device. 75Ω ON/OFF switch is used to terminate the word clock connection.

❹ CONTROL I/O, CONSOLE 1,2 IN OUT terminals
BNC terminal for sending/receiving control signals between DSP1D (-EX) and CS1D. For fail-safe operation, 2 I/O terminals are provided for each terminal.

❺ REMOTE
D-sub 9-pin terminal for controlling tape recorders and HD recorders. Connected recorder can be played back and stopped using serial commands.

❻ GPI
Connects GPI-compatible external devices. External devices can be controlled from PM1D and any function of PM1D can be executed from the external device.

❼ TIME CODE IN
XLR-3-31 terminal for inputting the time code using an external device.

❽ CONSOLE I/O
68-pin D-sub terminal for sending/receiving digital audio signals such as monitor, 2TR IN, talkback and cue between CS1D and DSP1D (-EX). For fail-safe operation, 2 terminals are provided.

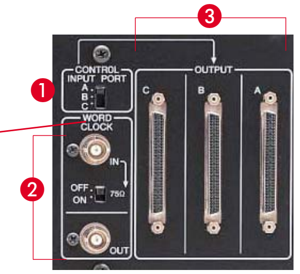
❾ CASCADE
68-pin D-sub terminal for sending/receiving multi-channel digital audio signals by cascading two DSP1D (-EX) units. For fail-safe operation, 2 terminals are provided.

❿ OUTPUT
68-pin D-sub terminal for outputting multi-channel digital audio to AO8 or DIO8 from DSP1D (-EX). 6 terminals are provided for output of up to 192 patches.

⓫ INPUT
68-pin D-sub terminal for inputting multi-channel digital audio signals from AI8 or DIO8 to DSP1D (-EX). 10 terminals are provided for output of up to 320 patches.

ANALOG INPUT BOX AI8

MIC/LINE INPUT UNIT AI8-ML8AB
AI8 can be set with eight LMY2-MLAB mic/line input cards. Compatible up to 16 mic/line inputs.



MIC/LINE INPUT UNIT AI8-ML8F
AI8 can be set with eight LMY4-MLF mic/line input cards. Compatible up to 32 mic/line inputs.

AD UNIT AI8-AD8
AI8 can be set with eight LMY4-AD AD cards. Compatible up to 32 line inputs.

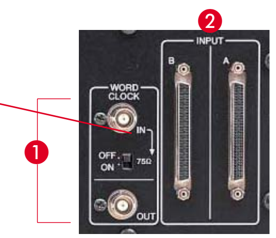
MIC/LINE AD UNIT AI8-ML4AD4
AI8 can be set with four LMY2-MLAB mic/line input cards and four LMY4-AD AD cards. Compatible up to 8 mic/line inputs and 16 line inputs.

❶ CONTROL INPUT PORT
Selects the terminal to receive control signals from DSP1D (-EX).

❷ WORD CLOCK IN
BNC terminal for receiving word clock from an external device. 75Ω ON/OFF switch is used to terminate the word clock connection.

❸ OUTPUT A/B/C
68-pin D-sub terminal for outputting multi-channel digital audio to DSP1D (-EX). The same signals are output to A, B and C for sharing of input signals by up to three DSP1D (-EX) units.
* Photo shown is AI8-ML8AB.

ANALOG OUTPUT BOX AO8

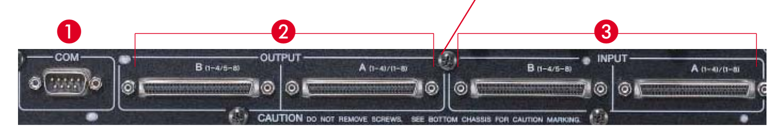


DA UNIT AO8-DA8
AO8 can be set with eight LMY4-DA DA cards. Compatible up to 32 line outputs.

❶ WORD CLOCK IN
BNC terminal for receiving word clock from an external device. 75Ω ON/OFF switch is used to terminate the word clock connection.

❷ INPUT A/B
68-pin D-sub terminal for inputting multi-channel digital audio from DSP1D (-EX). Switching can be made between A and B using the input selector on the front panel. Port switching can be made using CS1D when operating two DSP1D (-EX) units in the mirror mode.
* Photo shown is AO8-DA8.

DIGITAL IO BOX DIO8



❶ COM
D-sub 9-pin terminal for version upgrades.

❷ OUTPUT A
OUTPUT B
68-pin D-sub terminal for outputting multi-channel digital audio to DSP1D (-EX). The same signals can be output from OUTPUT A and OUTPUT B using the port select switch.

❸ INPUT A
INPUT B
68-pin D-sub terminal for inputting multi-channel digital audio from DSP1D (-EX). When outputting the same signals from OUTPUT A and OUTPUT B using the port select switch, input is limited to INPUT A only.

❹ WORD CLOCK IN-OUT
BNC terminal for sending/receiving word clock from an external device. Equipped with 4 word clock outputs. 75Ω ON/OFF switch is used to terminate the word clock connection.

MIC/LINE INPUT CARD
LMY2-MLAB

2-channel analog input card equipped with A/B select switch compatible with mic/line level. Each channel is independent and A/B input switching, gain trimming (-68dB ~ +10dB) and phantom power supply on/off can be controlled from CS1D. With the newly developed built-in head amplifier, richer and fuller sound quality is realized. 28-bit equivalent high-quality sound with dynamic range exceeding 120dB is realized by the AD floating control. Up to 8 cards can be mounted to AI8.

Input Terminals	GAIN	Actual Load Impedance	For Use With Nominal	Input Level		Connector
				Nominal	Max.Before Clip	
CH1A,CH1B CH2A,CH2B	-68dB	3kΩ	50-60Ω Mics & 600Ω Lines	-68dB	-54dB	XLR-3-31 (Balanced)
	+10dB			+10dB	+24dB	

MIC/LINE INPUT CARD
LMY4-MLF

4-channel analog input card compatible with mic/line level. Gain trimming (-68dB ~ +10dB) and phantom power supply on/off of each channel can be controlled from CS1D. With the newly developed built-in head amplifier, richer and fuller high-quality sound is realized with dynamic range exceeding 110dB. Up to 8 cards can be mounted to AI8.

Input Terminals	GAIN	Actual Load Impedance	For Use With Nominal	Input Level		Connector
				Nominal	Max.Before Clip	
CH1-4	-68dB	3kΩ	50-60Ω Mics & 600Ω Lines	-68dB	-54dB	XLR-3-31 (Balanced)
	+10dB			+10dB	+24dB	

AD CARD
LMY4-AD

4-channel analog input card compatible with line level. 28-bit equivalent high-quality sound with dynamic range exceeding 120dB is realized by the AD floating control. Up to 8 cards can be mounted to AI8.

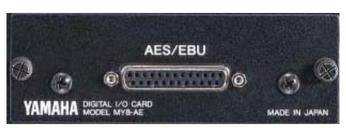
Input Terminals	Actual Load Impedance	For Use With Nominal	Input Level		Connector
			Nominal	Max.Before Clip	
CH1-4	10kΩ	600Ω Lines	+10dB	+24dB	XLR-3-31(Balanced)

DA CARD
LMY4-DA

4-channel analog output card compatible with line level. 28-bit equivalent high-quality sound with dynamic range exceeding 120dB is realized by the AD floating control. Up to 8 cards can be mounted to AI8.

Output Terminals	GAIN switch	Actual Source Impedance	For Use With Nominal	Output Level		Connector
				Nominal	Max.Before Clip	
CH1-4	+24dB	150Ω	600Ω Lines	+10dB	+24dB	XLR-3-32 (Balanced)
	+18dB			+4dB	+18dB	
	+15dB			+1dB	+15dB	

MY CARDS



MY8-AE(AES/EBU)

MY8-AT(ADAT)

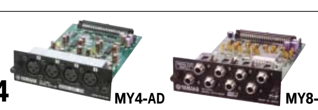
MY8-TD(TASCAM)

DIGITAL I/O CARD(Mini YDGAI CARD)
MY8-AE(AES/EBU)
MY8-AT(ADAT)
MY8-TD(TASCAM)

Digital I/O card compatible with digital formats of AES/EBU, ADAT and TASCAM. Realizing 24-bit resolution, each card is equipped with 8 IN/8 OUT. Up to 8 cards can be mounted to a single DIO8 unit. Mixing of formats is also possible.
For connectors, D-sub 25-pin is provided for MY8-AE, optical x 2 for MY8-AT and D-sub 25-pin (WORD CLOCK OUT x 1) for MY8-TD.

* ADAT is a registered trademark of Alesis Corporation. * TASCAM is a registered trademark of TEAC Corporation.

AD CARD
MY4-AD
MY8-AD24



MY4-AD realizes 24-bit resolution and is compatible with 4-channel AD conversion. MY8-AD24 realizes 24-bit resolution and is compatible with 8-channel AD conversion. For connectors, MY4-AD is equipped with four XLR3-31 terminals. MY8-AD24 is equipped with eight TRS phone terminals. Up to 8 cards can be mounted to a single DIO8 unit.

DA CARD
MY4-DA

DA card capable of 4-channel analog output in 20-bit resolution. For connectors, four XLR3-32 terminals are provided. Up to 8 cards can be mounted to a single DIO8 unit.

AD DA CARD
MY8-ADDA96

AD DA card supports 24-bit/8-channel analog inputs and outputs.

OPTION

Lighting lamp
LA1800

CS1D lighting lamp (up to 4 can be connected to CS1D).

Control cable
Control signal cable. Can be used up to 200m in length.

CC200 (200m) **CC100** (100m)
CC050 (50m) **CC010** (10m)

Digital audio cable
Digital audio cable. Can be used up to 200m in length.

CA200 (200m) **CA100** (100m)
CA050 (50m) **CA010** (10m)
CA003 (3m)

ARTIST 1D
For Yamaha Digital Audio Mixing System PM1D

Artist1D is an optic interface for the PM1D.
It was developed and manufactured by RIEDEL in collaboration with YAMAHA.
Using fiber-optic cable, Artist1D can transmit audio and control signals bidirectionally between DSP1D and CS1D, and send/receive audio signals of 160 channels and a variety of control signals at a range exceeding 500m. In addition, high reliability is also realized.

Rear Panel

Artist1D units placed at FOH and stage positions. Using fiber-optic cable, these units can transfer audio and control signals between CS1D and DSP1D.

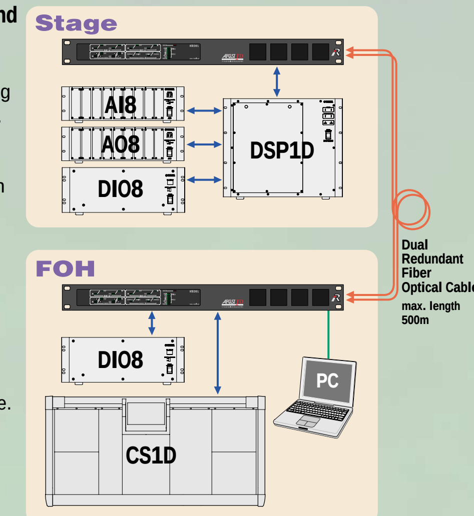
The Artist1D system consists of two units: one located by CS1D (the control surface), the other located by DSP1D (the DSP-engine). These units are connected by duplex LC fiber cable. Using Artist1D, the burden of transporting cables is reduced. Wiring can also be performed with ease. When connected by fiber-optic cable, the two Artist1D units can be placed up to 500m apart.

Bidirectionally-transmitted PM1D audio signals can consist of up to 160 channels

Using fiber-optic cable, up to 80 channels can be transmitted in each direction (send / receive) for a total of 160 channels in original PM1D format. Artist1D also comes equipped with a 68-pin female D-sub half-pitch input connector and equivalent output connector. These are compatible with AI8 analog input boxes, AO8 analog output boxes and DIO8 digital input/output boxes. These are I/O devices that can be easily connected to PM1D.

Long-distance, bidirectional transmission of CS1D/DSP1D, RS232 and word clock control data

Artist1D not only allows bidirectional transmission of audio data. It also allows for transmission of control data between CS1D, DSP1D and I/O units. Long-distance transmission can also be performed in a similar manner for RS232. The PC normally connected to the RS232 terminal of DSP1D can be connected to the RS232 terminal of the Artist1D unit closer to CS1D.The Word clock can also be transmitted via the fiber optic cable. Artist1D can function as a word clock master or slave.



High reliability of Aux fiber optic output, mirror-mode compatibility and dual power supply

In order to provide solid reliability, Artist1D has available two lines of Main and Aux for fiber optic connection. In the event of an accident to one of the cables, transmission will automatically be switched to the other. The use of 2 Artist1D sets allows for mirror-mode compatibility with PM1D for complete redundancy. In addition, Artist1D is equipped with two power supply terminals on the side as standard for redundancy of not only signals but also the power as well.

Artist1D is equipped with two power supply terminals.

Military-type fiber-optic cables that use connector conversion boxes are available in the market. They offer the highest reliability and tolerate even the roughest situations.

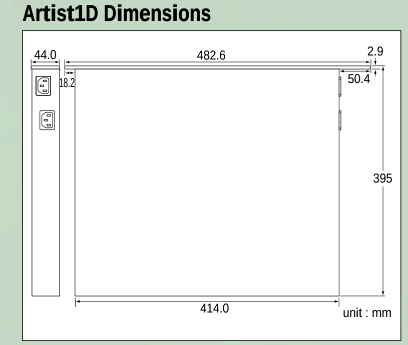
Compact 1U-size body for easy installation / transportation with numerous indicators and wide range of terminals

The compact 1U-size body allows easy installation and transportation and brings many advantages for use in facilities, live concert tours, etc.
The front panel is built with indicators superior in visibility that clearly indicate the operations of Audio Send/Receive, Control Send/Receive, Status of Sync, Network and Warning for easy monitoring of connection status and operation.The I/O section of the rear panel is built with LC type fiber-connectors, word clock input /output, a 68-pin female D-Sub half-pitch connector for connection to I/O-units, DSP1D and CSP1D, and two pairs of BNC sockets for duplex transmission of remote control data between Artist1D and CS1D.
In addition, serial port (9pin D-sub male) provides a transparent RS-232 connection between Artist1D units via fiber link.

Artist1D Specifications

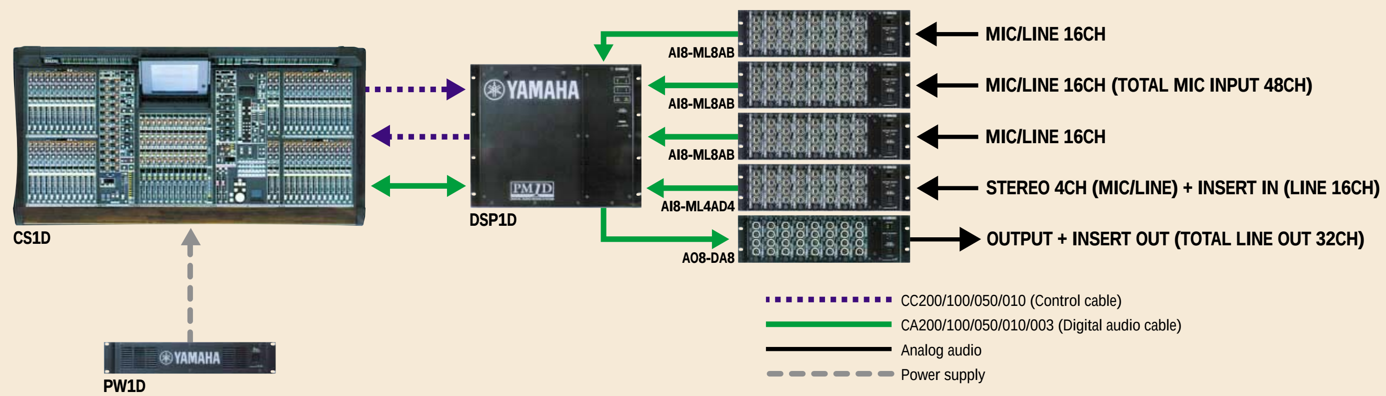
Fiberoptic Link I/O	2x Duplex LC connectors (Main and Aux), 4x 1.250GBits/sec, multimode fiber, 50/125um, fiber length max. 500m.
Audio I/O	6x 68-pin half pitch D-sub connectors, RS422 balanced I/O drivers, cable length max. 20m.
Word Clock I/O	2x word clock output, BNC connectors, TTL, 75Ohm, 48kHz sample rate (deviation max. +/-100ppm when ARTIST 1D is used as a master-clock)
	1x word clock input, BNC connector, TTL, 75Ohm, 48kHz sample rate, (deviation max. +/-200ppm when ARTIST 1D is used as a slave)
Control I/O	2 pairs of BNC connectors, 50Ohm, PM1D control protocol.
RS232	1x 9-pin D-Sub (male), 115.2 kBits/sec
Unit Update	1x RJ-45 connector, 10Base-T for update of Artist1D
Power Supply	Dual redundant power supplies, 100V - 240V AC, 47Hz - 63 Hz, 80W
Dimensions	19" rack-mountable, 1U, 44mm x 483mm x 395mm. [H x W x D]
Weight	9.5kg

Artist1D Dimensions



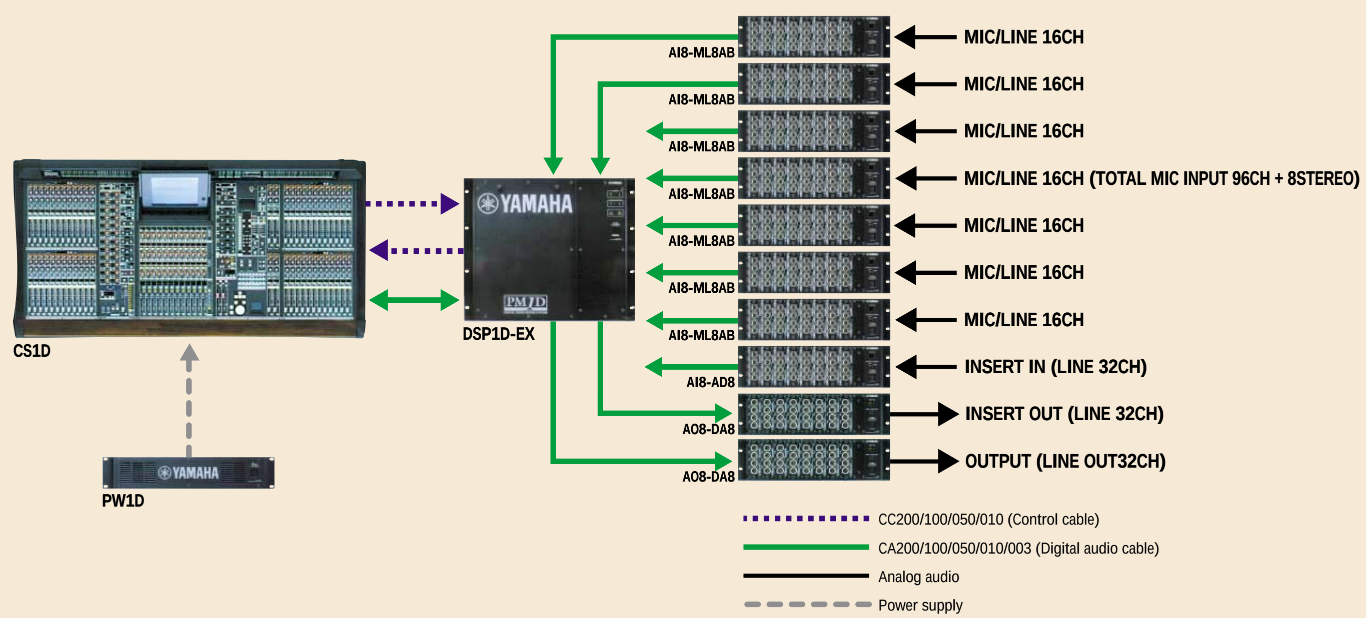
Riedel is a company based in Germany that designs, manufactures and distributes radio and intercom technologies for the broadcast, pro-audio, event, sports and theater industries. Riedel also provides complete radio and intercom services, event accreditation and ticketing systems as well as fiber-based audio and video transmission systems to worldwide events including high-profile events like Olympic Games, World Championships, Super-Bowl and Formula 1 races.
for details, see: **www.riedel.net**

48CH SYSTEM



Basic 48-channel system. Three AI8-ML8AB mic/line input units are used for mic/line input of 48 channels in total. Using mic/line/AD unit AI8-ML4AD4, 4 stereo inputs and 16 line inputs for insert-in are provided. Including insert-out, a total of 32 outputs are provided by the DA unit AO8-DA8.

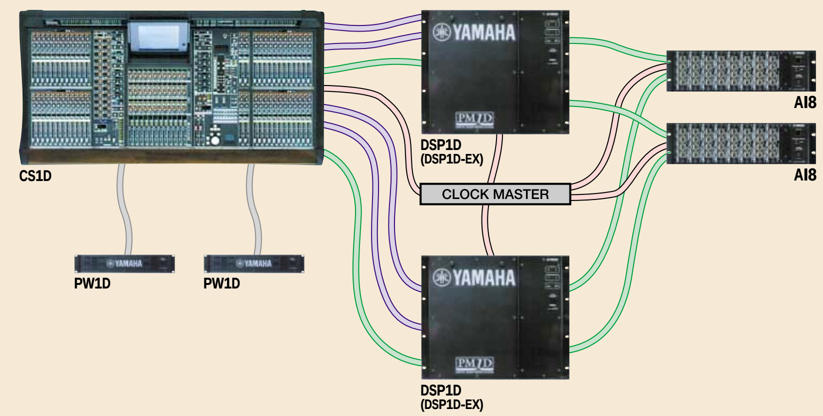
96CH SYSTEM



The 96-channel system uses seven AI8-ML8AB mic/line input units for input. Mic/line input of 112 channels is available and one of the units is input to 8 stereo inputs. AD unit AI8-AD8 is used for insert-in, providing line input of 32 channels in total. Furthermore, DA unit AO8-DA8 is used for insert-out, providing line output of 32 channels in total. AO8-DA8 is also used for output, providing output of 32 channels in total.

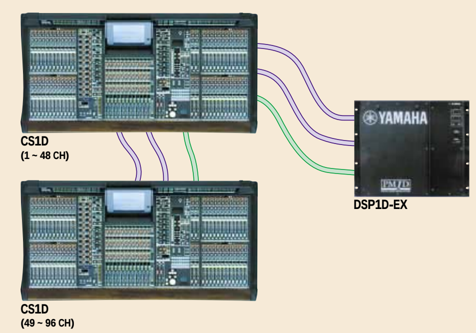
PM1D allows flexible system structuring superior in scalability. By connecting multiple CS1D and DSP1D (-EX) units, a wide range of system upgrades that respond to various needs for house, monitor, etc. are possible including large-scale PA systems compatible with enormous number of input/output, systems with emphasis on safety, systems that give priority to intuitive operations without the use of layers, and systems that use multiple CS1Ds for house, monitor, etc.

Fail-safe system using mirroring



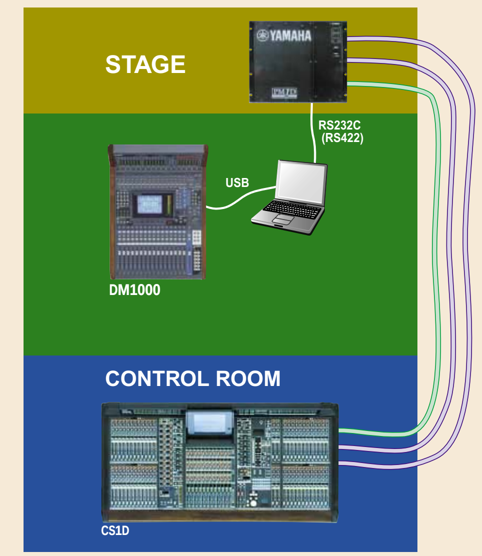
Complete fail-safe operation is realized using a clock master and setting two DSP1D (-EX) units to the mirror mode in the event one of the DSP1D units fails to operate. Complete attention has been given to PM1D concerning safety. CS1D and DSP1D, for example, can be connected with 2 sets of cables for both control signals and digital audio signals and switching is made automatically in the case of communication failure in one of the cables. Furthermore, 2 units of power supply unit PW1D can be connected to CS1D for 50% power supply from each unit under normal operation. In the event of failure to one of the units, 100% power is supplied by the other unit.

96-channel, non-layered system using two CS1D units



System that controls input of 96 channels + 8ST from DSP1D-EX without using layers. By using two CS1D units, all input are expanded to the fader to realize speedy control and superior visibility of all input levels of 96 channels + 8ST and panning displayed as lists. For fail-safe operation, 2 CONSOLE IN/OUT terminals are also provided.

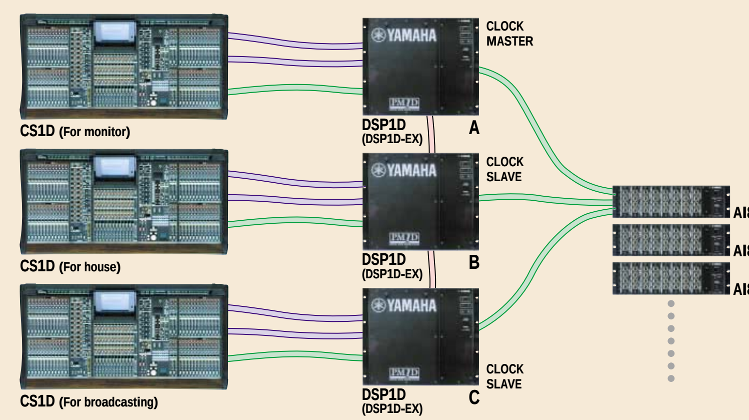
Remote Control System with Compact Digital Mixer Positioned Close to Stage



When the CS1D is positioned apart from the stage, such as in a control room, the DM1000 ~~or another device~~ can be set up as a remote controller in the Front-Of-House position to control the PM1D system. A laptop computer, with the PM1D Manager activated on it, is placed right next to the DM1000, and the DM1000 is connected to the laptop computer with a USB cable. The laptop computer is then connected to the DSP1D (DSP1D-EX) positioned at the stage side, or the CS1D set up in the control room using RS422*.

* PM1D is equipped with RS232C terminal. A conversion device is required when using the RS422.

System that shares the audio source from AI8 using three DSP1D units



AI8 is equipped with 3 outputs of A, B and C and the same signal is output from each output terminal. Three independent mixing systems for house, monitor and broadcasting can be structured by connecting three DSP1D units to each terminal.

* In the case of this application, one of DSP unit A, B and C will be the master for AI8 control (GAIN, A/B and phantom on/off of LMY2-MLAB card). If the engine for monitor is the master, for example, above control will not be available from DSP1D for house and DSP1D for broadcasting.

CC200/100/050/010 (Control cable)
CA200/100/050/010/003 (Digital audio cable)
WORD CLOCK
Power supply