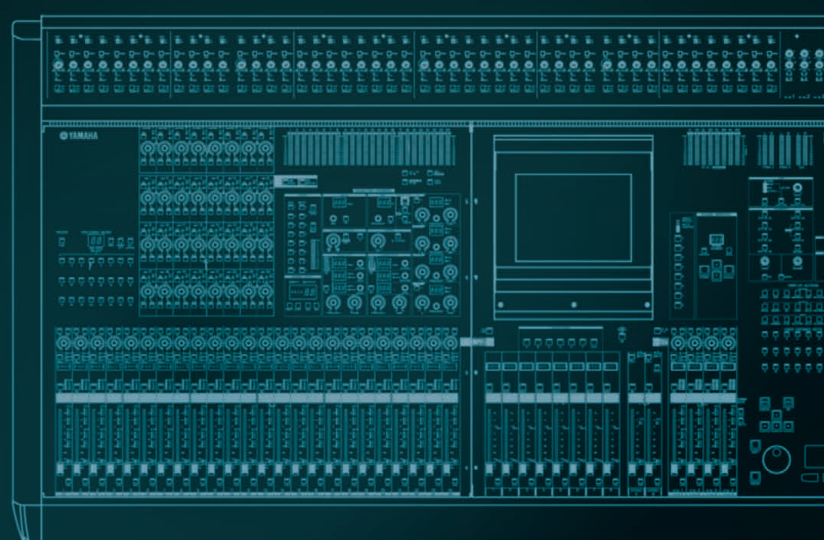
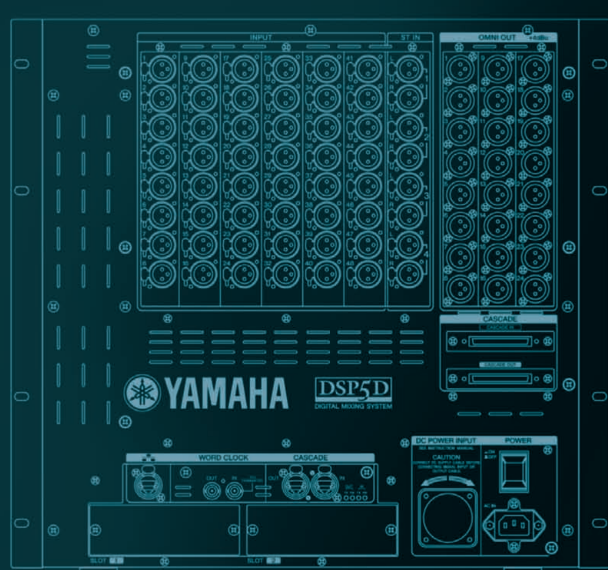
YAMAHA

PM5D
DIGITAL MIXING CONSOLE
Version 2

DSP5D
DIGITAL MIXING SYSTEM



PM5D V2
New Function Guide



We added new functionality to the PM5D digital mixing console, and now offer the new Version 2 software, in answer to market demand and requests from our customers. Version 2 software is available for free download from the following website:

http://www.yamahaproaudio.com/downloads/firm_soft/pm5d/pm5dv2_frm.html

You can also use the PM5D Version 2 software to connect to the DSP5D digital mixing system itself or along with the DCU5D digital cabling unit (both sold separately) and expand the number of input channels available.

For details on the functionality of the DSP5D and DCU5D, as well as sample applications, please see the following website:

http://www.yamahaproaudio.com/products/mixers/pm5d/index.html

The following technical documents are also available at the above sites:

Setup Guide for DSP5D and PM5DV2

PM5D Quick Start Guide

PM5D Short-Cut List

Cascade Setup Guide

New Functions on the PM5DV2

We have added the following functions.

New Festival Friendly Security Functions

Tailor-made for those situations when you share a console with other operators, we have added new data security functionality.

LOAD LOCK

Prevents users from using the LOAD function before the LOAD LOCK function is deselected with password.

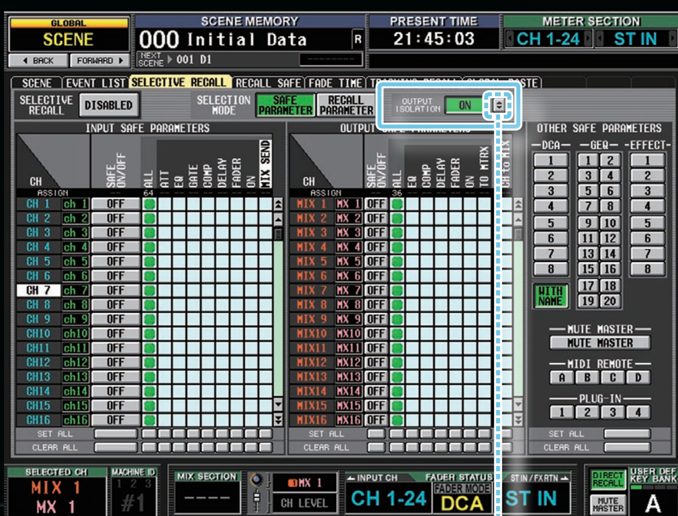
PARAMETER LOCK

In Version 1 of the PM5D software the parameters locked in the LOCK PARAMETER SELECT menu changed when data was loaded from an external source. Locked parameters do not change in Version 2. Furthermore, setting the RECALL LOCK button to ON exempts the selected parameters from being affected by any scene and library recalls.

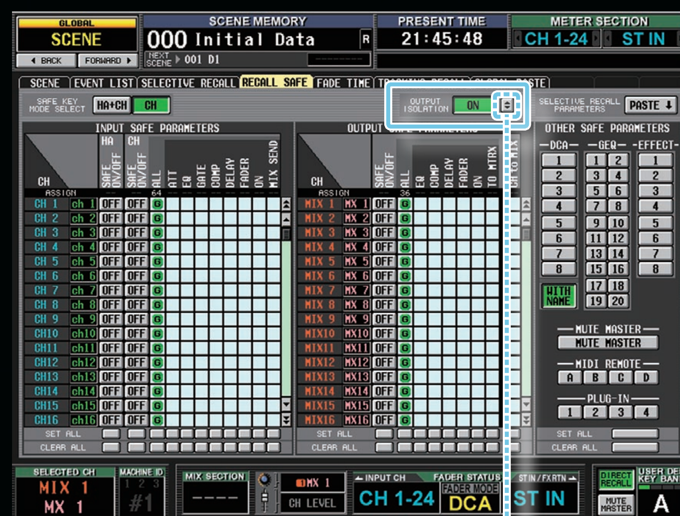
OUTPUT ISOLATION

When activated, the OUTPUT ISOLATION option exempts selected output channel settings, and the EFFECT and GEQ settings inserted to these channels, from being affected by recall. This function is different from RECALL SAFE in that when you load data into the console, no data is loaded for output channels selected in the current buffer memory, and your carefully designed output routing and processing will not be overloaded by visiting engineers.

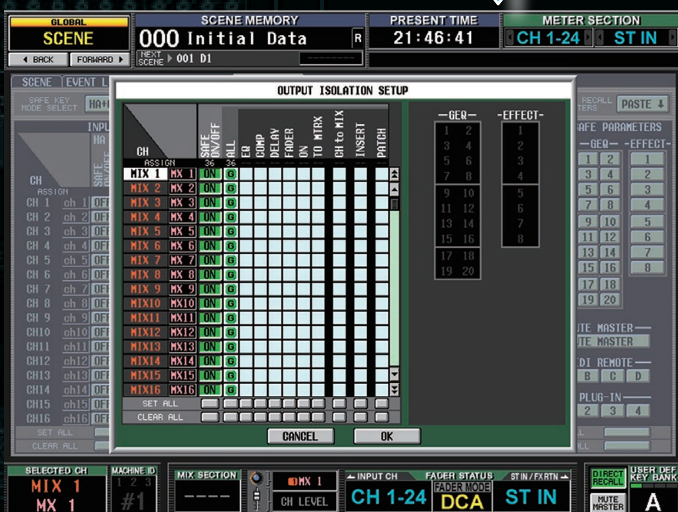
Note: For example, when you insert EFFECT1 to MIX1 which is isolated and recall Scene 000, output of the EFFECT1 shows changed patch information for ST return 1L and 1R because input patch is not isolated. But the insert patch to MIX1 is still active. The screen can show only one patch point, so the EFFECT screen says effect output is going to the ST return channel. (This means that the output of the effect unit is going to both of the ST return channel and insert in of the MIX1.)



SELECTIVE RECALL page



RECALL SAFE page

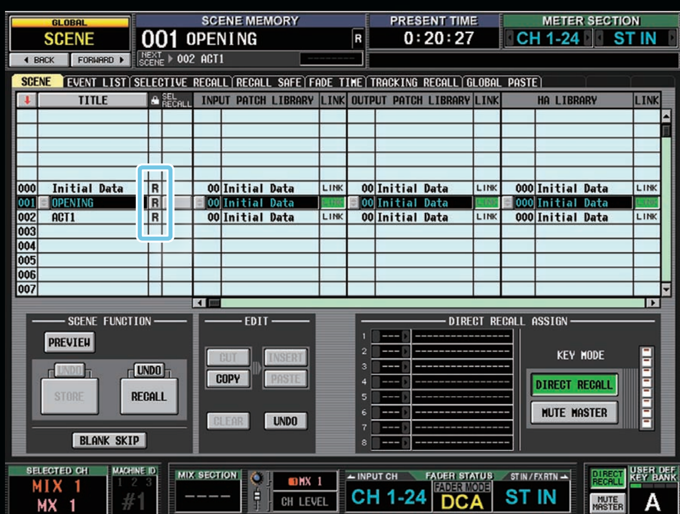


OUTPUT ISOLATION setting

Read-only scene

This function prevents scenes marked with R (read-only) from being overwritten by other scenes, or by files being loaded into the console.

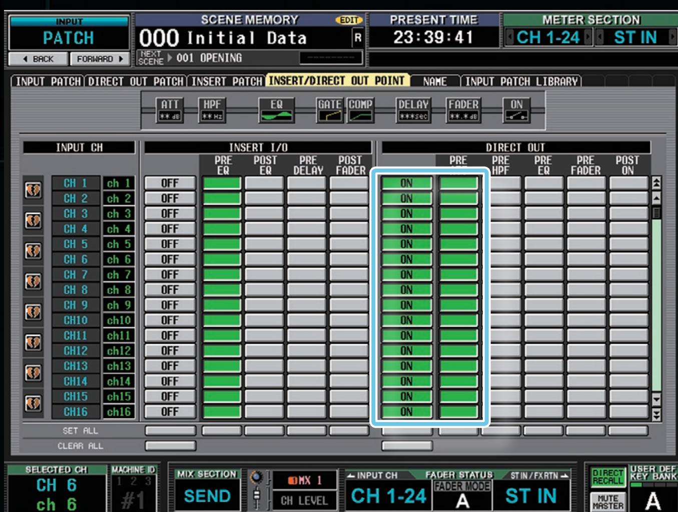
Note: Scene link libraries (INPUT / OUTPUT PATCH and HA) are overwritten. You can only apply this setting to consecutive scene numbers beginning at scene 000. These scenes are cleared when you initialize the console.



Improved recording and sound check functions

PRE ATT for direct out

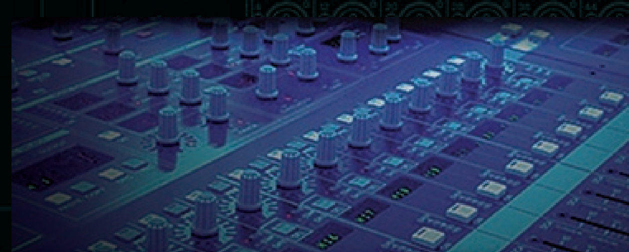
We added PRE ATT as a transmit location for direct outputs. This means that you can send signals directly from an AD converter to an external recorder or to another system, and separate the recording signal from all mixing operations.



VIRTUAL SOUNDCHECK

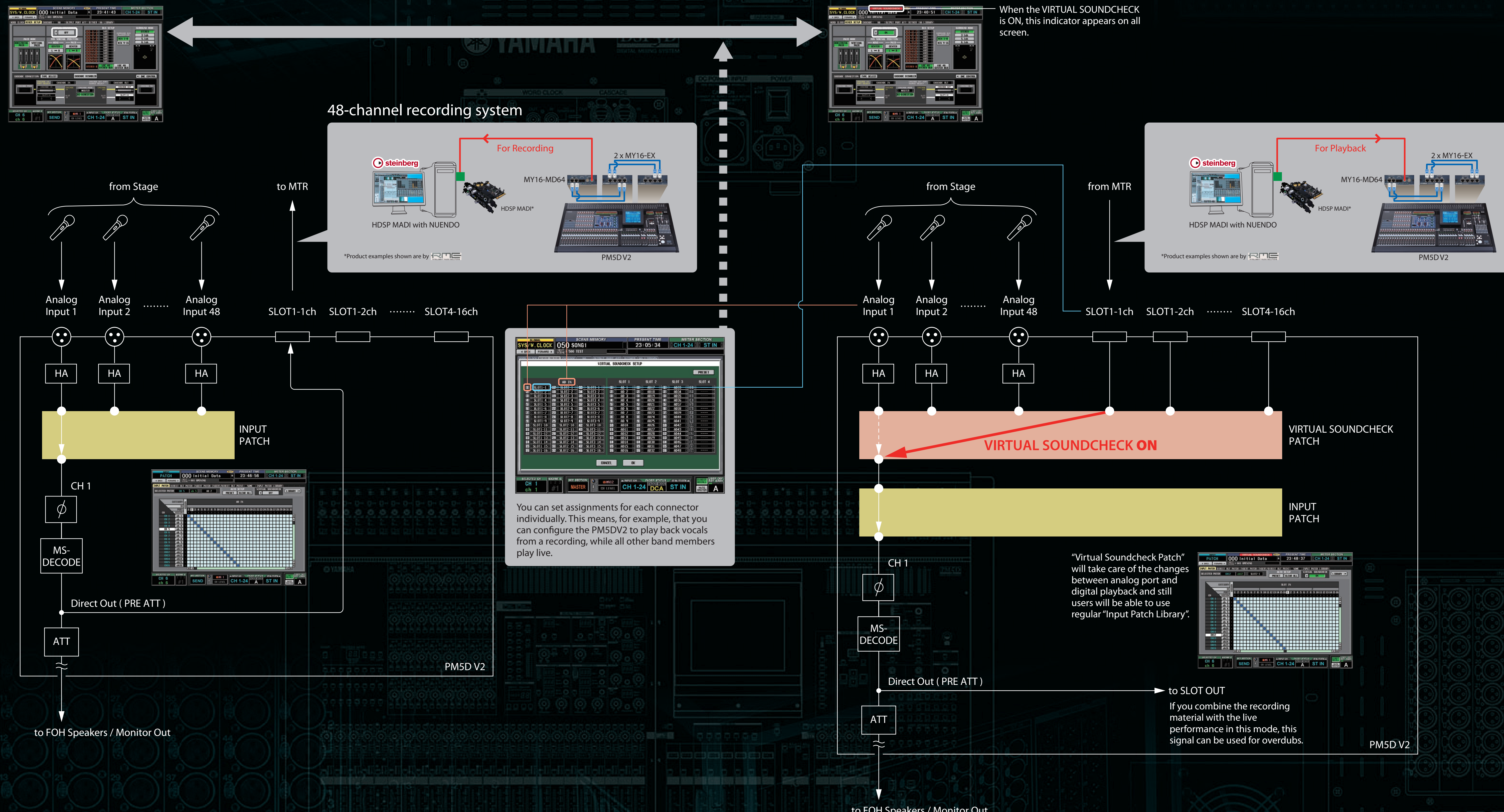
When performing a sound check, the new VIRTUAL SOUNDCHECK option lets you change input patches in the currently recalled scene temporarily, without permanently affecting the input patch configuration stored with that scene. For example, stage inputs and playback tracks from a prior recording using the PRE ATT direct out described above coming into MY slot connectors can be temporarily swapped at a sound check the following day, allowing you to make detailed adjustments to the EQ and effect settings without the performers even present. The use of multi-track recorded material, playback from Cubase, Nuendo, Protools or another recording package, as an aid to sound-checking has become popular. The VIRTUAL SOUNDCHECK feature makes this really easy. With a simple click you can instantly switch from your mic sources to your playback tracks or even a mixture of both; this proves really useful for rehearsals. Because VIRTUAL SOUNDCHECK setup is completely outside scene control, you can recall, edit and resave scenes in VIRTUAL SOUNDCHECK mode without worrying about storing patch changes. One tiny button on the screen is one powerful feature benefit to the sound engineer.

For details, see the diagram on next page.



VIRTUAL SOUNDCHECK **OFF** (for live performance / regular soundcheck)

VIRTUAL SOUNDCHECK **ON**



Muting sends with the DCA [MUTE] key from any send position

We have added a DCA MUTE TARGET option that lets you mute even those mix sends sent from a PRE FADER point.

POST ONLY: PRE FADER signals are not muted.

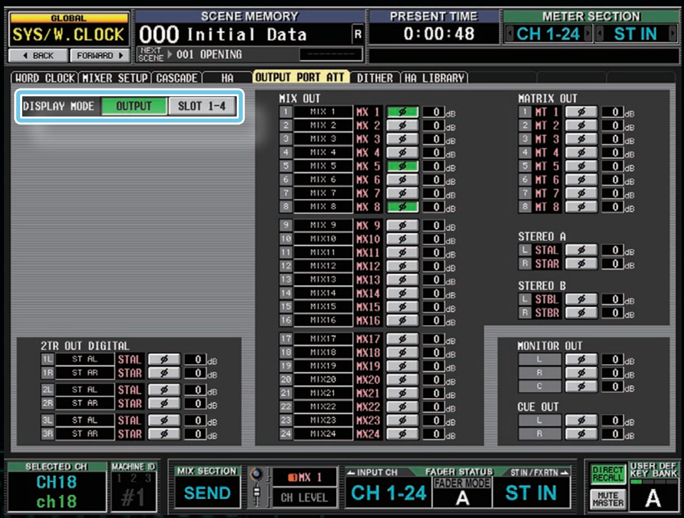
PRE & POST: As with the mute group function, PRE & POST mutes everything, regardless of whether the signal is PRE FADER or POST FADER.

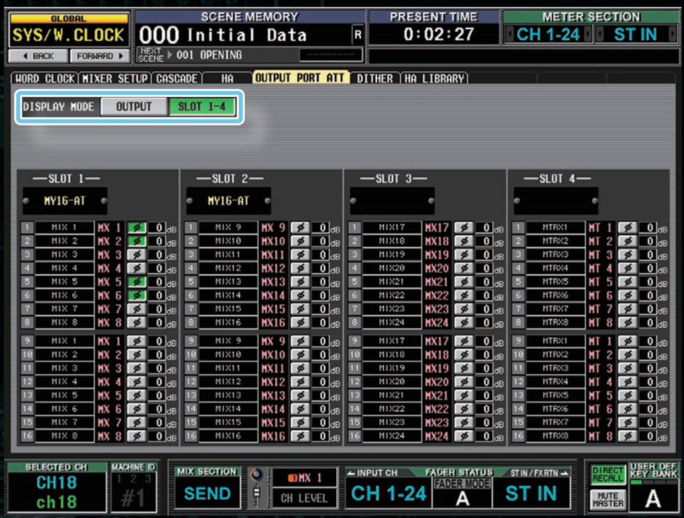
Note: Direct out will not be muted by this function.

Phase (polarity) switching option for output connectors

The PM5DV2 features a phase switching option for the following output connectors and output connectors on I/O cards inserted into a MY-SLOT, allowing you to change the phase of an output signal.

- MIX OUT 1-24
- MATRIX OUT 1-8
- STEREO A/B OUT L, R
- MONITOR OUT L, C, R
- CUE OUT L, R
- 2TR OUT DIGITAL 1-3 L, R
- MY-SLOT 1-4 output



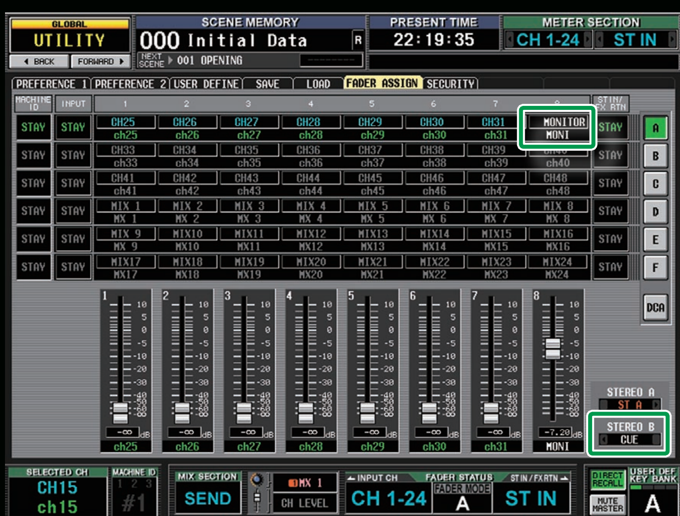


>>> For Monitor Mixing

Monitor and cue level control from STEREO and/or DCA strip sections

The PM5DV2 gives you monitor and/or cue level control and ON/OFF setting for them from a STEREO and/or DCA strip section. Now you can use the faders to adjust those monitor levels you had to alter with the volume knobs in the past, and set levels accurately for your monitor speakers in those situations where you need to make frequent adjustments.

Note: Connect cables to MONITOR OUT and/or CUE OUT connector on the back of the console. The control of the fader assigned to monitor/cue level and the original monitor/cue knob are in the signal line in serial. This means when one of them is at minimum, there is no output from the MONITOR OUT or the CUE OUT connector.



Panning and gain attenuation when flipping faders

In another example of increased usability, you can now use the multi-function encoders to control head amp gain and attenuation, and the panning of the signal sent to the paired MIX busses, even when the FADER [FLIP] key is set to ON. This is especially convenient when you are using in-ear monitors, as you can control send levels and panning for paired MIX busses.

Automatic GEQ selection using the channel [SEL] key on the GEQ PARAM screen

When the GEQ PARAM screen is displayed, the GEQ module inserted into that channel is automatically selected when you press the [SEL] key on the panel, and also when the [SEL] key is activated via any linked settings. This linking occurs in the following cases:

- When you press the [SEL] key on the panel (this function was included in firmware versions up to and including V1.21).
- When the [SEL] key is activated by the cueing operation with CUE/SEL LINK set to ON.
- When the channel in the SELECTED CHANNEL section is changed with the CH [INC]/CH [DEC] keys.
- When you actively link the channel on screen.

>>> Enhancement of the Processors

Effect library additions

DE-ESSER

DE-ESSER has been added.

Additional ADD-ON EFFECTS

Yamaha VCM (Virtual Circuitry Modeling) technology-based AE-011 (Compressor 276/276S, Compressor 260/260S, Equalizer 601) and AE021 (OPEN DECK) add-on effects are included as standard on the PM5DV2. VCM Technology models the circuits of a variety of standard electronic devices at an elemental level (resistors and capacitors, for example), making the most of their strengths while offering the advanced musicality of the digital domain. For further details, refer to page 24.

Compressor 276/276S: The Compressor 276/276S emulates the characteristics of analog compressors that have become standard recording studio equipment, and uses the same analog valve circuits as the original for its modeling. This effect offers a warm sound for vocals, and a fat bottom end for drums and bass. Compressor 276 is a mono split effect. You can control two monaural channels independently. (ex. Kick and Snare) Compressor 276S is a stereo effect. You can link its parameters for the L and R channels.

Compressor 260/260S: The Compressor 260/260S emulates the sound of the compressor / limiters that have been used for live sound reinforcement since the late seventies. The modeled circuit is of a classic analog VCA style compressor design. This effect gives you a simpler way to obtain a clearly defined sound. The Compressor 260/260S also gives you control over the attack and release parameters lacking in the original analog compressor / limiters. Compressor 260 is a mono split effect. You can control two monaural channels independently. You can also control some parameters simultaneously by making a stereo link. Compressor 260S is a stereo effect. You can link its parameters for the L and R channels.

Equalizer 601: The Equalizer 601 emulates the characteristics of a seventies-vintage analog equalizer. This effect reproduces the distortion unique to analog circuitry to give you a raw, edgy sound.

OPEN DECK: The OPEN DECK emulates the tape compression previously obtained by utilizing both a recording and a playback open-reel tape recorder. This effect lets you build combinations of parameters like deck type, tape quality and playback speed, enabling you to alter sound output quality to suit your needs.

Compressor 276

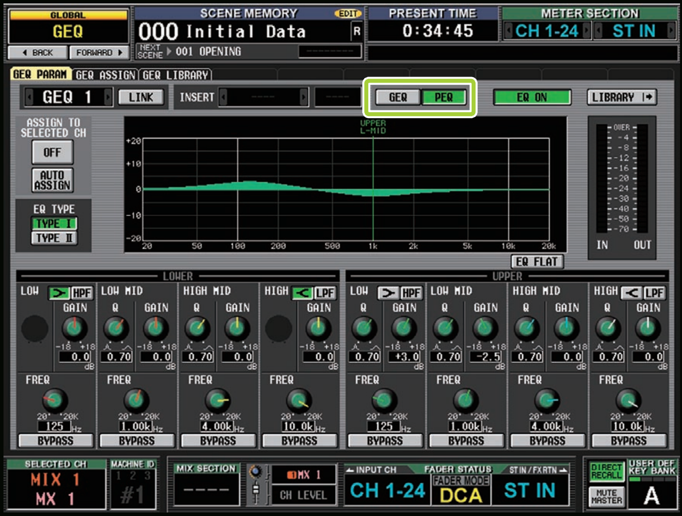
Compressor 260

Equalizer 601

OPEN DECK

GEQ/PEQ functions for internal effects, and PEQ functions for GEQ

We have added a DSP CONFIGURATION option to the PM5DV2, so that you can now use either 31-band graphic EQ or 8-band parametric EQ for internal effects 1 through 8. You can also allocate any of the 8 internal effect processors for GEQ, which means that you can increase the number of GEQs available to a maximum of 20 from the default 12. What's more, the PM5DV2 now lets you switch from 31-band graphic EQ to 8-band parametric EQ. And EQ control knobs in the SELECTED CHANNEL can be used for the PEQ when you access the GEQ PARAM screen if the ASSIGN TO SELECTED CH ON/OFF button is on. For example, for MATRIX OUT, STEREO OUT or MIX OUT channels with limited-band PEQ, you can increase the number of bands available by changing GEQ to PEQ, then inserting the EQ into the appropriate output channel, for more precise control.



Expanded Q factor range for parametric EQ

In addition to the 0.10 to 10.0 range offered in previous firmware versions, you can now select a Q factor of 11.0, 12.5, 14.0, and 16.0 for input channels, output channels, and parametric EQs that have been converted from GEQs.

Extended GATE threshold level for input channels

We have added the -55 dB to -72 dB range (1 dB steps), in addition to the previous values for GATE threshold levels for input channels.

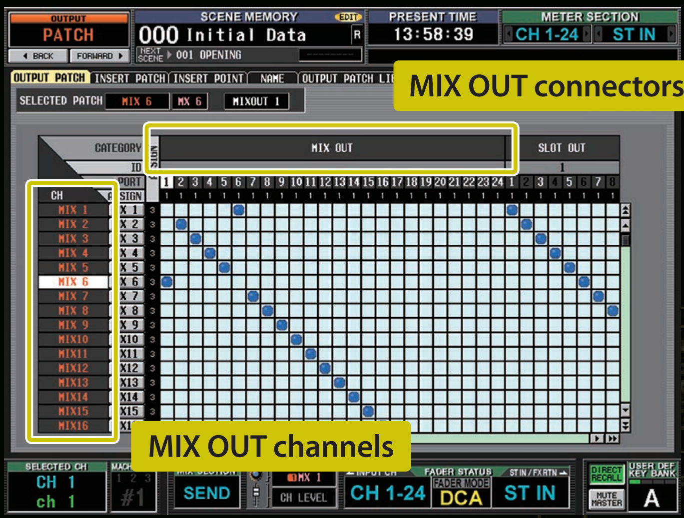
Other Enhancements for Better Operating Experience

Changeable output patching to MIX OUT

You can now change output patching to MIX OUT connectors 1 through 24. Patchable signals:

- STEREO A/B OUT
- MIX OUT 1-24
- MATRIX OUT 1-8
- INSERT OUT
- OSCILLATOR OUT (L ch)
- TALKBACK OUT
- MONITOR OUT L, R, C

1) Even when the cables are connected in a different configuration to the channel assignments in the current scene or in scene memory, you can change the patch state of the MIX OUT connectors to get the right connections without having to re-connect cables or replace one channel data to another.
2) You can send identical sound signals – MIX and MATRIX, for example – from multiple outputs.
3) You can send inserts for analog signals to external effect units without using MY-SLOT cards, and return the inserts via input channels or a MY analog card.



Channel move

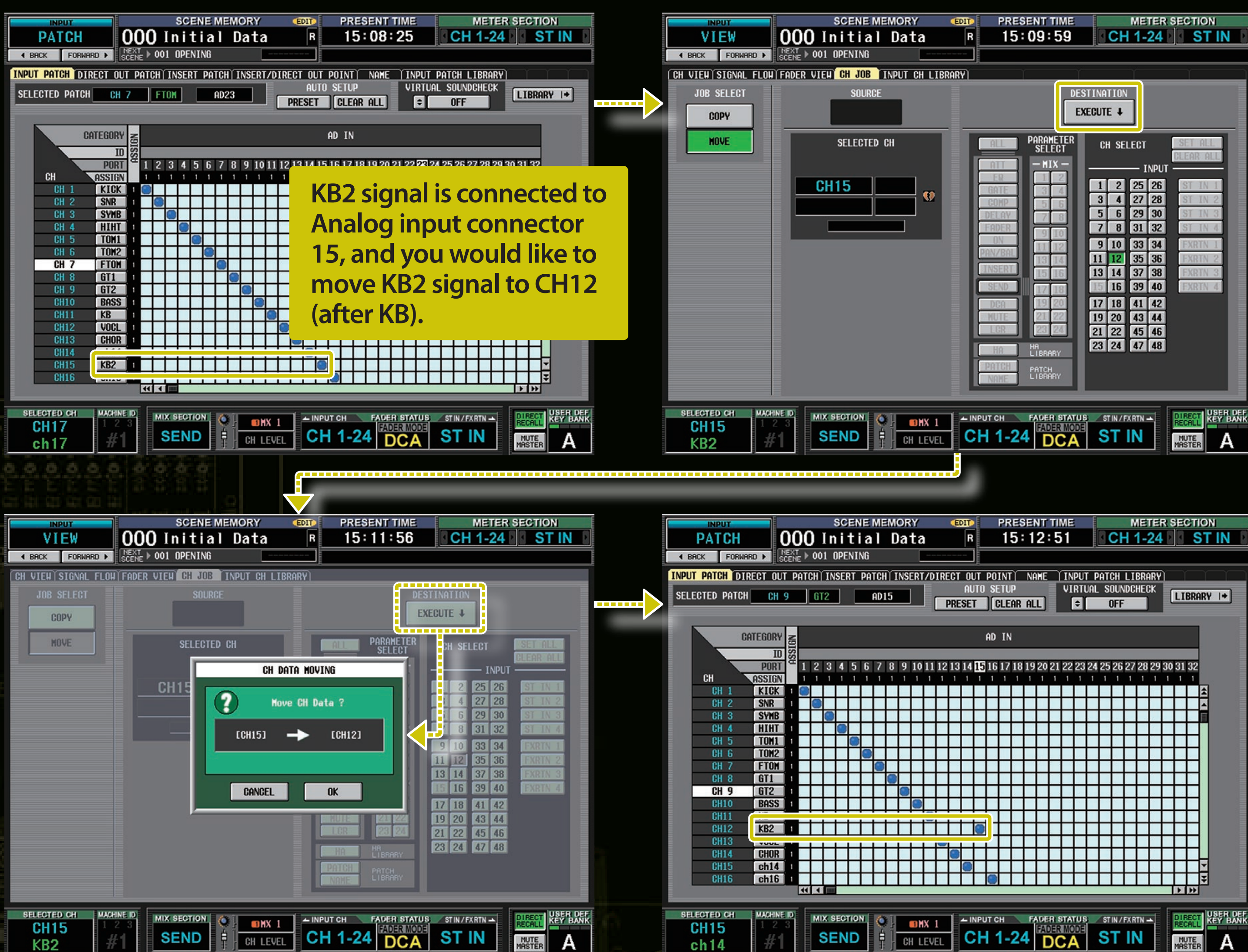
We have added a MOVE option to the input channels so that you can now move channels whenever you need to. Using the channel move function also moves the channels between the source channel and its selected destination as often as necessary, and Input Patch also changes automatically. This is great when you want to move a channel to another location on the surface in order to accommodate an additional microphone, or when you want to swap channels.

Note 1: A single channel can be moved only if the following two conditions are satisfied.

- There are no paired channels between the move-source and move-destination.
- You have not selected a parameter of which only one exists for every two adjacent odd-numbered/ even-numbered channels, such as DELAY GANG.

Note 2: Channels can be moved only within the following channel sections.

- INPUT channels 1–48
- ST IN channels 1–4
- FX RTN channels1–4



Additional USER DEFINED KEYS functions

The PM5DV2 brings you the following additions to the functions that can be assigned to USER DEFINED KEYS:

DSP5D CONTROL: This has two options. MACHINE SELECT option selects the PM5D or DSP5D (#2 or #3) as the machine that will be controlled by panel operations (Lit if the specified machine is selected). CH STRIP LAYER DIRECT option switches the INPUT channel strip or ST IN/FX RTN channel strip to a layer of the desired machine (Lit if the specified layer is selected).

SET NOMINAL LEVEL: Sets the channel fader levels as nominal when this key is held down and press the [SEL] key. And in FADER FLIP mode, MIX send level can be set to nominal.

SET DEFAULT VALUE: Returns knob and fader settings on the LCD screen to their default settings when the cursor is on the parameter and the ENTER key is pressed while the user defined key is held down.

ENCODER MODE KEY: Works same as the [PAN], [GAIN/ATT], [ALT LAYER], and MIX SEND SELECT [1]-[24] keys in the ENCODER MODE section, and the FADER [FLIP] key in the FADER FLIP section of the top panel.

PAGE CHANGE: NEXT TAB option (displays the next page of the selected function), and PREVIOUS TAB option (displays the previous page of the selected function tab) are added.

TAP TEMPO: ALL EFFECTS option is added. Usually all effect with delay has same tempo for a song. Now you can input tempo with one key for all effect units. In addition, the key flashes in time with the effect.

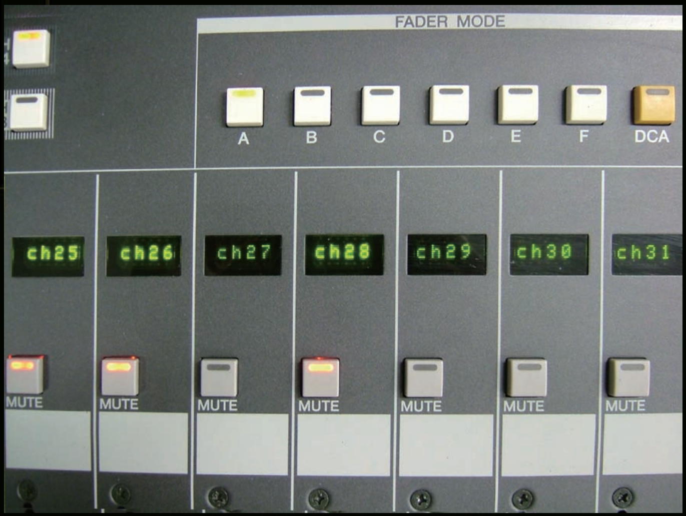
EFFECT linking using the channel [SEL] key on the EFFECT PARAM screen

When the EFFECT PARAM screen is displayed, the EFFECT module inserted into that channel is automatically linked and selected when you press the [SEL] key on the panel, and also when the [SEL] key is activated via any linked settings. This linking occurs in the following cases:

- When you press the [SEL] key on the panel.
- When the [SEL] key is activated by the cueing operation with CUE/SEL LINK set to ON.
- When the channel in the SELECTED CHANNEL section is changed with the CH [INC]/CH [DEC] keys.
- When you actively link the channel on screen.

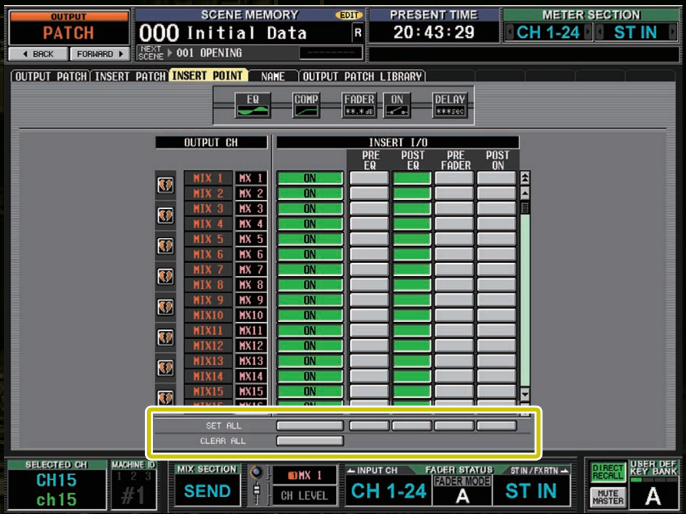
Channel ON/OFF using the DCA [MUTE] key in the FADER MODE section

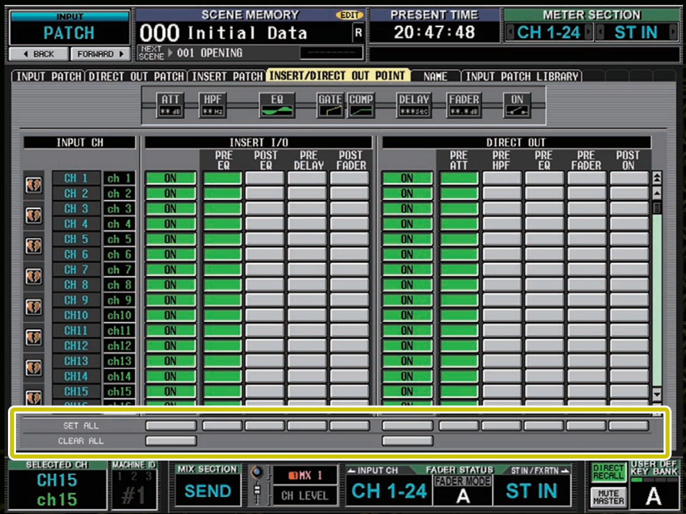
You can now use the DCA [MUTE] key to turn channels that have been selected in the FADER MODE section on and off. The switches follow the same logic: light ON is channel ON. If the DCA is selected, the DCA [MUTE] key acts as a DCA group muting.



SET ALL and CLEAR ALL functions for INSERT/DIRECT OUT POINT settings

The OUTPUT PATCH function INSERT POINT screen and the INPUT PATCH function INSERT/DIRECT OUT POINT screen feature new SET ALL and CLEAR ALL buttons that allow you to change points and turn all channels on or off at once.



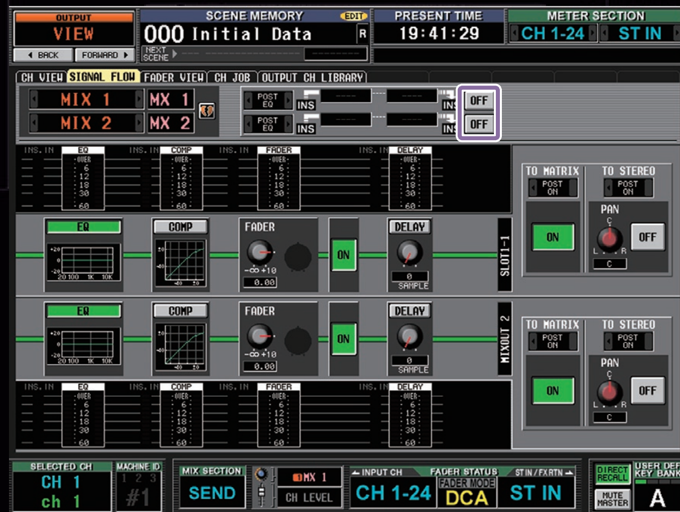


Fail-safe Options and Others

Automated ON/OFF for insert-in when using GEQ inserts

When you insert a GEQ, the PM5DV2 automatically switches on the appropriate insert-in, and switches it off again when you remove the GEQ. This means that you can use GEQ in real time, and remove it without muting the sound signal.

When GEQ is inserted into MIX 1



When GEQ is removed from MIX 1

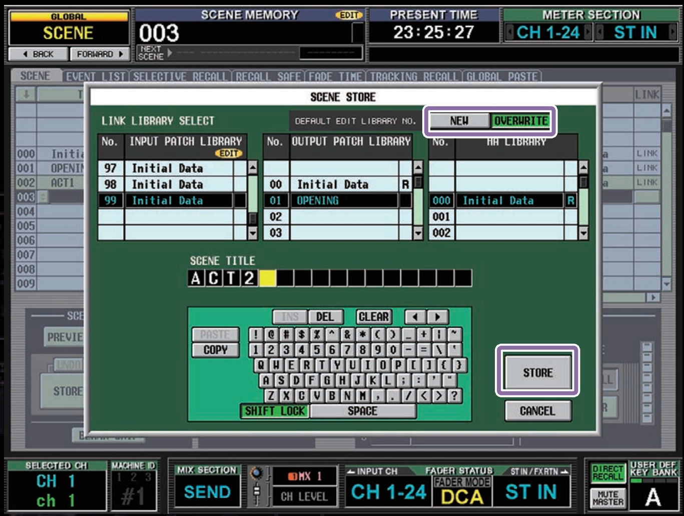
Attenuator locking for encoders

We have added an ATT OPERATION ON PANEL option to the PM5DV2 that allows you to prevent operators from using the panel encoders to operate the attenuator. This helps prevent those “I was just trying to adjust the head amp gain, but I changed the attenuator!” moments.

Overwrite lock functions when attempting to store a new scene with no empty libraries

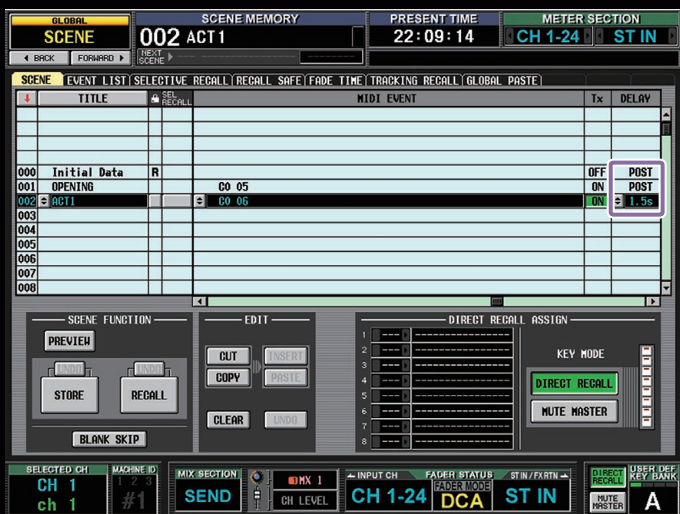
If there are no empty library numbers available when you select "NEW" to attempt to store a scene, the PM5DV2 will prevent you from saving the scene and overwriting existing libraries.

You cannot push STORE button.



Timing adjustment function for MIDI program change sends during scene recall

The timing of program changes and MIDI events sent during scene recalls can now be changed from the DELAY parameter settings screen. Furthermore, these parameters can be set for each scene. You can now change the settings on MIDI-compatible external devices, such as effect processors, synchronizing with scene recall in more precise timing.

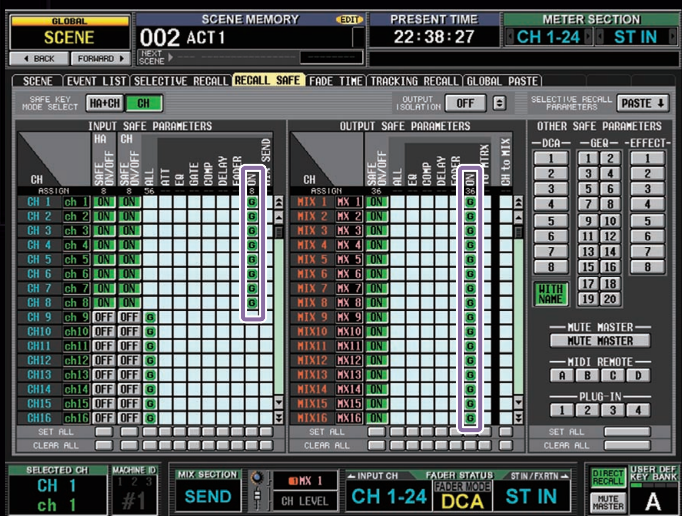


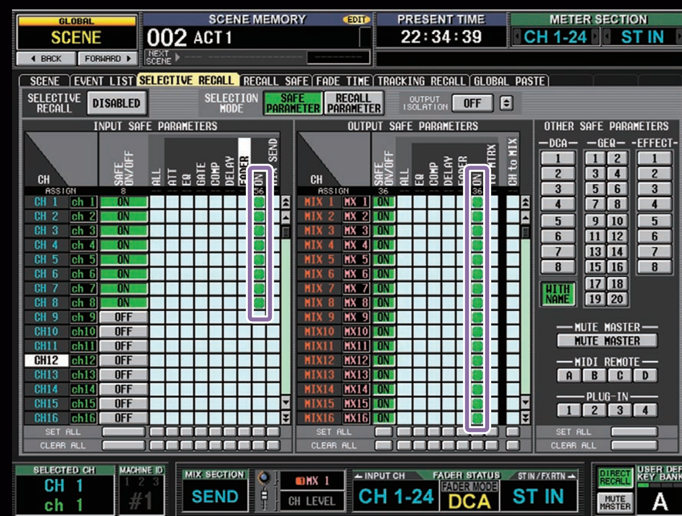
POST: Program changes and MIDI events are sent after scene recall processing is completed (POST works the same way as firmware versions prior to and including version 1.2).

INTERVAL: Program changes and MIDI events are sent after a fixed period of time once scene recall has started. You can preset this time from 0.0 seconds to 9.9 seconds, in 0.1 second increments.

ON/OFF parameters for the RECALL SAFE and SELECTIVE RECALL functions

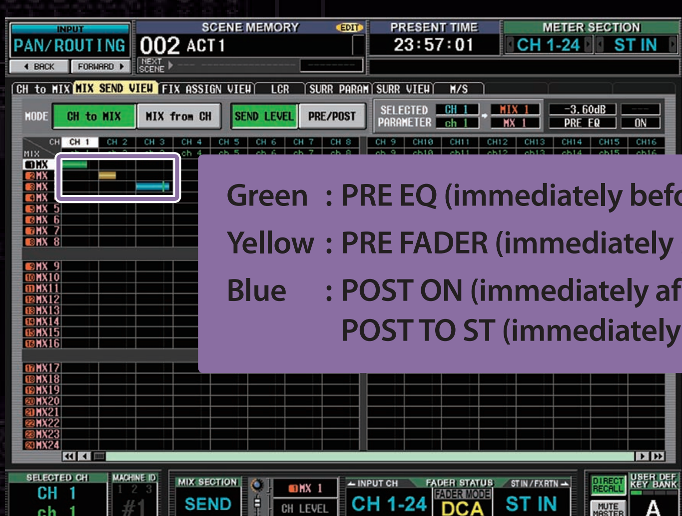
The RECALL SAFE and SELECTIVE RECALL functions now feature an ON parameter that lets you select whether the ON/OFF settings for each channel are affected by scene recall.





Color differentiation of PRE/POST in the MIX SEND VIEW screen

To make the parameters on the MIX SEND VIEW screen easier to see, the send positions (PRE/POST) for signals from the MIX bus are now displayed in different colors on the bar graph.



Green : PRE EQ (immediately before the EQ)
Yellow : PRE FADER (immediately before the fader)
Blue : POST ON (immediately after the [ON] key) or POST TO ST (immediately after the [TO ST] key)

Virtual Circuitry Modeling
A Technology for Yamaha's Digital Mixing Consoles

Abstract – High quality audio processing technology is proposed. The technology is called "Virtual Circuitry Modeling" (VCM) and is based on the concept that component-level modeling of analogue circuitry can be applicable onto the audio effect area. Not only the modeling complexity and accuracy but also the sound quality are achieved at a level of no-compromise, and implemented on digital audio consoles of Yamaha like PM5D and DM/0 series.

By Toshifumi Kunimoto
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Introduction

Yamaha has developed state-of-the-art technologies for digital emulation of analogue studio gear and implemented them onto new updates for Yamaha's latest digital mixing consoles. This innovative technology, Virtual Circuitry Modeling technology (VCM) covers a big range of audio effects like compressors, equalizers, analog-tape-recording and guitar stomp boxes. Rather than simply attempting to approach the desired sound using conventional digital audio methods, VCM technology actually models the analogue circuitry down to individual resistors, capacitors and operational amplifiers.

This technology was originally developed at Yamaha when we developed the VL1 and VP1 – the world's first physical modeling synthesizers. VCM technology goes well beyond simply analyzing and modeling electronic components and emulating the sound of old equipment. It is capable of capturing subtleties that simple digital simulations cannot even approach, and in fact creating ideal examples of sought-after vintage gear. As digital recording/reproduction systems are becoming increasingly popular in the industry, the importance of analogue sounding signal processing is becoming more obvious to engineers in this industry.

Some theoretical aspects of these technologies, especially with regard to the analogue tape recorder emulation, are described in this paper to illustrate how these new technologies lead to authentic analogue sound characteristics.

Anatomy of the tape recorders

Figure 1 illustrates the recording/reproduction procedure of an analogue tape recording system. It is a very complicated procedure and there are many components here that should be simulated if you want to achieve good musical-sounding analogue tape recording emulation in the digital domain.

We analyzed this procedure in great detail. Some of the components such as the differentiator and the NAB frequency equalizer can be simulated rather easily by digital filters with adequate frequency responses. The problem seemed to be non-linearity and losses. Since this behavior is very complex, it is difficult to estimate these characteristics. After some theoretical considerations, we understood that all the components here can be realized by digital software on a DSP chip. As a result, figure 1 is not only an illustration of real analogue tape recording but also the model we employed on our "Open Deck" audio effect for Yamaha's digital mixing consoles.

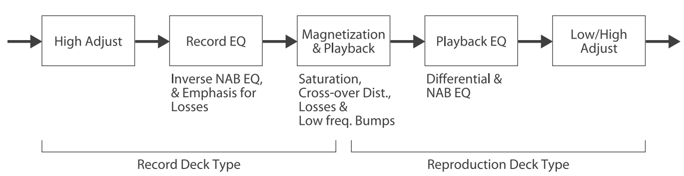

Figure 1.
The procedure of recording/reproduction in an analogue tape recording system

Measurement of real analogue tape recorders

We measured the highest-quality analogue tape recorders. As our modeling process employs component-level technology, the recording amplifier, the recording head, the tape, the reproduction head and the reproduction NAB EQ can each be realized individually in DSP. Consequently, our consultant (an expert analogue recording engineer) suggested that it could be very interesting to allow elements of recording and playback characteristics from different analogue decks to be combined. For example, when working with real analogue tape recorders, engineers may use an Ampex deck for mixdown, then use a Studer machine for mastering. This sort of flexibility should be provided in a digital emulation of analogue tape recording.

So we booked several professional recording studios, particularly focusing on those studios which had very well-maintained analogue tape recorders. We also hired engineers who had extensive experience with regard to the maintenance and alignment of analogue decks. It was necessary to spend several days to measure every aspect of four different types of analogue tape decks that were of particular interest to us.

Our measurements included not only normal bias conditions, but also over-bias and less-bias conditions, different tape types such as BASF 900 and Quantegy 456, 15 ips (inches per second) and 30 ips, different settings for Lo/Hi adjustments and levels, etc, etc. We also measured the responses of different combinations of the decks. We recorded these measurement data onto digital audio files and brought them back to our own laboratory to investigate them in detail from various aspects.

Figure 2 shows the recording/reproduction frequency magnitude response of four types of decks in a very normal situation (normal bias, BASF900, 500nWeb/m).

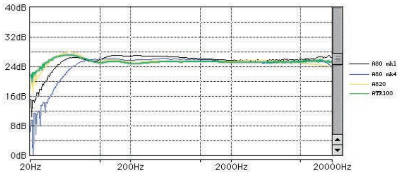

Figure 2.
Frequency responses of 4 types of measured decks

Analysis and application to the "Open Deck"

The procedure of magnetization is understandable as a sort of transfer response having particular magnitude frequency responses and very strange non-linearities such as saturation and distortion at zero-crossing points.

The frequency magnitude responses caused by NAB and de-NAB EQ are realized with some other compensation (pre-emphasis) filter responses on the modeling. The losses in the higher frequency area are very important to achieve the exact sound of the analogue tape recorders that are compensated for by the pre-emphasis filter. Also the frequency responses of the playback head (called "Contour effect") are depicted easily from measurement data and applied onto the model. The responses have unique, individual characteristics that vary on each of the decks. These individualities are mainly caused by the physical layout of the reproduction heads. They are characterized by bumps in the lower frequency area so that changes in the reproduction-side decks cause the sound of kick drums, snares and basses to be influenced and modified significantly.

The speed of the tapes has a very simple influence onto the magnitude frequency responses of the record and reproduction processes. Frequency responses are shifted up by one octave at 30 ips compared to 15 ips.

The saturation characteristics were extracted as parameters of our modeling in a straightforward manner. The behavior of the saturation effects on the individual analogue tape decks has been carefully reconstructed. With regard to this aspect, it was interesting to observe the considerable differences in linearity between various tape decks. This aspect of the tape deck response characterizes the sound of transient audio signals like cymbals. Sound engineers recognize this sound as dynamic compression from analogue tape decks that have strong saturating non-linearity. Consequently, this saturation effect of real decks is often referred to as "tape compression".

Magnetization on the surface of the tapes has zero-crossing type non-linearity. The importance of this phenomenon can be seen when you apply less and over-bias on each of the decks. In particular, a less bias situation causes big zero-crossing distortion and results in harsh and husky sounds on vocals. It is also reconstructed on our modeling. The difference between old and new types of tape is reconstructed as variations of the linearity on our modeling.

Many properties of each analogue tape deck's individual components are depicted. Once they are exactly reconstructed by the parameters of the models, they are very easily maintained on the digital software. So different combinations of different tape decks are very easily obtained due to component-level structure of our modeling. We reorganized these parameters and models as Swiss 70, Swiss 75, Swiss 82 and American 75. They are a nice set of the choices of good parameters that came from the decks we measured. Figure 3 shows the frequency responses of the final set we realized.

Graphical user interface panel of the "Open Deck" is shown on Figure 4. The "Open Deck" can realize not only the frequency response of the tape recording process but also the non-linearity and the musical sounds it has.

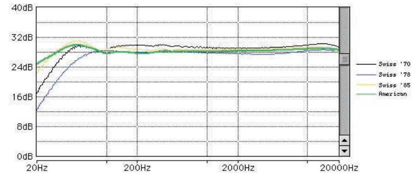

Figure 3.
Frequency responses of 4 parameter sets in the "Open Deck"

Figure 4.
Graphical user interface panel of the "Open Deck"

Other VCM family...the Compressor 276

In addition to the "Open Deck", the "Compressor 260", the "Compressor 276" and the "Equalizer 601" also employ VCM technology to emulate analogue out-board processors. We analyzed many vintage outboard effects, compressors and equalizers and we believe we have achieved some ideal analogue processors in the digital domain.

In particular, we feel that "Compressor 276" combines the strongest points of several different analogue compressors rather than simply attempting to emulate one specific processor. One of our evaluators described it as sounding similar to the UREI 1176 while another said it sounds like a Neve. Yet another said its timbre resembles the saturating characteristics of some tube-compressors. We can say they all are correct, because our intention was to combine many good analogue compression characteristics into one digital processor, rather than to strictly emulate one single analogue compressor.

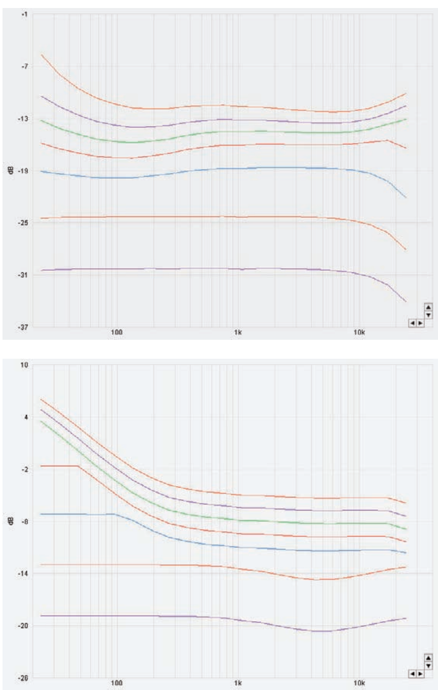

Figure 5.
Frequency-magnitude responses for various levels of stepped-frequency sinusoidal input to the Urei LA-2A (above) and the "Compressor 276" (below).

Figure 5 (above) exhibits the frequency responses of the LA2A using various levels of stepped frequency sinusoidal for input waveforms. Figure 5 (below) shows VCM/"Compressor 276". We defined a complicated dynamic frequency magnitude response of "Compressor 276" referencing many of the analogue compressors to have a sound character many engineers prefer and think it is ideal.

Graphical user interface panel of the "Compressor 276" is shown on Figure 6.

Figure 6.
Graphical user interface panel of the "Compressor 276"

And the Equalizer 601

As we were studying many of the analogue and vintage equalizers, we noticed many of them employ the parallel-formed circuitry as opposed to the cascaded-formed in digital equalizers. This parallel-structure of analogue gears basically defines the sound quality and the ease-of-use of the analogue equalizers.

Figure 7 (above) shows the frequency response of a certain analogue equalizer and our "Equalizer 601". Refer to the frequency response of the conventional digital equalizers on figure 7 (below). On 7 (above) you will notice the synthesized frequency response follows peaks of the elemental bands. On 7 (below), it does not. On analogue equalizers and the VCM counterpart, frequency-magnitude response of the equalizer can follow the engineers' image of the sounds easily, because synthesis of the response simply traces the levels of the knobs or faders.

The parallel-structure can have a problem when we implement it in the digital domain. (It has been referred to as the "delay-free-loop" problem in the digital signal processing field from a long time ago. It's a classic mathematic problem, so to speak.) VCM technology can solve this problem easily.

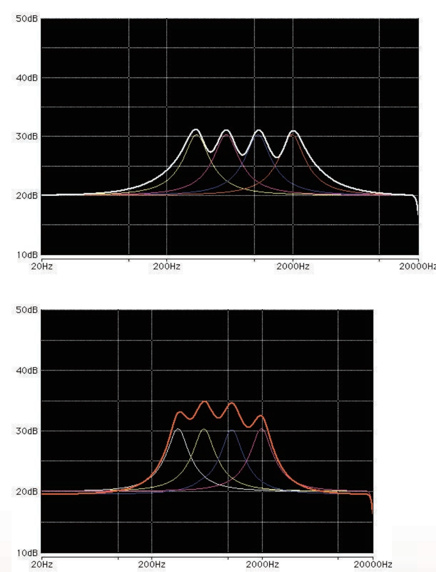

Figure 7.
Frequency-magnitude responses of analog equalizers/"Equalizer 601" (above) and conventional digital equalizers (below)

Graphical user interface panel of the "Equalizer 601" is shown on Figure 8. The "Equalizer 601" is also the result of the integration of many vintage equalizers like you see on "Compressor 276".

Figure 8.
Graphical user interface panel of the "Equalizer 601"

New comer, the phasers

Since scientists live in a kind of "ideal world", they have a tendency to think that simple is best. With regard to modeling of stompers/phasers, digital-acoustic engineers have been thinking they are simple comb filters. Figure 9 shows simple and ideal comb filter response of conventional digital phasers that contains 10 stages of all-pass filters. And it is true with regard to cheaper analogue products, they have the same kind of simple frequency responses. But actually, responses of analogue vintage stomper/phasers that sounds good in a musical sense, have very complicated frequency-magnitude responses.

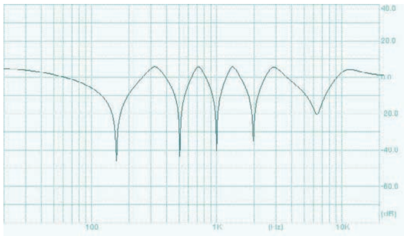

Figure 9.
Frequency-magnitude response of conventional digital phasers of 10-stage APF

Figure 10 shows one example of a vintage phaser, the "MXR Phase 100". Very precise reconstruction of frequency-magnitude responses of vintage stomper/phasers are achieved using VCM technology and taking care about just one capacitor usually omitted. Reproduction of each valleys and peaks in detail is necessary to emulate the musical quality of those processors.

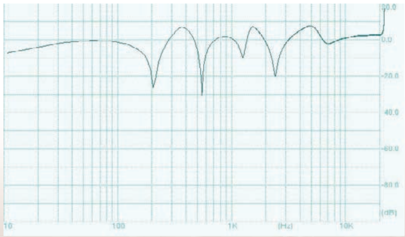

Figure 10.
Frequency-magnitude response of a vintage analog phaser, "MXR Phase 100"

Also some analogue gear is used to employ special devices like opto-resistance. It also should be emulated in physical sense. The dynamic behavior of an opto-resistance device is very strange. Responses for faster modulation and slower modulation are completely different. Not only that, but also the behaviors of changing slower to faster or faster to slower are so complicated and they result in very smooth timbre changes when tweaking the speed knobs of the phasers. VCM is the only technology that emulates such complexities.

Figure 11.
Graphical user interface panel of the "Max 100"

Conclusion

Many evaluation sessions were held in Tokyo, New York and London. More than a dozen very experienced engineers, such Elliot Scheiner, participated in these evaluations. These engineers made valuable and insightful comments about the proto-type Add-on effects being developed for our digital consoles. We prepared many VCM technologies such as compressors, EQ, phasers and tape-emulation algorithms for these evaluations. They were very helpful when we tweaked our algorithms.

It is interesting to point out that many recent modern recordings utilize analogue gear, especially analogue tape recorders for multi-track recording and mastering to provide the "tape compression" effect that adds the punch and warmth of analogue sound. A good example of this is the recent Steely Dan release "Everything Must Go" on which Elliot Scheiner employed a combination of analogue and digital multi-track recording in order to benefit from the best aspects of each technology.

VCM offers great opportunities for advancing digital audio signal processing technology. This type of modeling can be applied to the emulation of many other types of audio effects and processors including tube gear, guitar stomp boxes, analogue console channel strips, and many others. Very significantly, by modeling at the component level, VCM can allow the specifically appealing characteristics of different vintage equipment to be combined to create completely new digital processors.

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