ADD-ON EFFECTS

V2 Console Owners—Upgrade to VCM Class with Add-on Effects!

Your DM2000V2, DM1000V2, 02R96V2, or 01V96V2 digital mixing console can be upgraded to VCM series capability by simply adding the appropriate optional Add-on Effect packages. Owners of the original pre-V2 consoles will need to upgrade to V2 before installing the Add-on Effects.

CHANNEL STRIP PACKAGE (AE-011)

Virtual Circuitry Modeling · Developed by K's Lab

This Package includes 5 models that employ VCM (Virtual Circuitry Modeling) technology to recreate the sound and characteristics of several classic compression and EQ units from the 70's.

- Includes five models that employ VCM technology to recreate the sound and characteristics of classic compression and EQ units from the 70's.
- Fine-tuned by leading engineers, and featuring carefully selected parameters in a simple interface.

MASTER STRIP PACKAGE (AE-021)

Virtual Circuitry Modeling · Developed by K's Lab

The Master Strip Package Open Deck employs Virtual Circuitry Modeling technology to recreate both the analog circuitry and tape characteristics that shaped the sound of open-reel tape recorders.

- Employs VCM technology to recreate both the analog circuitry and tape characteristics that shaped the sound of open-reel tape recorders.
- The Open Deck provides models of four machine types: Swiss '70, Swiss '78, Swiss '85, and American '70. You can even combine different record and playback decks for a wider range of variation.
- You also have a 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.

REVERB PACKAGE (AE-031)

REV-X

The REV-X programs feature the richest reverberation and smoothest decay available, based on years of dedicated research and development.

- Reverb ADD-ON EFFECTS employing the latest REV-X algorithms first introduced in Yamaha's SPX2000 Professional Multi Effect Processor.
- The REV-X programs feature the richest reverberation and smoothest decay available, based on years of dedicated research and development.
- Hall, Room, and Plate programs are provided.
- The Hall and Room programs have a very open sound, while Plate delivers a brighter tonality that is ideal for vocals.

SURROUND POST PACKAGE (AE-041)

iSSP

The Surround Post Package uses Yamaha's Interactive Spatial Sound Processing technology that takes full advantage of the 96-kHz audio DSP power of the Yamaha digital consoles.
The AE-041 includes three plug-ins. Room ER, Auto Doppler and Field Rotation. These unique Plug-ins not only can vastly simplify the complex operation in Post-Production requirements, but also can be used creatively in the musical context.

VINTAGE STOMP PACKAGE (AE-051)

Virtual Circuitry Modeling · Developed by K's Lab

In this package Virtual Circuitry Modeling technology delivers faithful models of classic much-in-demand stomp boxes from the 70's that helped shape the sound of music history.
The AE-051 package includes three phaser models: the MAX100, Vintage Phaser, and Dual Phase. Although the vintage equipments are hard to come by, they are in considerable demand for both live performance and studio production.
All models feature graphical user interfaces that reflect the image of the times.

- ADD-ON EFFECTS can be only used with the PM5D/DM2000/DM1000/02R96/01V96.
- To use ADD-ON EFFECTS with the DM2000 Version 2, DM1000 Version 2, 02R96 Version 2 or 01V96 Version 2, please note the following.

1) The DM2000, DM1000, 02R96 or 01V96 must be Version 2 or higher.
2) Your personal computer must have a USB port and internet connection capability.
3) You must get web approval from Yamaha using the access key issued by the Yamaha approval server.
* When applying for web approval, use the CD-ROM and approval code within each ADD-ON EFFECTS package.

For details please contact:

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YAMAHA

DM/0 VCM Series

DM 2000 VCM
DIGITAL PRODUCTION CONSOLE

DM 1000 VCM
DIGITAL PRODUCTION CONSOLE

02R96 VCM
DIGITAL MIXING CONSOLE

01V96 VCM
DIGITAL MIXING CONSOLE

The World's Most Popular Digital Consoles Gain Even More Production Power and Versatility

Yamaha digital mixing consoles are established standards in the professional audio field. They have become the first choice of discerning audio professionals worldwide because of their unsurpassed audio quality, superior versatility, outstanding reliability, and advanced monitoring capabilities. The new VCM models – DM1000VCM, 02R96VCM, and 01V96VCM – now join the current DM2000VCM in providing a new dimension of production power with the addition of a formidable array of state-of-the-art processing technologies. Depending on the console you choose you'll have extraordinary recreations of classic compression and EQ units from the 70's, simulations of legendary analog open-reel tape decks, a complete suite of unmatched REV-X reverb effects, a surround postproduction package that is second to none, and even a selection of vintage stomp boxes to spice up your mixes right at your fingertips. There's no longer any need to patch in external processors when special processing requirements arise. The technology that has made all of this possible is some of the most advanced in the world, and the sound is simply superb. In short, some of the world's most popular digital production consoles just got a lot more powerful. The plug-ins pre-installed in each of the VCM series Plug-ins listed as "Optional" can be purchased as ADD-ON EFFECTS.

VCM Technology

VCM technology is responsible for the classic compressor, EQ, analog tape deck, and stomp-box effect simulations in the VCM series consoles. VCM (Virtual Circuitry Modeling) technology actually models the characteristics of analog circuitry – right down to the last resistor and capacitor. VCM technology goes well beyond simply analyzing and modeling electronic components and emulating the sound of old equipment. It's capable of capturing subtleties that simple digital simulations cannot even approach, recreating ideal examples of sought-after vintage gear.

iSSP Technology

iSSP technology is the key to the incredible performance of the surround post-production effects. iSSP stands for "Interactive Spatial Sound Processing," and is Yamaha's original new spatial sound effect system. Designed through extensive research and exhaustive testing, this technology offers unparalleled reality, operability and originality for surround processing applications. It delivers unprecedented sound-field positioning precision and versatility, as well as realistic sound source movement effects with simple operation that allows simulations of an almost unlimited variety of spatial environments.

REV-X

"REV-X" is the advanced algorithm behind Yamaha's newest generation of reverb and ambience programs, offering unprecedented reverberation depth and realism with smooth decay. REV-X technology takes full advantage of the 24bit/96kHz processing capability of the VCM series consoles for reverb and ambience effects that have the reassuring warmth and reality of natural acoustic environments.

	Channel Strip	Master Strip	REV-X Reverb	Surround Post	Vintage Stomp
DM 2000 VCM	Installed	Installed	Installed	Installed	Installed
DM 1000 VCM	Installed	Installed	Installed	Installed	Installed
02R96 VCM	Installed	Installed	Installed	Installed	Installed
01V96 VCM	Installed	Optional	Installed	N/A	Optional

Extraordinary New Plug-ins

The VCM series digital consoles offer unprecedented effect performance with a selection of new effect programs that employ Yamaha's revolutionary VCM (Virtual Circuitry Modeling) and ISSP (Interactive Spatial Sound Processing) technologies for unprecedented effect quality and control.

CHANNEL STRIP

• Pre-installed: DM2000VCM, DM1000VCM, 02R96VCM, 01V96VCM

The Channel Strip plug-in includes 5 models that employ VCM (Virtual Circuitry Modeling) technology to recreate the sound and characteristics of several classic compression and EQ units from the 70's. Not only do these models faithfully capture the unique saturation of analog circuitry – in part thanks to precise modeling of the original FET gain reduction, tube/transformer buffer amplifier, VCA (Voltage Controlled Amplifier) and RMS detector circuits – but they have also been fine-tuned by leading engineers and feature carefully selected parameters in a simple interface that makes it easier than ever to create the ideal sound.

Compressor 276 (mono), Compressor 276S (stereo)

These models recreate the fast response, frequency characteristics, and tube-amp saturation of the most in-demand analog compressors for studio use, delivering classic style compression with all the punch and fatness you'd expect from a fine piece of studio grade analog gear. Not limited to processing drums and bass, these compressors are also an excellent choice for vocals and master stereo mix compression. The 276 is a dual mono unit, while the 276S operates in stereo.

Compressor 276S (stereo)

Compressor 260 (mono), Compressor 260S (stereo)

Featuring faithful modeling of the solid-state voltage-controlled amplifier and RMS detection circuitry of the late 70's, these plug-ins bring back the sound of classic comp/limiters used primarily for live sound reinforcement applications. They offer three selectable compression knee types – hard, medium, and soft – and although variable attack and release are provided, presets recreate the fixed settings of the vintage gear. Top-level sound-reinforcement engineers have carefully tweaked the parameters for optimum response in live situations. The 260 is a dual mono unit, while the 260S operates in stereo.

Compressor 260 (mono)

Equalizer 601

The 601 equalizer offers two equalizer types: Clean and Drive. The Drive type models the distortion characteristics of 70's analog EQ circuitry, delivering musical-sounding drive and saturation. The 601 is a stereo six-band parametric equalizer with LO and HI shelving filters and four MID peaking filters, and it accurately reproduces both the boost and cut frequency response and band interaction of vintage analog gear. And you get EQ capability over a wide 16Hz – 40kHz range when operating at 88.2/96kHz. The 601 features a familiar knob style interface as well as graphical editing capability.

EQ 601

Snake Newton

FOH engineer, currently working for Duran Duran. Also worked for Craig David and Pet Shop Boys.

"When I am not on the road, much of my time is spent in front of a studio based around multiple Macs running Cubase SX. If these components are the heart and brain of the studio then the myriad of VST plug-ins that I use are it's life blood. Now Yamaha, with a remarkable set of new effects, have brought live mixing a big step closer to this flexibility. The ability to choose a 'vintage' type compressor or EQ with the click of a mouse is taken for granted in the studio. This is finally within reach thanks to the new range of plug-in type of effects from Yamaha!"

MASTER STRIP

• Pre-installed: DM2000VCM, DM1000VCM, 02R96VCM • Optional: 01V96VCM

The Master Strip plug-in Open Deck employs Virtual Circuitry Modeling technology to recreate both the analog circuitry and tape characteristics that shaped the sound of open-reel tape recorders. Because of their ability to smooth out peak levels and tidy up the response, many high-end recording studios still maintain open-reel recorders such as the Studer A80 mk I, A80 mk IV and A820, and the Ampex ATR100 and others from the 70's and 80's to be used to provide tape compression at the mastering stage. Different types of tape – new BASF, old Ampex, etc. – are also selected and used according to the unique sounds they produce. Open Deck provides models of four machine types: Swiss '70, Swiss '78, Swiss '85, and American '70. You can even combine different record and playback decks for a wider range of variation. You also have a 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. Now you can easily take advantage of top-end analog sound-shaping techniques in real time using Yamaha VCM series digital consoles.

America '70 + Swiss '78

America '70 + America '70

Swiss '70 + Swiss '70

Swiss '78 + Swiss '78

Swiss '85 + Swiss '85

Rick Pope

PM1D: 2001 tour with Jamiroquai – PM5D with Clear Channel, doing 'instant live' recording directly to CD, first started using the new effects then. – Now using PM5D (&DM1000) for Jamiroquai's tour.

"I use Open Deck all the time as a 'finalizer'. We don't have time to do proper finalizing with Clear Channels' live recordings, so the Open Deck gives it that finished result. Sounds like it's mastered off a half-inch. I've tried all the types, and they are very subtle differences. I tend to use Swiss 85. You can really notice the difference between new and old tape. The Old Tape setting really sounds as if its been through the heads several times."

Fumitoshi Nakamura, Office Invillage

Engineer to some of Japan's top musicians, Fumitoshi Nakamura is also a busy producer, arranger, and programmer. Having started to use Yamaha digital mixers with the 02R, he recognized great improvements in the head amps and monitor section on models such as the DM2000 and 02R96, but wondered if they would really meet his needs, stating that a good dynamics system would truly seal the deal. So, we had him put the ADD-ON EFFECTS series to the test.

"I've really been impressed with how the compressors (from Channel Strip) can be used instead of analog gear, but Open Deck is also great. It sounds just like a tape-based compressor should, and switching amplification on the recording and reproduction decks produces some very interesting effects. In particular, the sensation of tape saturation is unique to this effect and is implemented very well. I was actually in the middle of a mixdown when asked to try Open Desk, and I used it on the master to be submitted. The sound that this produced had a very different feel to what could be achieved by simply mixing within Pro Tools."

REVERB

• Pre-installed: DM2000VCM, DM1000VCM, 02R96VCM, 01V96VCM

These reverb plug-ins employ the latest "REV-X" algorithms first introduced in Yamaha's SPX2000 Digital Multi Effects Processor. The REV-X programs feature the richest reverberation and smoothest decay available, based on years of dedicated research and development. REV-X Hall, REV-X Room, and REV-X Plate programs are provided, with new parameters such as room size and decay envelopes that offer unprecedented definition and finer nuance control. The REV-X Hall and REV-X Room programs have a very open sound, while REV-X Plate delivers a brighter tonality that is ideal for vocals. All models deliver dense, warm reverb that does not interfere with the natural timbre of the source.

Steve Levine

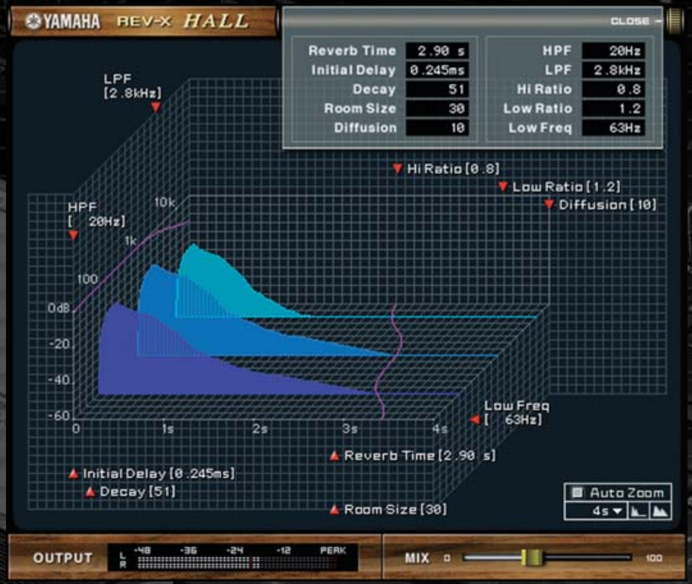
Recording and mixing engineer, worked for many artists including Culture Club, The Beach Boys, Honeyz, and Gary Moore.

"I am very impressed with the new REV-X reverbs. These reverbs sound so good, a match for any current hardware reverb unit – the REV-X Room simulation is the best "room sound" I have heard since the famous Quantec room simulator."

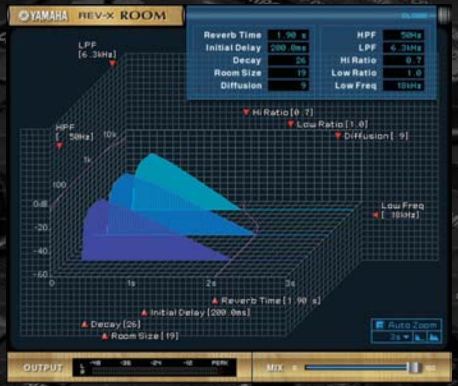
Satoshi Inoue,
ViViA (TV Asahi Productions)

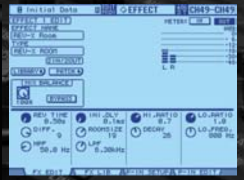
An engineer at TV Asahi Productions' Engineering Section, Satoshi Inoue is an expert in the field of on-location audio broadcasting, and in fact, was one of the first to successfully implement 5.1 surround-sound broadcasting of sports events, concerts, and the like. He tells us that he currently uses DM1000 and DM2000 consoles fitted with ADD-ON EFFECTS (with the same specifications as VCM) for broadcasting of the popular Japanese music program Music Station and other similar on-location productions.

"I didn't think twice about installing the DM1000 and DM2000 consoles, as I knew there was no better digital solution for surround sound in its truest sense. Their compact design is crucial when on-location, and we have never found ourselves stuck for space when loading the consoles onto trucks and setting up off-site. The first time we used these consoles on-location, we were working with the vocals of a very famous artist. Previously, I had always used analog compressors on live vocals, but on that occasion, I opted for Compressor 276. Even a relatively small amount of compression is sufficient to smooth out a vocal line, and in addition, it sits really well in the mix. I also use REV-X Plate together with this compression, and it sounds so sweet on vocals that we now always select that reverb instead of a Lexicon. REV-X reverbs have great attenuation, and this makes them ideal for a wide range of situations."

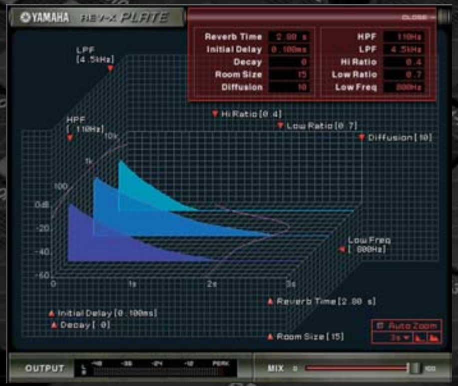


REV-X (Hall)





REV-X (Room)



REV-X (Plate)

SPX2000 PROFESSIONAL MULTI-EFFECT PROCESSOR

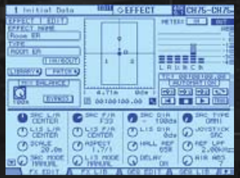
The SPX2000, while inheriting the standard interface and common programs from its predecessors, brings a new sound quality with the "REV-X" reverb algorithm and the 24 bit/96-kHz audio DSP.

SURROUND POST

• Pre-installed: DM2000VCM, DM1000VCM, 02R96VCM

These three plug-ins take full advantage of Yamaha's remarkable iSSP (Interactive Spatial Sound Processing) technology to deliver precisely-controllable spatial processing capabilities that are particularly suited to cinema or television sound post-production and mixing facilities. All plug-ins are applicable to a range of surround formats, providing unprecedented precision in matching visual motion with sound, and vast creative control for the creation of fantastic sonic environments. The Surround Post effects can be controlled directly from the console's joystick, where applicable.

Room-ER

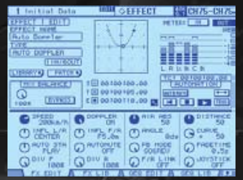


Room-ER is capable of simulating the acoustic properties of a room of about 30 meters in length, with accurate reproduction of the direct sound and early reflections as affected by distance from the source, source motion, speed of motion, and room surface characteristics. This plug-in ideal for placing a mono source in a precisely controllable surround environment.

• Application Ideas

As it's name implies, Room-ER is basically a room simulation that allows a monaural source to be positioned and moved within a simulated room. This can be useful, for example, to process a speaker moving away from the viewer in a movie or video scene. The sonic effects of the speaker moving away, turning to face the viewer, and moving back toward the viewer can be reproduced with remarkable precision. In addition to surround applications, this same effect can also be used to add a sense of depth to stereo music tracks, or to realistically simulate the effect of performers moving around the stage.

Auto Doppler



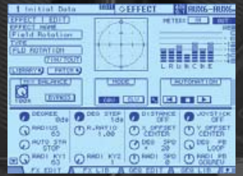
Perhaps the most common example of the Doppler effect is the change in pitch of an ambulance siren as it moves toward and then away from the listener. Auto Doppler effectively simulates this effect in a wide variety of scenarios. In addition to objects moving linearly past the listener, Auto Doppler can recreate the effect of objects moving toward and then away from the listener, for example, with precise speed and distance control. Timecode automation is also possible.

• Application Ideas

Auto Doppler is the ideal effect for simulating motion in a wide variety of situations. Basic point A to point B simulation can be used for the motion of cars crossing a scene, or aircraft taking off or landing at an airport. Point A through point B to point A' simulation is also available, and could be used, for example, in scene of a race car rounding a hairpin bend on a racecourse. Auto Doppler can simulate listener-to-source distances of up to about 1 kilometer, providing more than enough range for a wide variety of processing applications.

Field Rotation

The Field Rotation plug-in can be used to rotate or distort the sound field around the listener. The listener can be at the center of rotation, or the listener can be rotated or moved around a sound source. The axis of rotation, amount of movement, distance from the center of rotation, and speed of motion can be specified and controlled manually via a joystick, or automated as required.



• Application Ideas

Any scene that involves rotation is a potential application for this effect. Place the viewer on the coffee-cup ride or carousel at an amusement park, add realistic sound motion to a boomerang in flight, UFOs, propellers ...anything that spins or follows an elliptical path. This effect will undoubtedly find many uses in 3D video games, too.

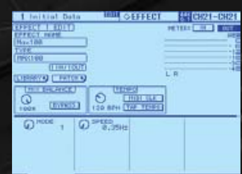
VINTAGE STOMP

• Pre-installed: DM2000VCM, DM1000VCM, 02R96VCM • Optional: 01V96VCM

This package includes a number of super-realistic recreations of vintage guitar stomp-box plug-ins that are highly valued for their rich, warm sound. VCM technology brings these outstanding effects back to life with greater controllability and flexibility than ever!

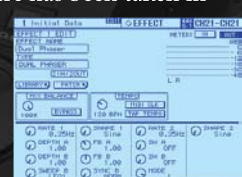
MAX100

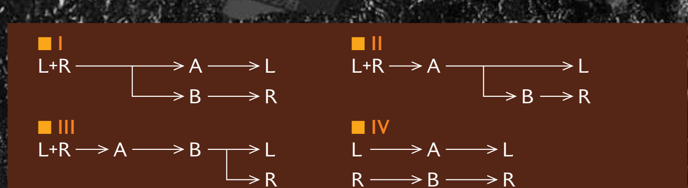
Born in the late '70's, this phaser is still available in reissue form. There are many who believe the original '70's models sounded better than the current models, so the K's Lab team painstakingly modeled the original circuit and components. Even the original light-sensitive CdS cell that was used for modulation has been modeled so the subtle change in character with modulation speed of the original is recreated in perfect detail.



DUAL PHASE

Like the MAX100, there are many guitarists who will go to any lengths to get an original version of this stomp box to enhance their sound. This is a faithful reproduction of the original with dual phaser circuits and dual LFOs that can be configured to deliver a dazzling array of effects. Special care has been taken in modeling the effect of the CdS cell in the phase-shifting circuit so that the exquisite balance at all modulation speeds that was a major part of the sound of the original has been retained.





VINTAGE PHASER

Rather than a simulation of a specific phaser, this model has been designed to deliver the best qualities of the most sought after classic phasers in one versatile plug-in. Different mode settings transform this effect into dramatically different phaser types. Stereo and mono versions are provided.

• Application Ideas

Vintage Stomp is a set of three plug-ins – MAX100, Dual Phase, and Vintage Phaser – used to model different types of phaser. With the birth of a wide range of musical styles in the latter half of the Sixties, many different musicians – notably guitarists and keyboard players – began to experiment with the aim of realizing their own unique sound or creating interesting effects. In both the live and recording environments, many different artists gained fame as a result of the innovative sounds used within their music. Around that time, therefore, effects such as the phaser and fuzz became critical elements of a modern sound.

One interesting example is the use of rotating speakers with organs in order to produce an undulating sound. The phaser is a simple means of recreating this type of effect, and to date, many different companies have produced phaser effects with a vast range of different aural characteristics. MAX100 and Dual Phase can accurately model some of the more distinctive of these phasers.

MAX100 reproduces a stompbox phaser with a crisp sound often heard on the guitar tracks of West Coast style music of the Seventies. In addition, this plug-in also works really well with held organ tones. Particularly suitable for use with tracks that have already been recorded, MAX100 sounds great on tonewheel-type organs, producing a clear, vivid sound.

Dual Phase produces dramatic sounds using a pair of phasers arranged in parallel. Ideal not only for organs, this type of device has also been used by guitarists and synth players alike to create stunning effects. In fact, many of these phaser's unique sounds cannot be recreated by other effects units. For example, when used with single-note cutting on a guitar, Dual Phase produces ethereal passages that actually sound like they came from a synthesizer instead. In addition, this plug-in also faithfully reproduces the complicated overtones generated by changing the speed of rotating speakers from slow to fast and vice-versa.

Last but by no means least, Vintage Phaser produces the sound that would be expected were Yamaha to produce an analog phaser today. Although loosely modeled on the stompbox-type guitar phaser produced by Yamaha in the Eighties as part of the PSE Series, this effect offers a wider range of controls and many more phaser stages. As such, Vintage Phase is all you need to fully experience the wonder of the classic phaser. And with the original footswitch also incorporated into the visual design, it may also rekindle fond memories of the good-old days of analog.

The Team and the Technology Behind the Sound

"Modeling is a means to an end, not the final goal." Mr. Toshifumi Kunimoto, the central figure of Yamaha's physical modeling technology team, has a fine track record when it comes to meeting challenging goals. The division known at Yamaha as "K's Lab" ("K" for "Kunimoto") was established in 1987 to develop new modeling technology that would become the next phase in synthesizer evolution after the FM and PCM tone generators that were the mainstay of the synthesizer world at the time. The result was the world's first physical modeling synthesizers – the VL1 and VP1 – released in 1993. Research and development has continued relentlessly ever since, and in 2001 the K's Lab team began aiming its formidable technological capabilities at physical modeling for effects, and that's when Mr. Kunimoto's goal began to take on primary importance. The goal? In a word: "musicality."

The K's Lab team were aware that the earliest effect modeling technologies were focused more on superficial reproduction of specific characteristics and tonalities than on actually making music, and it was clear that by applying the same physical modeling technology that was used in the original VL1 and VP1 synthesizers, although in a significantly more evolved form, it would be possible to deliver truly accurate, eminently musical effects. And rather than relying on frequency response graphs and other "precision" measurements to evaluate final performance, many critical performance decisions were made using the trained ears of top-level music and sound specialists.

The Birth of VCM

It took more than two years of concentrated work, but by 2003 K's Lab had refined and re-purposed physical modeling to the point where it was ready for practical implementation ... in the form of Virtual Circuit Modeling. This technology is the cornerstone of Yamaha's VCM plug-ins, and achieves it's stunning sonic and musical performance by actually modeling the individual characteristics of the multitude of parts and components that contributed to the final sound of the original analog circuits: transistors, tape, tape heads, etc. Even subtle saturation effects have been painstakingly modeled to bring the warmth and richness of the original analog gear back to life in stable, easy-to-operate digital form.

Making Space

A new addition to Yamaha's powerful effect arsenal is iSSP (Interactive Spatial Sound Processing). This innovative effect takes surround sound to new levels of reality and creative control. iSSP is actually a combination of two advanced modeling technologies that add up to the most realistic spatial simulation available anywhere:

- Room acoustics modeling that both predicts sound reflection patterns based on room shape, and actually models the decay of the reflections based on source directivity and room surface materials.
- Matrix sound processing that converts source position data to parameters that precisely control the output of each matrix channel, and simulates distance-related decay through delay and filter processing.

DM/0 VCM Series Lineup

Comparison Chart

	DM2000VCM	02R96VCM	DM1000VCM	01V96VCM
Input (Mixing Capacity)	96 in@96 kHz	56 in@96 kHz	48 in@96 kHz	40 in@96 kHz
Mic Input (Head Amp)	24 (XLR/TRS)	16 (XLR/TRS)	16 (XLR)	12 (XLR/TRS)
Line Input	24 Channel Inserts	8 Line in, 16 Channel Inserts	4 OMNI in	4 Line in, 12 Channel Inserts
Outputs	1 Stereo Out, 8 OMNI Out, 24 Insert Out, 1 Studio Monitor Out, 2 C-R Monitor Out, 1 Phones	1 Stereo Out, 8 OMNI Out, 16 Insert Out, 1 Studio Monitor Out, 1 C-R Monitor Out, 1 Phones	12 OMNI Out	1 Stereo Out, 4 OMNI Out, 12 Insert Out, 1 Monitor Out, 1 Monitor Out, 1 Phones
Digital Inputs	2TR IN DIGITAL (2 x AES/EBU, IEC-60958), CASCADE IN	2TR IN DIGITAL (AES/EBU, 2 x IEC-60958), CASCADE IN	2TR IN DIGITAL (AES/EBU, IEC-60958)	2TR IN DIGITAL (IEC-60958), ADAT IN
Digital Outputs	2TR OUT DIGITAL (2 x AES/EBU, IEC-60958), CASCADE OUT	2TR OUT DIGITAL (AES/EBU, 2 x IEC-60958), CASCADE OUT	2TR OUT DIGITAL (AES/EBU, IEC-60958)	2TR OUT DIGITAL (IEC-60958), ADAT OUT
Bus	8 mix buses, 12 AUX, Main ST Bus	8 mix buses, 8 AUX, ST bus	8 mix buses, 8 AUX, Main ST Bus	8 mix buses, 8 AUX, ST bus
Matrix	4 Stereo	—	—	—
MY Card slots	6	4	2	1
Faders	24+1	24+1	16+1	16+1
Multi Effects/Graphic EQ	8 / 6	4 / —	4 / —	4 / —
Dimensions (W x H x D)	906 x 257 x 821 mm (35.7" x 10.2" x 32.3") With MB and SP: 968 x 371 x 883 mm (38.1" x 14.6" x 34.8")	667 x 239 x 697 mm (26.3" x 9.4" x 27.4") With MB and SP: 700 x 352 x 762 mm (27.6" x 13.9" x 30.0")	436 x 200 x 585 mm (17.2" x 7.9" x 23.0") With MB and SP: 486 x 295 x 635 mm (19.1" x 11.6" x 25.0")	436 x 150 x 540 mm (17.2" x 5.9" x 21.3")
Weight	43.0 kg (94.8 lbs) With MB and SP: 51.6 kg (113.8 lbs)	34.0 kg (75 lbs) With MB and SP: 39.4 kg (86.9 lbs)	20.0 kg (44.1 lbs) With MB and SP: 23.6 kg (52.0 lbs)	15.0kg (33.1 lbs.)

* All "MB (Meter Bridge)" and "SP (Side Pad)" are optional.

DM2000VCM (Rear Panel)

02R96VCM (Rear Panel)

Connectors Gallary

* DM2000VCM and 02R96VCM's rear panels shown with optional MY Cards installed.
* All "SP (Side Pad)" and MY cards are optional.

01V96VCM (Front Panel)

DM1000VCM (Rear Panel)

01V96VCM (Rear Panel)

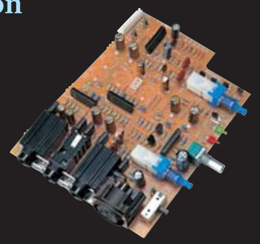
Superior Sound and Control in Any Application

Although they vary in capacity, the four VCM series digital mixing consoles presented in this catalogue share a significant number of features that characterize Yamaha digital consoles, and in many ways account for their leading position in the production, sound reinforcement, and broadcast fields.

Dedicated Yamaha DSP LSIs Deliver Unprecedented 24bit, 96kHz Performance

Unlike digital consoles that achieve operation in 96K mode with reduced number of tracks, Yamaha digital mixing consoles impose no such limitations at any of the provided sampling rates. In all Yamaha VCM series consoles, 96kHz with 32bit internal processing is the standard. Multiple high-resolution Yamaha DSP7 and DSP6 LSIs are used for mix processing and effects, achieving performance that stretches the limits of the most advanced technology currently available. By way of comparison, the original Yamaha 02R – the digital mixing console that almost single handedly started the digital production revolution, and rapidly became the industry standard – had only one ninth of the processing power provided by today's DM2000VCM console. We're talking about super-clean, super-dynamic, noise-free 24bit/96kHz audio, plus all the additional effects and processing you'll ever need for most applications.

Outstanding Mic Preamps with Onboard 24bit/96kHz AD/DA Conversion



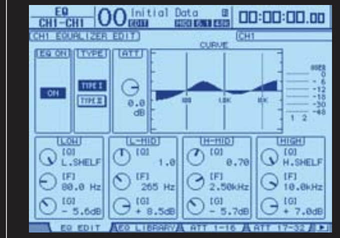
Taking the high-resolution discussion a step further, what about analog-to-digital and digital-to-analog conversion? The same applies: if you don't have all 24bits at the full 96kHz in top-quality converters, you're definitely going to be missing something (i.e. part of your sound). Once again, the VCM series digital mixing consoles impose no limitations. All onboard A/D and D/A conversion makes use of top-performance 24bit/96kHz converters. This is particularly important in these consoles because they feature some of the finest analog microphone preamps available in any console, anywhere. The on-board converters ensure that you get an excellent digital representation of the warm, transparent output from these remarkable preamps. A range of Mini-YGDAI digital and analog I/O cards also provide full 24bit/96kHz capability.

Internal Effects Fully Support 96kHz Processing

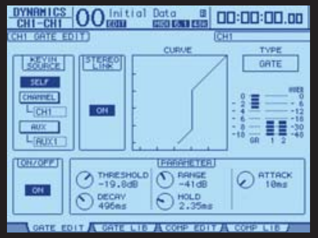
What's the point of having 24bit/96kHz audio if you have to convert down to a lower sampling rate for effect processing? That's exactly what's happening if you're using hardware or software processors that don't offer 24bit/96kHz performance anywhere in your signal chain. That's why Yamaha included a comprehensive range of 96kHz compatible stereo effects – plus several designed specifically for surround. And you can use multiple effect processors simultaneously!

Top-quality Compression, Gating, EQ and Delay

These powerful mixing consoles feature flexible, independent channel compression and gating/ducking processors for dynamics control, 4-band parametric channel equalizers that offer extra versatility with switchable "type I" or "type II" EQ algorithms to deliver the response you prefer, and a channel delay.
The equalizers offer particularly fine control with ±18dB gain in 0.1dB increments, full frequency sweep from 20Hz to 20kHz, and 41-point variable Q on each band. There's also surround pan for up to 6.1 mixes on all models except the 01V96VCM, stereo pan, and phase switching. All channel functions except gating are also provided on all output buses (main, auxiliary, and stereo). An extensive list of preset EQ, Comp, and Gate "libraries" are provided for fast, easy setup, and user libraries are available for storage and instant recall of your own setups. Comp/gate gain reduction metering is also provided on the meter display.



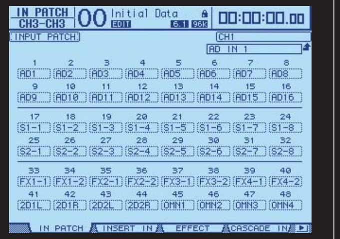
EQ Edit



Gate Edit

Extraordinary Patching Flexibility

All available inputs, outputs, effects, and channel inserts can be assigned to any of the console's channels or outputs via a remarkably versatile, easy-to-use digital patching system. For example, any of the effect processors can be assigned to an auxiliary buss for send-type operation, or inserted directly into any input channel as required. A direct out function also allows the signal from any input channel to be routed directly to any digital or analog output in the system. The DM2000VCM features an additional matrix mix system that can be used to provide cue monitor mixes, downmix monitoring for surround production, or zone level control for sound reinforcement applications.



Input Patch

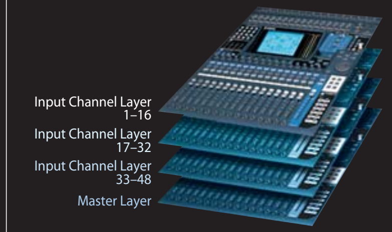
Comprehensive Automation and Scene Control

Automation and Scene Memory recall capability are essential elements of modern digital mixing consoles. Yamaha takes these functions to the highest level of precision and ease-of-use. While providing full automation of virtually all parameters, these consoles feature smooth and quiet motorized faders that make writing and updating automated mixes faster and more intuitive than ever. And all automation data is recorded at 1/4-frame accuracy to ensure excellent precision. Global Fade Time and Global Recall Safe are provided in addition to independent fade time and recall safe settings for each scene, so you can set global fade time and recall safe settings that apply to all scenes. This ability can dramatically reduce setup time when you will be using multiple scenes with the same settings.

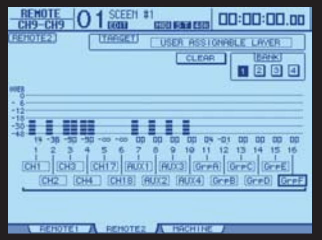
Input Fade Time

Layered Channels

One of the advantages of working with digital is that it allows maximum power and flexibility to be packed into minimum space. The DM2000VCM's 24 precision 100-millimeter motorized channel faders, for example, can be instantaneously layer-switched to control any of 96 channels. On the DM1000VCM you have 48 channels in three layers, on the 02R96VCM there are 48 channels in two layers, and the 01V96VCM has 32 channels in two layers. You have maximum channels in minimum space, and switching between layers with the channels right in front of you can be a lot faster and easier than trying to locate a desired channel on a massive spread-out console. More importantly, all operations can be carried out without having to move away from the monitoring "sweet spot".



(ex. DM1000VCM)



User Assignable Layer

Versatile Channel Pairing and Grouping Functions

In addition to being able to pair faders "horizontally," corresponding faders in adjacent layers can be "vertically" paired, allowing each physical channel fader to be used for stereo channel control. Multiple stereo channels can thus be controlled from a single layer with a whole list of linked parameters. You can group faders and mutes from any selected input channels or output busses and store the settings in multiple banks. Furthermore, EQ and compressor parameter settings can be linked for simultaneous operation. The group and link functions are very convenient for various applications including sound reinforcement and surround production.

Intuitive Interface Designed for Maximum Productivity

Anyone who is familiar with the original 02R will immediately feel comfortable with any of the VCM series digital mixers. While the comprehensive, efficient display format of the 02R has been inherited by all models in the series, the control surface and user interface system have evolved to allow analog-style hands-on operation with minimum need to refer to the LCD. The motto: mix with your ears, not with your eyes. User-defined keys, which can be assigned to functions of your choice, are also provided.

I/O Expandability and Plug-in Capability

Extra I/O versatility is provided in the form of Mini-YGDAI expansion slots: six on the DM2000VCM, two on the DM1000VCM, four on the 02R96VCM, and one on the 01V96VCM. The expansion slots are 24bit/96kHz compatible, so you can select I/O and processing cards to provide the input/output configuration and processing capabilities that are perfect for your needs. Whether you need digital I/O in ADAT, TASCAM, or AES/EBU format, CobraNet, EtherSound or MADI connectivity, or extra analog I/O capability, the appropriate Mini-YGDAI cards are available. Third-party cards are also available for other formats and functions.

Mini-YGDAI Slots with MY Cards (DM1000VCM)

Intuitive "Selected Channel" Interface


Selected Channel Section (DM2000VCM)

The Selected Channel controls are the "hands-on" channel controls for the currently selected input and output channel, with analog-style buttons and knobs for direct access and control of essential parameters. This original Yamaha configuration has become a "classic" as well the de-facto industry standard, and is familiar territory to most experienced engineers.

Integrated DAW Control

The VCM series digital consoles have been designed to integrate tightly with leading digital audio workstations to create a complete production and mixing environment. The new Advanced DAW Control Protocol, initiated by Yamaha and Steinberg, enables you to control DAW software such as Nuendo® and Cubase® as well as other HUI compatible DAW applications from the console's Selected Channel section, including full control of mixing and processing parameters plus transport/track-arming control and access to editing functions. (Controllable functions vary depending on the DAW software and version you are using.)

Advanced Surround Solutions

These advanced consoles also provide facilities for comfortable surround processing and production. Features you need for surround processing, panning and monitoring – including a joystick on all models except the 01V96VCM – are provided as standard equipment. The joystick is the perfect (and generally preferred) tool for smooth, continuous positioning of 5.1 or 6.1 surround sound for DVDs or other surround media. Although the 01V96VCM does not have a joystick for surround panning, it does allow for 5.1 output. The DM2000VCM, DM1000VCM and 02R96VCM offers ideal surround mixing environments including a downmix matrix which can deliver 3-1 (LCRS) and stereo mixes while you are burning a surround mix to DVD, bass management, and speaker alignment facilities for optimum speaker system tuning. They will even handle multiple surround stem mixes with ease.

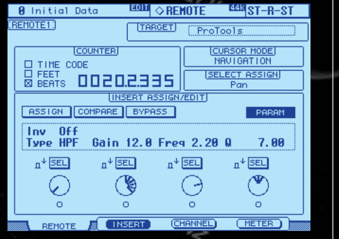
Studio Manager Version 2 Software Supplied

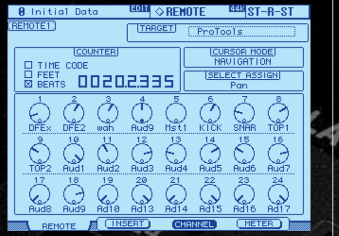
Control from a personal computer? Of course! And Yamaha even supplies the software. The VCM series consoles come with the Studio Manager application for both Macintosh® and Windows® platforms, allowing total control and management of all parameters via a comprehensive graphic interface. Studio Manager Version 2 offers even more advanced networking potential than the original version, functioning as a complete central management system for digital mixing, including a sophisticated visual interface for editing and operating VCM and iSSP effects.

For Production

Whether you're recording basic tracks or crafting a complex final mix, the VCM series digital mixing consoles provide a stress-free working environment for all types of audio production.

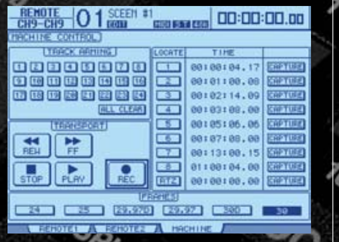
Nuendo® and HUI DAW Control


Remote (Insert)


Remote (Channel)

When used with an HUI-compatible DAW system such as Steinberg's Nuendo®, the VCM series consoles provide physical control of mixer functions as well as recorder control. By simply connecting the console to a computer running the HUI DAW system via the TO HOST connector (combined USB and serial), the console's faders and encoders can be used for DAW control to create a seamless, efficient production environment. Standard libraries of control functions are provided for Nuendo® and other HUI-compatible applications, and these can be assigned to the console's remote layer as required. What's more, just about any other DAW package can be accommodated via a General DAW Mode. And if your DAW features surround panning functionality, you can achieve direct control of this panning via the console's joystick (provided on all but the 01V96VCM).

Machine and Locate Control


Machine Control

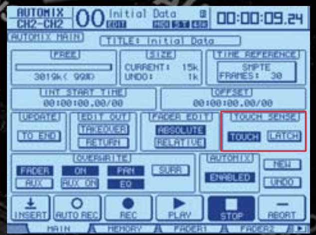
Since the VCM series consoles will usually be used with some sort of multitrack recorder – tape, hard-disk, or DAW – they have been provided with a comprehensive range of facilities for external machine control. The MMC protocol is supported, and control can be switched between MTR and master target machines. The DM2000VCM and DM1000VCM can control external recorders via P2 commands, while the DM1000VCM also supports the ESAM II machine protocol used in the broadcast Industry. The consoles also allow MMC equipment to be remotely controlled directly from the DAW layer, so you can simultaneously control a DAW and MMC recorders without having to switch layers. The DM2000VCM and 02R96VCM additionally feature eight direct locate keys for fast, easy location and cueing, while all consoles in the series feature user defined keys that can be assigned as dedicated tape transport or track arming buttons.

Automix Static Insert


Automix Static Insert

With the sole exception of the 01V96VCM, the VCM series consoles offer an automix static insert function that allows pre-defined parameter settings to be punched in and out to, for example, adjust the EQ for a short dialog sequence during preproduction.

Touch-sensitive Intelligent Automix Parameter Punch I/O


Touch Sense

This flexible system provided in the DM2000VCM, DM1000VCM and 02R96VCM lets you set up the controls for the most efficient operation according to the signal flow and mixing task at hand. A fader touch-sense function allows automix parameter punch in/out operations to be carried out with unprecedented speed and efficiency. When a fader is touched the parameter for that fader is punched in and the automix parameter overwrite mode is engaged. Two modes are provided: in the TOUCH mode the fader parameter is punched out and overwrite ends when the fader is released, and in the LATCH mode overwrite continues even after the fader is released.

Built-In ADAT Optical Interface (01V96VCM only)

ADAT Connector (01V96VCM)

The 01V96VCM comes with an industry-standard ADAT optical digital I/O interface built right in – no options necessary. The ADAT "Lightpipe" optical I/O is standard on a wide range of current digital sound gear, so you can simply plug in via optical cables for 8 digital inputs and 8 digital outputs that will handle your digital signals without compromise. Additional optical I/O capacity can be added via the 01V96 expansion slot, as necessary.

For Live Sound

You'll find the VCM series digital mixing consoles and their predecessors hard at work in a variety of live sound applications, in installations as well as on the road.

Channel Functions and Effects

Many of the functions that make the VCM series digital mixing consoles perfect for production are a boon for sound reinforcement, too: independent gates and compressors on every channel, 4-band fully parametric EQ, delay, and more. All main buses, auxiliary buses, and the stereo bus feature the same channel functions (except gating) for extraordinary control. The VCM series consoles give you all of the ambience and other effects you need for sound reinforcement without having to drag an outboard effect rack around. Live sound engineers love the REV-X reverbs, and many find the Open Deck effects ideal for creating a warm, analog vibe. But setting up EQ, compression, and other parameters for a mix from scratch can be a daunting task, so Yamaha has provided an extensive selection of presets in a range of "libraries" that can simply be selected and used unmodified, or edited to suit specific requirements. Libraries are provided for effects, compression, gating, EQ, I/O patching, and channel setups. Of course, your own setups can be added to the libraries for instant recall whenever they are needed.

COMP276 (CHANNEL STRIP)

OPEN DECK (MASTER STRIP)

REV-X (REVERB)

Remote Head Amp Control

By using highest quality 8-channel Yamaha AD8HR head amp/AD converter* units you can set up a top-performance, high input capacity sound reinforcement system with fully digital transmission from the head amps to the console. And since the AD8HR is remotely controllable via the DM2000VCM or DM1000VCM "REMOTE" connector, your head amps can be set up on stage for maximum sound quality and minimum in-house cabling. The digital output from the head amplifiers can be transmitted to the console in AES/EBU format over distances of up to 200 meters (@44.1/48kHz) with absolutely no loss in signal quality or added noise.

* The AD8HR operates at 44.1/48/88.2/96kHz.

AD8HR

Total Recall

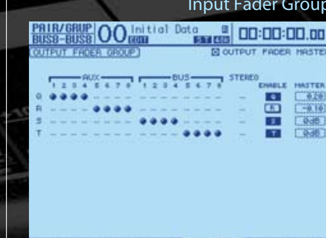
The ability to store and recall all console parameters in an instant is a huge advantage for sound reinforcement applications. Of course it allows you to instantly switch "scenes" during a performance, but it also lets you recall the basic settings for a show at each venue, and then tweak to optimize the sound for that environment. This can dramatically reduce setup time. All scene data can also be managed on a personal computer using the supplied Studio Manager V2 software. You can do basic setup on your laptop, and then transfer the data to the console at the venue. The VCM series digital mixing consoles also let you recall a scene with fade time, or apply "recall safe" for only the specified parameters and channels, or globally – for added creative control and flexibility. There's even a global paste function that lets you simultaneously paste selected parameters from one scene to multiple scenes – your EQ and AUX settings from final rehearsal, for example, can easily be copied to all other scenes that will be used during the performance.

Fader Group Master

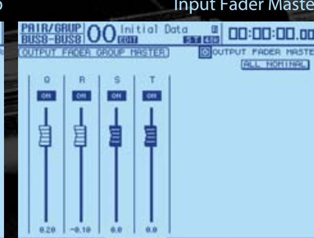
A group master function allows convenient group control of specified faders. By assigning the input and output fader masters to the console's user-assignable layer you have the operational equivalent of a large analog console equipped with multiple VCA faders.

Input Fader Group

Input Fader Master



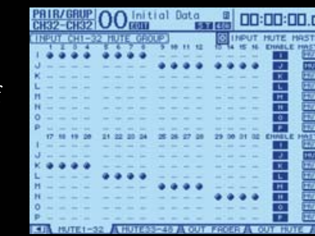
Output Fader Group



Output Fader Master

Group Master Mute

For convenient multi-channel muting a group master mute function can be assigned to the user defined keys. Any of the console's inputs and outputs can be assigned to mute groups as required, then muting of the assigned group can be engaged or disengaged with one touch via the user defined keys – a tremendous advantage in live sound applications.

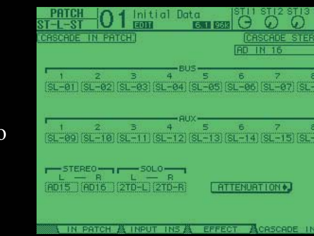


Input Mute

Output Mute

01V96 Cascade Link (01V96VCM only)

When you really need high capacity – particularly for sound reinforcement applications – the 01V96 offers "01V96 Cascade Link" capability that allows two 01V96 units to be cascaded to create up to an 80-channel mixing system at an unbelievably affordable price!



Cascade in Patch

Other Functions for Flexible Live Sound Mixing

- Global Fade Time and Global Recall Safe (not provided on the 01V96VCM)
- AUX Pre-Fader/Pre-ON(not provided on the 01V96VCM)
- AUX/SOLO Link for Instant AUX Monitoring (not provided on the 01V96VCM)
- Gain Reduction Meter

For Broadcast

Most of the VCM series digital mixing consoles offer a number of features that make them an ideal choice for broadcast applications as well as production and live sound. The features described below apply to the DM2000VCM, DM1000VCM, and 02R96VCM.

Fader Solo Release and Pre-Fader with Pan

The fader solo release and pre-fader with pan functions included in the VCM series consoles will be of particular interest to broadcast engineers. Fader solo release allows instant, automatic switchover from solo source monitoring to mixing. Pre-fader with pan also provides a post-pan monitoring option.

Versatile Channel Pairing and Grouping Functions

In addition to being able to pair faders "horizontally", corresponding faders in layers 1 and 2 can be "vertically" paired, allowing each physical channel fader to be used for stereo channel control. A number of stereo channels can thus be controlled from a single layer with a whole list of linked parameters.

Mix Minus

Here's a VCM series feature that's specifically aimed at broadcast applications. Mix Minus operation is essential for broadcast-to-telephone interfaces and other on-air setups where certain signals must be eliminated from a feed to prevent feedback and unwanted echo effects.

Mix Minus

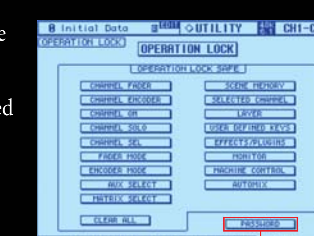
Dual Oscillator

This versatile dual oscillator is capable of simultaneously sending 400Hz and 1kHz sine waves to the L, R and odd/even buses, respectively, to check the signal path.

Oscillator

Operation lock

Broadcast "accidents" are a serious issue for broadcast facilities, and must be avoided at all cost. A password-protected operation lock feature can be used to "lock" specified functions and parameters so that they cannot be accessed by unauthorized personnel or accidentally altered during critical live broadcasts.



Set Up a Password

GPI Interface

A standard GPI interface provides control interoperability with other broadcast studio functions, such as fader start and talk back on/off switches.

Built-in MS Decoding

Built-in MS decoding function that eliminates the need for external matrix transformers for MS microphones when you are using one of these consoles for ambient location recording.

Customizable Output Level

Broadcast stations around the world have different requirements for the standard maximum analog output level. The VCM series default level is set at +24dBu. This can be modified at a Yamaha service center upon request to +20, +18, or +15 dBu (please note that this modification is chargeable). A quick solution is provided by an “Output Port Attenuator” menu, which provides attenuation from the default 0dB through -9dB via software.

DM1000VCM Provides ESAM II Support

The DM1000VCM is controllable from many ESAM II compatible video editors, via the REMOTE port connection.

REMOTE Port Pin Assignments

Use the following pin assignment to control the DM1000VCM from a video editor.

① ② ③ ④ ⑤
⑥ ⑦ ⑧ ⑨

Pin#	Video Editor	DM1000VCM REMOTE Port
1	Frame Ground	Frame Ground
2	Receive A	Receive A (*)
3	Transmit B	Transmit B (*)
4	Transmit Signal Common	Transmit Signal Common
5	Spare	Spare
6	Receive Signal Common	Receive Signal Common
7	Receive B	Receive B (*)
8	Transmit A	Transmit A (*)
9	Frame Ground	Frame Ground

* For bidirectional control, cross-connect Pins 2 and 8, and Pins 3 and 7 on an I/O cable.

For Surround

Surround sound is a major and constantly evolving aspect of today’s audio scene, both in the studio and live. The VCM series digital mixing consoles bring you right up to date with advanced surround monitoring and processing features.

Up to 6.1 Surround Monitoring and Processing

SURROUND MONITORING

The DM2000VCM, DM1000VCM and 02R96VCM are fully compatible with 3-1, 5.1 and 6.1 surround processing, panning and monitoring requirements. You can also change the order of the surround channel to bus out assignment according to project requirements. And since accurate monitoring is so essential to surround production, extra care was taken to ensure that these consoles offer the ideal mixing environment, including a downmix matrix which can deliver 3-1 (LCRS) and stereo mixes while you are burning a surround mix to DVD, bass management, and speaker alignment facilities for optimum speaker system tuning. You can even handle multiple surround stem mixes with ease. The 01V96VCM offers basic surround panning and output capability.

Surround Bus Setup

Joystick Surround Panning

The DM2000VCM, DM1000VCM and 02R96VCM are all equipped with joysticks for smooth, continuous positioning of 6.1 surround sound. They also feature an abundance of graphic surround displays – surround pan, trajectory patterns, and parameters – to assist in accurate positioning and efficient “moves”. Surround panning can be turned on or off as required by the application. When off, sources such as dialog that require no panning can be directly fed to the center bus. This capability can simplify signal routing in many situations.

Surround Panning

Joystick (DM1000VCM)

Surround Effects Built In

With the exception of the 01V96VCM, the internal digital effect systems on the VCM series consoles include “Reverb 5.1”, “Comp5.1”, “Expand 5.1”, and a number of other effects specifically designed for surround production. Bus EQ and dynamics can also be grouped for efficient surround processing.

Reverb 5.1

6.1-to-Stereo Downmix Recording

A Bus-to-Stereo function on all VCM series consoles except the 01V96 can be used to provide a 6.1 to stereo, 5.1 to stereo, or 3-1 to stereo downmix recording while you are working on a surround mix. Furthermore, the 3-1 output can be fed to a 2-track master recorder via a Dolby Surround® encoder, and then back to the console via a decoder to allow instant real-time comparison between the pre-encode and post-decode sound.

Snap to SPL 85dB

A “Snap to 85dB” function instantly sets the surround monitor level to the standard 85dB SPL. And if you’re mixing to THX™ standards, you can use a short-cut key to set levels instantly and precisely to the theater-standard of 85dB SPL. In fact, the DM2000VCM, DM1000VCM, and 02R96VCM are the world’s first consoles to receive THX pm3™ Approval, so by combining any of these consoles with Yamaha’s MSP7 STUDIO powered monitor speakers – also THX pm3™ Approved – you have a state-of-the-art production studio that is capable of turning out sound of the highest quality.

Snap to SPL

Comprehensive Bass Management and Other Surround Features

Bass Management is important for optimizing channel signals and subwoofer delivery for the monitoring environment. The DM2000VCM, DM1000VCM and 02R96VCM have 8 preset bass management modes (included 3 THX Presets) for DVD or film mixing and authoring. You can also fine-tune individual filter and attenuation parameters. Other features include an oscillator for testing speakers, individual attenuator and delay parameters for monitor alignment, individual bus (speaker) muting, and overall level control for all monitor outputs.

Bass Management

VCM

THX pm3™ Approval

The DM2000, DM1000 & 02R96 digital mixing consoles are the worlds first digital consoles to come equipped with complete surround monitoring facilities, eliminating the need to connect and feed the signal to external monitoring equipment. They are ideal surround solutions, particularly when used in combination with the MSP7 Studio Powered Monitor Speaker.

Known worldwide for high quality entertainment sound and picture, the THX pm3™ (Professional Multi-Channel Mixing & Monitoring) Studio Certification Program addresses the need for reliable, translatable, and superior performance in professional multi-channel mixing and monitoring studios worldwide. THX has created a performance standard that focuses on the listening and viewing environment, selection of audio and video equipment, layout of the working area, and calibration. The DM2000, DM1000 & 02R96 are included in the THX pm3™ Approved Equipment list as Studio Monitoring Systems, and the MSP7 Studio Powered Monitor Speaker is included as Front & Surround speakers.

The Yamaha DM2000, DM1000 & 02R96 digital mixing consoles have the following surround functions built-in.

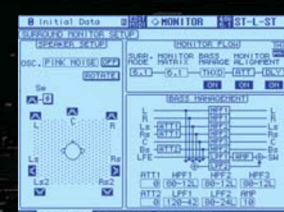
Surround production functions

- Fully compatible with 3-1, 5.1 and 6.1 surround processing, panning and monitoring
- Flexible surround bus set up
- Built-in Joy stick
- Graphical user interface and parameters to assist accurate surround PAN positioning and efficient moves of sound image.
- Built-in surround effects including “Reverb 5.1”, “Comp 5.1”, “Expand 5.1” etc.


Surround Pan Positioning

THX pm3™ Approved surround monitoring functions

- Downmix monitoring matrix
- Bass Management: comprehensive filter and attenuator setting and THX pm3(tm) presets
- Monitor Alignment functions (Attenuator and delay for individual speakers)
- Built-in Oscillator
- “Snap to 85dB SPL” function


Surround Monitor Setup
(THX pm3™ Monitor Flow)

THX Bass Management Presets:

The following presets have been approved by THX™ Ltd. for use in THX pm3™ Certified Studios*. They are designed to provide dedicated parameters for the proper playback of multi-channel audio content in bass managed systems and to be compatible with subwoofer-satellite type consumer systems.

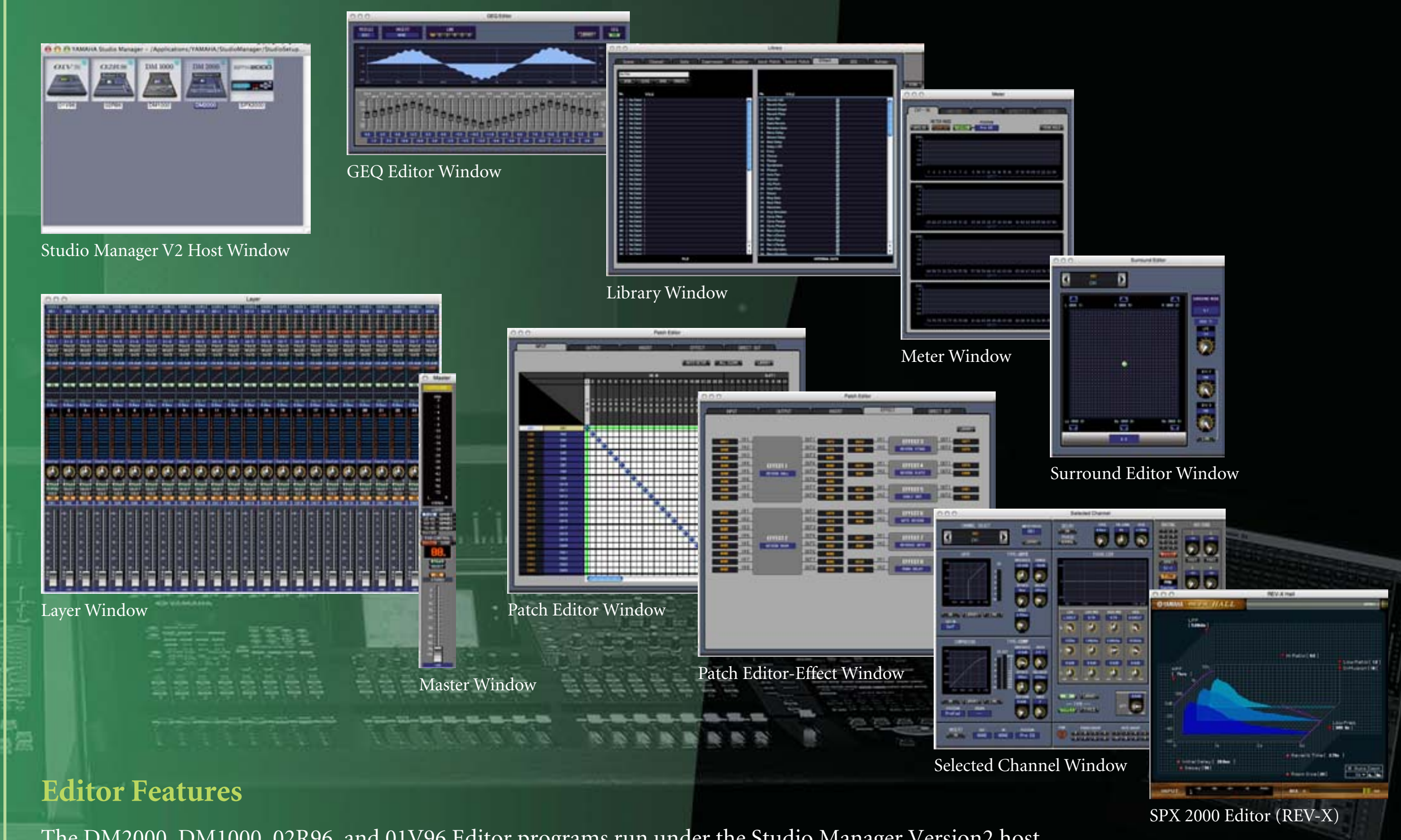
* Use of a THX preset does not permit a studio to use the designation THX pm3™ Certified Studio. The THX pm3™ Studio Certification Program uses performance and design specifications to create calibrated environments for optimum sound and picture presentation. For more information, visit the THX website at http://www.thx.com

[THXD] THX DVD	This preset is configured for DVD-Video production. Use this preset when mixing and/or monitoring audio content not from a theatrical film source. The parameters cannot be changed.
[THXF] THX Film	This preset is configured for Film pre-production. Use this preset when mixing and/or monitoring theatrical film-based content (such as a pre-mix for film). The parameters cannot be changed.
[THXM] THX Music	This preset is configured for DVD-Music production. Use this preset when mixing and/or monitoring multi-channel music content (including DVD-Audio and SACD). Only one parameter can be changed. The LFE gain (AMP) can be set to +10dB (default) or 0dB. Select the level that complies with the standards of the target media. Please note: The LFE output gain on some DVD players, receivers, and/or decoders may already be set to +10dB. Select the 0dB setting only if the destination environment (home theatre, etc.) has the LFE gain set to 0dB. Otherwise, use the default setting.

For more information on THX pm3™, please visit THX website at http://www.thx.com.
Visit Yamaha website at http://www.yamahaproaudio.com/ to find DM2000/1000, 02R96 surround set up manual, Quick Guide and Surround Tutorial Booklet.

Studio Manager Version 2

The original Studio Manager application has undergone a significant evolution and has been reborn as Studio Manager Version 2. The hybrid Windows®/Macintosh® Studio Manager application has been rewritten as a host application which hosts the DM2000, DM1000, 02R96, or 01V96 Editor that actually controls the console, and which can be used simultaneously with other editors for centralized, versatile multi-console control.
Simply connect the console to a computer via its TO HOST port (combined USB/serial), and the computer functions as comprehensive control center for the entire system. You can even open and close Studio Manager Version 2 windows from the console controls, for seamless system integration and optimum operation efficiency in any application. Studio Manager Version 2 also provides an advanced GUI for the VCM plug-ins provided in the VCM series consoles.


Studio Manager V2 Host Window
GEQ Editor Window
Library Window
Meter Window
Surround Editor Window
Layer Window
Patch Editor Window
Master Window
Patch Editor-Effect Window
Selected Channel Window
SPX 2000 Editor (REV-X)

Editor Features

The DM2000, DM1000, 02R96, and 01V96 Editor programs run under the Studio Manager Version2 host application, and offers features and functionality that have been refined and updated for professional-level control. Some of the most significant refinements include:

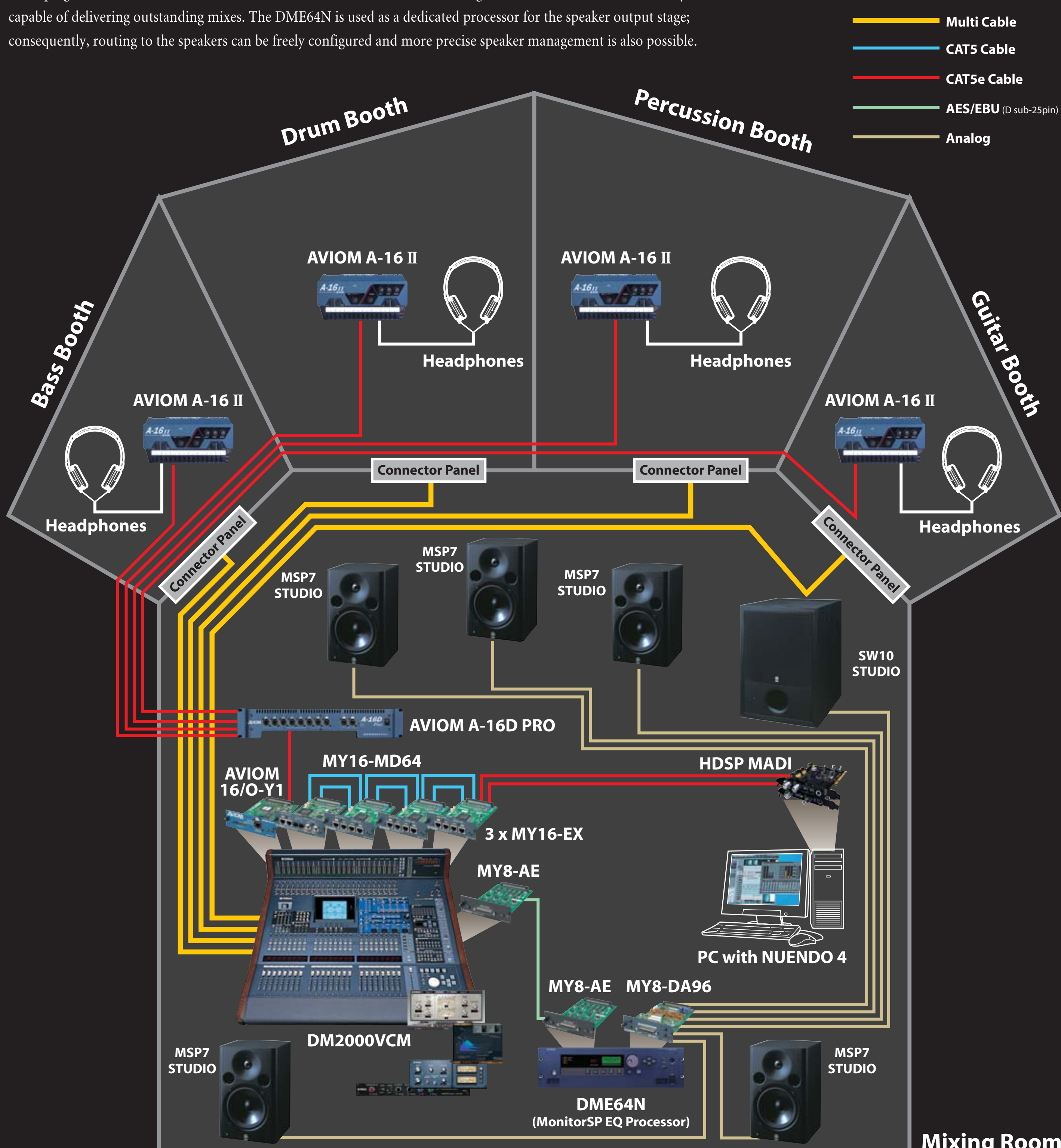
- Master Fader Window provides independent master fader display and control.
- Meter Window shows levels on all channels.
- A new Automix Library Window has been added to the library windows.
- Layer Window allows selection and display of effects and other sources above the panel pan controls.
- Selected Channel Window adds graphic gate displays and long-stroke channel metering.
- Patch Edit Window is now resizable, and displays effect block inputs and outputs.
- Effect Editor Window includes an enhanced interface and fine control for the VCM plug-ins.

Enhanced Editing Windows

Version 2 provides a number of enhanced editing windows, including a “Master Fader Window” that shows only the master fader, and a “Meter Window” that shows meters for all channels. In the “Layer Window,” the master fader block now can be separated, while the fader level is shown by a numeric value as well as graphically. The “Selected Channel Window,” now includes gate type indication and long-stroke channel metering. The “Patch Editor Window” has been redesigned for easier viewing and can be resized as desired, while also showing both inputs and outputs to and from each effect block. In the “Library Window” you can now perform multiple simultaneous scenes/library operations. Studio Manager Version 2 also includes the “Effect Editor Window” for comprehensive control of the VCM plug-ins.

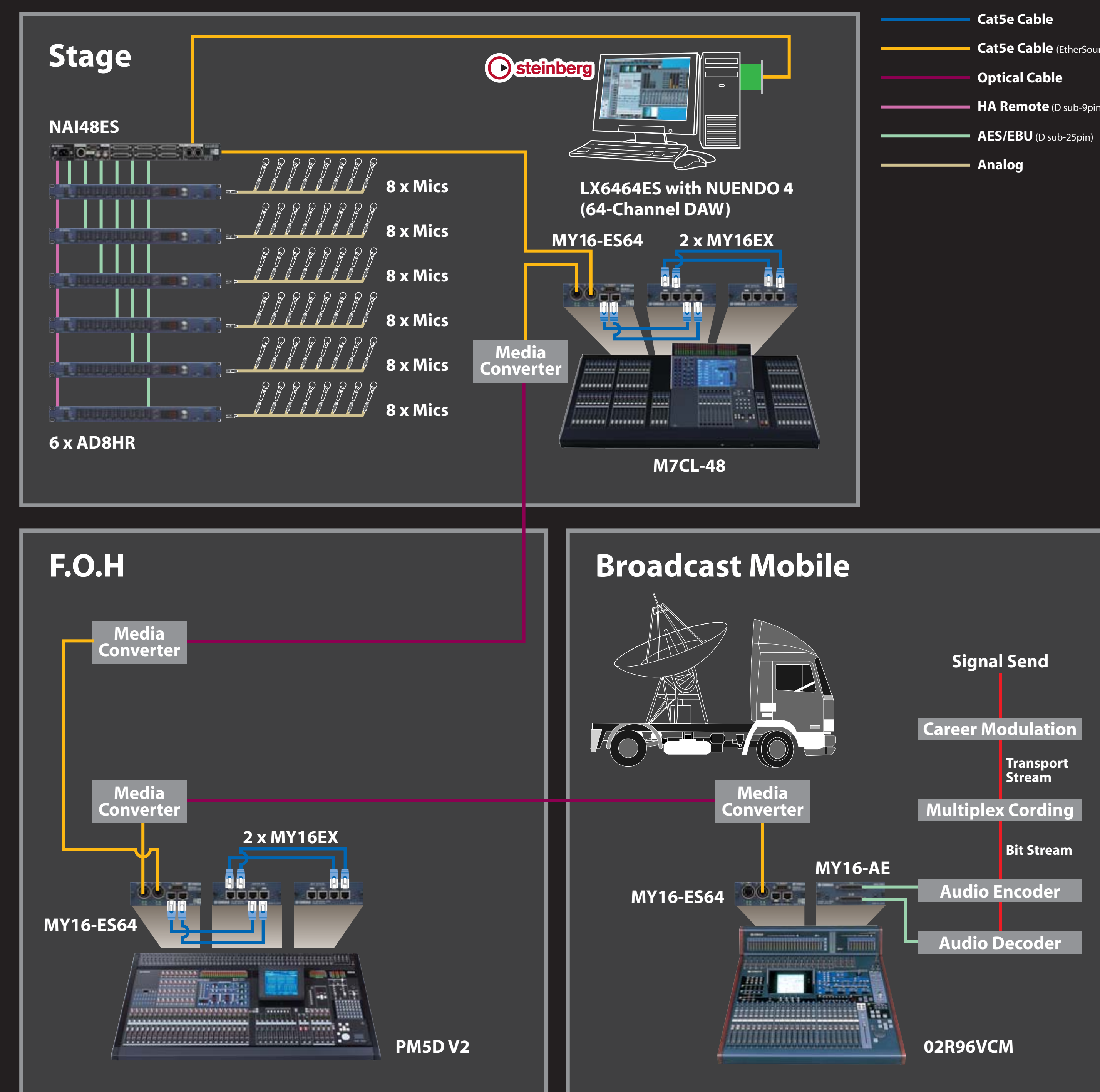
DM2000VCM Surround Recording Studio

The studio system shown here allows advanced 5.1 surround recording, taking full advantage of the DM2000VCM's comprehensive surround features as well as a top-performance monitor system comprised of five MSP7 STUDIO powered monitor speakers and an SW10 STUDIO subwoofer with exceptionally versatile output processing provided by a DME64N Digital Mixing Engine. As THX pm3 certified devices, both the MSP7 STUDIO and DM2000VCM can also be used in surround sound recording, which has seen rapid growth in recent years. The DM2000VCM console receives input from connector panels in the isolation booths while monitor signals are returned to the booths via an Aviom A-NET system using unobtrusive Ethernet cabling. The DM2000VCM is fitted with 64-channel MADI I/O expansion consisting of an MY16-MD64 card and three MY16-EX card that handle data transfer between the console and a computer running Steinberg Nuendo DAW software for recording and editing. A significant advantage provided by the DM2000VCM in this type of application is that the internal VCM plug-ins mean that outboard devices can be minimized or eliminated altogether for a streamlined, efficient system that is capable of delivering outstanding mixes. The DME64N is used as a dedicated processor for the speaker output stage; consequently, routing to the speakers can be freely configured and more precise speaker management is also possible.



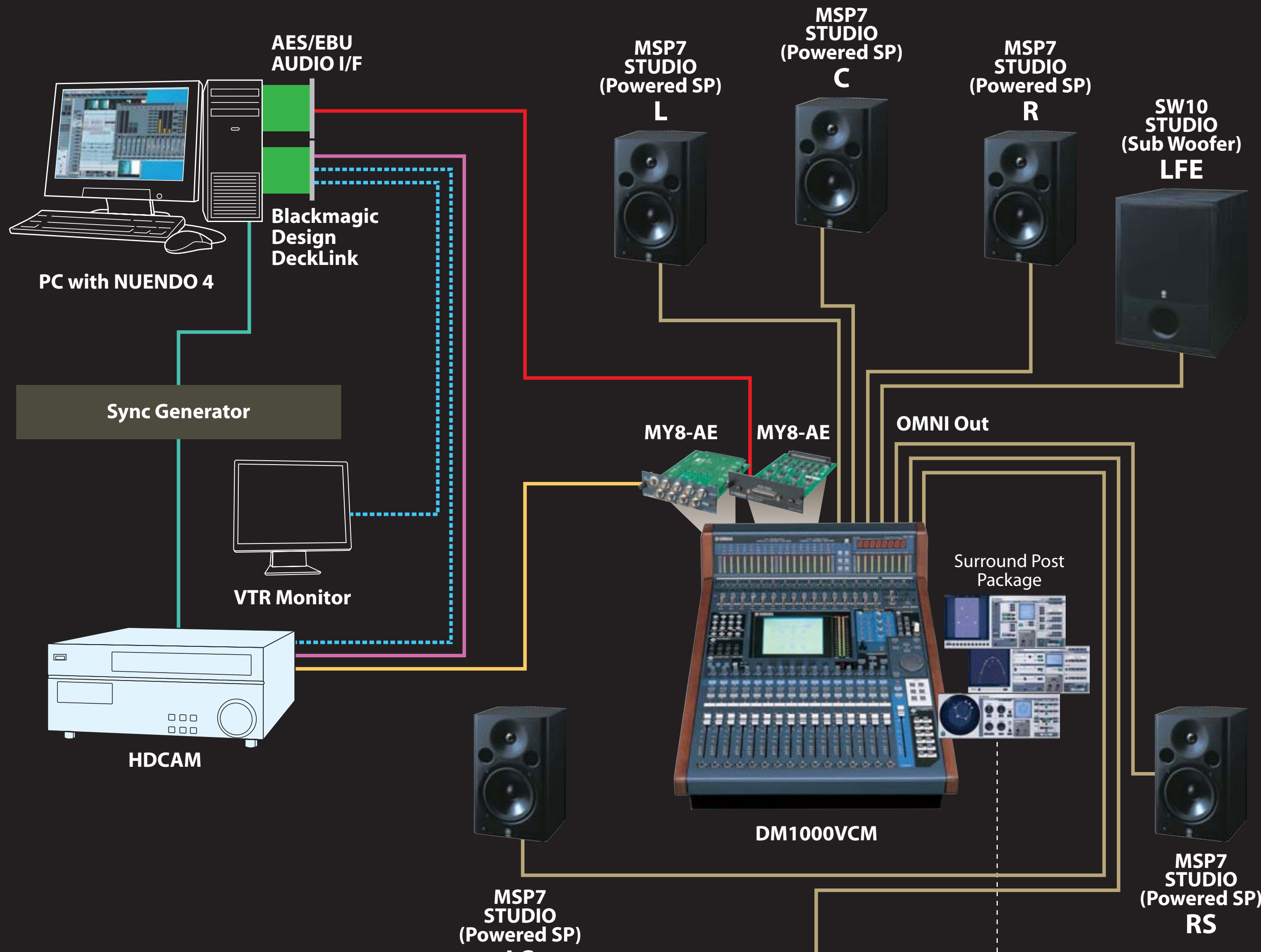
02R96VCM Broadcast Mobile

In this large live optically-connected sound system with a broadcast link, audio is digitally transferred from the FOH mixer to the 02R96VCM console in the broadcast mobile via an EtherSound interface consisting of an MY16-ES64 and two MY16-EX cards installed in the FOH console, and a media converter driving optical fiber cable that allows runs of up to 2,000 meters. The 02R96VCM console receives the optical feed from all 16 of the FOH mixer's buses via a media converter unit. The same signal is returned to a computer running Steinberg Nuendo DAW software at the stage monitor position for live recording, also via EtherSound and optical cable. Output from the console is provided in AES/EBU format via an MY16-AE card, and fed to the appropriate encoding and transmission equipment. Video can be simultaneously processed and transmitted if required. The 02R96VCM is an ideal choice for use in the limited space available in this type of mobile broadcast facility.



Audio-Follow-Video Editing

This video editing studio features a 6.1 surround monitoring system comprised of six MSP7 STUDIO powered monitors and an SW10 STUDIO subwoofer connected to the console's OMNI outputs. The 02R96VCM console includes bass management facilities for full-range playback as well the Surround Post plug-in package for extraordinary surround processing versatility. Both the 02R96VCM and Yamaha's MSP7 STUDIO powered monitor speakers have been officially approved for use in THX pm3™ Certified Studios, and are thus ideal choices for the most advanced video/audio authoring applications. A computer running Steinberg Nuendo DAW software handles audio recording, playback, and editing, and a Blackmagic Design DeckLink card can be installed on the same computer for addition audio/video integration and video capture capability.

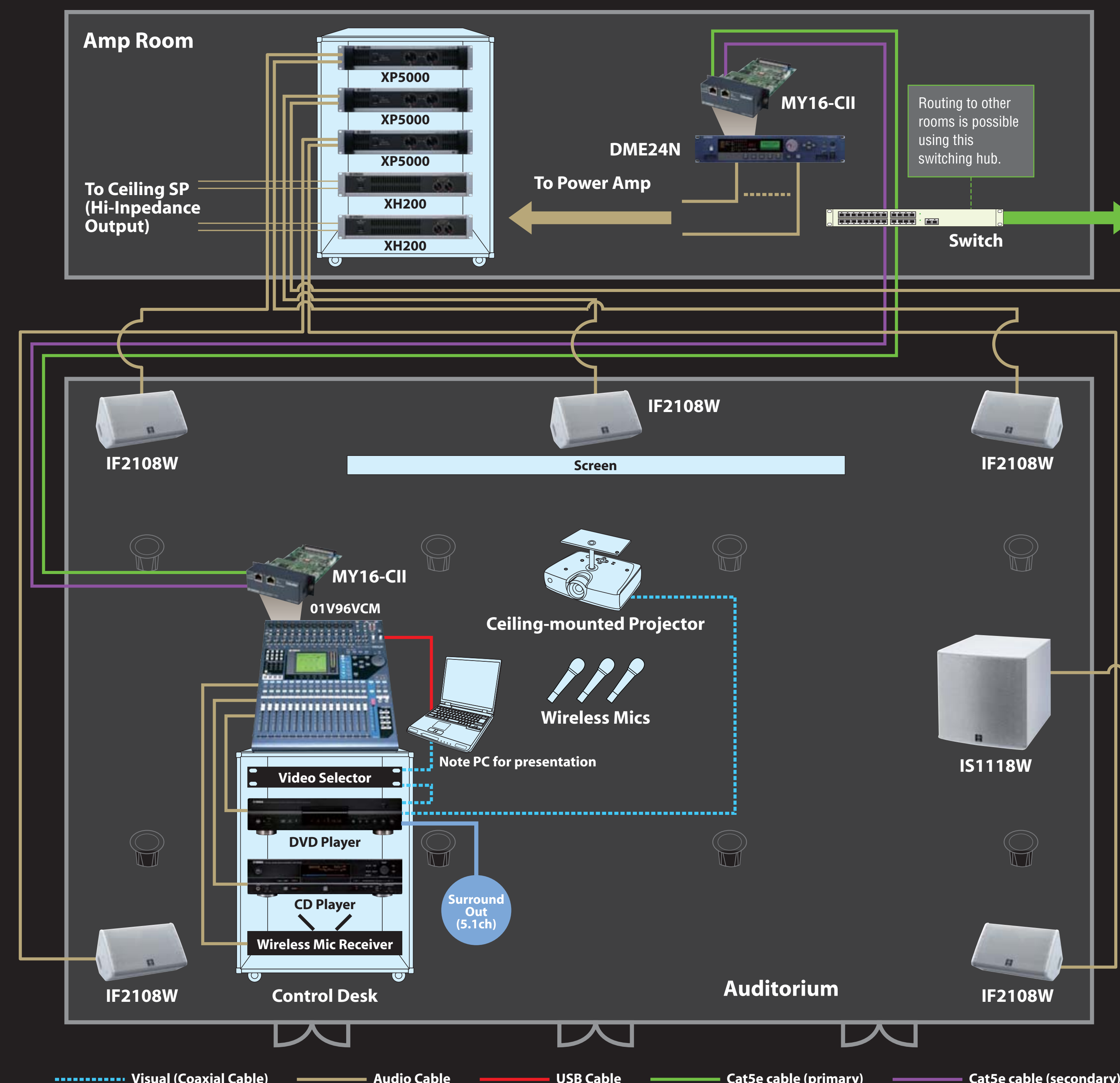


Best plug-in for surround post production

The Surround Post Package uses Yamaha's Interactive Spatial Sound Processing technology that takes full advantage of the 96-kHz audio DSP power of the Yamaha digital consoles. The Surround Post will include three plug-ins: Room ER, Auto Doppler and Field Rotation. These unique plug-ins not only can vastly simplify the complex operation in Post-Production requirements, but also can be used creatively in the musical context. Please refer to page 7.

Corporate AV System

The 01V96VCM is an ideal choice for a mid-size system of this type. In addition to speeches and presentations, the system will handle movie and commercial screenings. The 01V96VCM has enough outputs to deliver 5.1 surround sound, and scene recall and/or the user-defined keys can be used to switch between movie, speech, or other pre-programmed setups on the fly. Output from the console is transferred to a DME24N Digital Mixing Engine in the amp room via a CobraNet network implemented via MY16-CII cards installed in the console and DME unit. CobraNet allows redundant connections for failsafe operation, and sockets can be provided allowing the console to be easily moved to the most convenient location for any type of presentation or other application. The DME unit provides output and speaker processing before feeding the amplifiers that power the main surround speakers and subwoofer as well as a high-impedance ceiling array. Of course the console's built-in VCM plug-ins are a great way to sweeten the impact of movie or presentation sound. A network is realized by placing the DME24N at the core of the installation, and the exchange of audio with other rooms is made possible.

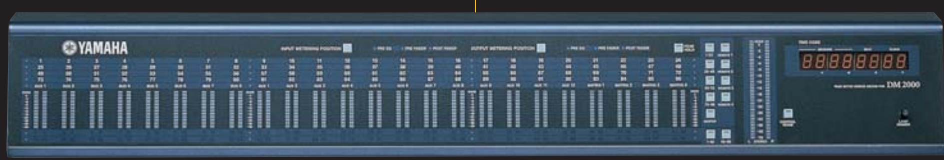


Options

DM2000VCM

Gooseneck Lamp LA5000

Peak Meter Bridge MB2000

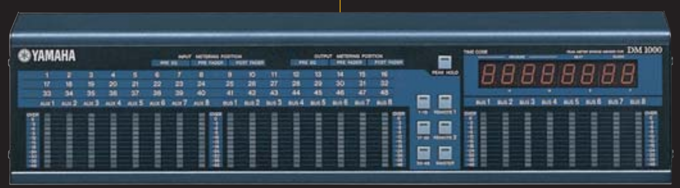


Side Pad SP2000

DM1000VCM

Rack Mount Kit RK1

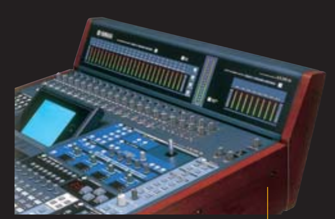
Peak Meter Bridge MB1000



Side Pad SP1000

02R96VCM

Peak Meter Bridge MB02R96



Side Pad SP02R96

01V96VCM

ADD-ON EFFECTS The Add-On Effects packages listed below can be purchased separately and added to the 01V96VCM console.

Rack Mount Kit RK1

Master Strip Package AE-021

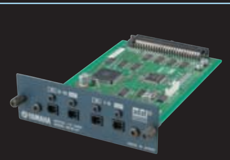
Vintage Stomp Package AE-051

mini-YGDAI Cards

The DM2000's real I/O versatility comes in the form of six mini-YGDAI expansion slots. The expansion slots are 24bit/96kHz compatible, so you can select mini YGDAI plug-in cards to create the input/output configuration that's perfect for your needs. Whether you need digital I/O in ADAT, TASCAM, or AES/EBU format, Ethernet or CobraNet connectivity, extra analog I/O capability, or other functions, the appropriate cards are available.

Digital I/O Cards

MY16-AE
16 channel AES/EBU format I/O



MY16-AT
16 channel ADAT format I/O

MY16-TD
16 channel TDIF format I/O

MY16-CII
16 channel CobraNet™ I/O

MY16-ES64
16 channel EtherSound I/O

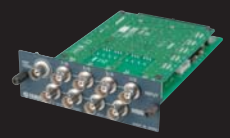
MY16-MD64
16 channel MADI format I/O

MY16-EX
16 channel Expansion I/O

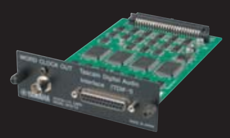
MY8-AE96
8 channel AES/EBU format I/O

MY8-AE96S
8 channel AES/EBU format I/O
(w/Sample rate converter)

MY8-AE
8 channel AES/EBU format I/O



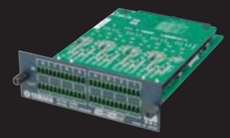
MY8-AEB
8 channel AES/EBU (AES-3id) format I/O



MY8-TD
8 channel TDIF format I/O

MY8-AT
8 channel ADAT format I/O

AD/DA Card

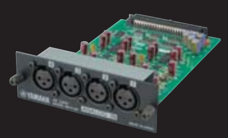


MY8-ADDA96
8 channel Analog I/O

AD Cards

MY8-AD96
8 channel Analog Input Card

MY8-AD24
8 channel Analog Input Card



MY4-AD
4 channel Analog Input Card

DA Cards

MY8-DA96
8 channel Analog Output Card

MY4-DA
4 channel Analog Output Card
(20bit)

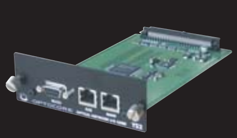
Third Party Cards

AuviTran AVY16-ES
16 channel EtherSound I/O

AVIOM 16/o-Y1
16 channel AVIOM A-Net Output Card

AVIOM 6416Y2
16 channel AVIOM A-Net Input/Output Card

OPTOCORE YG2
16 channel Optocore I/O



OPTOCORE YS2
16 channel Expansion I/O

DM2000VCM Specifications

GENERAL SPECIFICATIONS

Internal processing	32bit (Accumulator=58bit)
Number of scene memories	99
Sampling frequency rate	Internal: 44.1kHz,48kHz,88.2kHz,96kHz External: Normal rate 44.1kHz(-10%) to 48kHz(+6%) Double rate 88.2kHz(-10%) to 96kHz(+6%)
Signal Delay	≥ 2.3 ms CH INPUT to STEREO OUT (fs=48 kHz) ≥ 1.2 ms CH INPUT to STEREO OUT (fs=96 kHz)
Total harmonic distortion *1 Input Gain=Min.	CH INPUT to STEREO OUT ≥ 0.05%, 20Hz to 20kHz @+14dBu into 600Ω ≥ 0.01%, 1kHz @+18dBu into 600Ω (@Sampling frequency=48kHz) ≥ 0.05%, 20Hz to 40kHz @+14dBu into 600Ω ≥ 0.01%, 1kHz @+18dBu into 600Ω (@Sampling frequency=96kHz)
Frequency response	CH INPUT to STEREO OUT 0.5, -1.5dB, 20Hz to 20kHz @+4dBu into 600Ω (@Sampling frequency=48kHz) 0.5, -1.5dB, 20Hz to 40kHz @+4dBu into 600Ω (@Sampling frequency=96kHz)
Dynamic range (maximum level to noise level)	110dB typ, DA Converter(STEREO OUT) 108dB typ, AD+DA(to STEREO OUT)
Hum & noise level *2 (20Hz to 20kHz) Rs=150ohms Input Gain=Max Input Pad=0dB Input Sensitivity=-60dB	-128dBu Equivalent Input Noise -94dBu residual output noise. STEREO OUT STEREO OUT off -94dBu(96dB S/N) STEREO OUT STEREO fader at nominal level and all CH INPUT faders at minimum level -64dBu(68dB S/N) STEREO OUT STEREO fader at nominal level and one CH INPUT fader at nominal level
Crosstalk(@1kHz) Input GAIN=min	80dB adjacent input channels(CH1-24) 80dB input to output
Power requirements	Japan: AC100V 50/60Hz, 300W North America: AC120V, 60Hz, 300W Other Areas: AC220-240V, 50/60Hz, 300W
Dimensions (W x H x D)	DM2000: 906 x 257 x 821 mm (35.7" x 10.2" x 32.3") With MB and SP: 968 x 371 x 883 mm (38.1" x 14.6" x 34.8")
Weightt	DM2000: 43.0 kg (94.8 lbs) With MB and SP: 51.6 kg (113.8 lbs)

*1. Total Harmonic Distortion are measured with a 6dB/octave filter @80kHz
*2. Hum&Noise are measured with 6dB/octave filter @12.7kHz;equivalent to a 20kHz filter with infinite dB/octave attenuation.

ANALOG INPUT CHARACTERISTICS

Input terminal	PAD	GAIN	Actual load impedance	For use with nominal	Input level			Connector
					Sensitivity	Nominal	Max. before clip	
CH INPUT A/B 1-24	0	-60dB	3kΩ	50-600Ω Mics & 600Ω Lines	-70dBu	-60dBu	-46dBu	A:XLR3-31 type (Balanced) B:TRS Phone jack (Balanced)
	0	-16dB			-26dBu	-16dBu	-2dBu	
	26	-16dB			0dV	+10dBu	+24dBu	
INSERT IN 1-24			10kΩ	600Ω Lines	-6dBu	+4dBu	+18dBu	TRS Phone jack (Balanced)
2TR IN ANALOG1(L,R)			10kΩ	600Ω Lines	+4dBu	+4dBu	+18dBu	TRS Phone jack (Balanced)
2TR IN ANALOG2(L,R)			10kΩ	600Ω Lines	-10dBu	-10dBu	+4dBu	RCA pin jack (Unbalanced)

ANALOG OUTPUT CHARACTERISTICS

Output terminals	Actual source impedance	For use with nominal	GAIN SW	Output terminals		Connectors
				Nominal	Max. before Clip	
STEREO OUT (L,R)	600Ω	10kΩ Lines	—	-10dBV	+4dBV	RCA pin jack (Unbalanced)
	150Ω	600Ω Lines	—	+4dBu	+18dBu	XLR3-32 type (Balanced)
STUDIO MONITOR OUT(L,R)	150Ω	10kΩ Lines	—	+4dBu	+18dBu	TRS Phone jack (Balanced)
C-R MONITOR OUT LARGE(L,R)	150Ω	600Ω Lines	—	+4dBu	+18dBu	XLR3-32 type (Balanced)
C-R MONITOR OUT SMALL(L,R)	150Ω	600Ω Lines	—	+4dBu	+18dBu	XLR3-32 type (Balanced)
OMNI OUT 1-8	150Ω	10kΩ Lines	+18dBu (default)	+4dBu	+18dBu	TRS Phone jack (Balanced)
			+4dBu	-10dBu	+4dBu	
INSERT OUT 1-24	600Ω	10kΩ Lines	—	+4dBu	+18dBu	TRS Phone jack (Balanced)
PHONES	100Ω	8Ω Lines	—	4mW	25mW	ST Phone jack (Unbalanced)
		40Ω Lines	—	12mW	75mW	

DIGITAL INPUT CHARACTERISTICS

Terminal		Format	Data length	Level	Connector
2TR IN DIGITAL	1	AES/EBU	24bit	RS422	XLR3-31 type (Balanced)
	2	AES/EBU	24bit	RS422	XLR3-31 type (Balanced)
	3	IEC-60958	24bit	0.5Vpp/75Ω	RCA pin jack
CASCADE IN		—	—	RS422	D-sub Half Pitch Connector 68P (female)

DIGITAL OUTPUT CHARACTERISTICS

Terminal		Format	Data length	Level	Connector
2TR OUT DIGITAL	1	AES/EBU (Professional use)	24bit	RS422	XLR3-31 type (Balanced)
	2	AES/EBU (Professional use)	24bit	RS422	XLR3-31 type (Balanced)
	3	IEC-60958 (Consumer Use)	24bit	0.5Vpp/75Ω	RCA pin jack
CASCADE OUT		—	—	RS422	D-sub Half Pitch Connector 68P (female)

0dB=0.775Vrms; 0dBV=1.00Vrms

CONTROL I/O CHARACTERISTICS

Terminal		Format	Level	Connector in Console
TO HOST	Serial	—	RS422	Mini DIN Connector 8P
	USB	USB 1.1	0 V~3.3 V	B type USB connector
MIDI	IN	MIDI	—	DIN Connector 5P
	OUT	MIDI	—	DIN Connector 5P
	THRU	MIDI	—	DIN Connector 5P
TIME CODE IN	MTC	MIDI	—	DIN Connector 5P
	SMPTE	SMPTE	Nominal –10 dB/10kΩ	XLR-3-31 type (Balanced) *1
WORD CLOCK	IN	—	TTL/75Ω (ON/OFF) *2	BNC Connector
	OUT 1, 2	—	Nominal –10 dB/10kΩ	BNC Connector
CONTROL		—	—	D-SUB Connector 25P (Female)
REMOTE		—	RS422	D-SUB Connector 9P (Male)
KEYBOARD		PS/2	—	DIN Connector 6P
STORAGE CARD SLOT		—	—	SmartMedia slot
METER		—	RS422	D-SUB Connector 15P (Female)

*1. XLR-3-31 type connectors are balanced (1=GND, 2=HOT, 3=COLD).
*2. This switch is on the rear panel.

DM1000VCM Specifications

GENERAL SPECIFICATIONS

Internal processing	32bit (Accumulator=58bit)
Number of scene memories	99
Sampling frequency	Internal: 44.1kHz,48kHz,88.2kHz,96kHz External: Normal rate 44.1kHz(-10%) to 48kHz(+6%) Double rate 88.2kHz(-10%) to 96kHz(+6%)
Signal Delay	≤ 1.6 ms CH INPUT to OMNI OUT (fs=48 kHz) ≤ 0.8 ms CH INPUT to OMNI OUT (fs=96 kHz)
Total harmonic distortion *1 Input Gain=Min.	CH INPUT to OMNI OUT ≤ 0.05%, 20Hz to 20kHz @+4dBu into 600Ω ≤ 0.01%, 1kHz @+18dBu into 600Ω (@Sampling frequency=44.1/48kHz) ≤ 0.05%, 20Hz to 40kHz @+14dBu into 600Ω ≤ 0.01%, 1kHz @+18dBu into 600Ω (@Sampling frequency=88.2/96kHz)
Frequency response	CH INPUT to OMNI OUT 0.5, -1.5dB 20Hz - 20kHz @+4dBu into 600Ω (@Sampling frequency=44.1/48kHz) 0.5, -1.5dB 20Hz - 40kHz @+4dBu into 600Ω (@Sampling frequency=88.2/96kHz)
Dynamic range (maximum level to noise level)	110dB typ, DA Converter (OMNI OUT) 106dB typ, AD+DA (to OMNI OUT)
Hum & noise level *2 (20Hz to 20kHz) Rs=150Ω Input Gain=Max Input Pad=0dB Input Sensitivity=-60dB	-128dBu Equivalent Input Noise -86dBu residual output noise, OMNI OUT STEREO OUT off -86dBu(90dB S/N), OMNI OUT STEREO fader at nominal level and all CH INPUT faders -64dBu(68dB S/N), OMNI OUT Master fader at nominal level and one CH INPUT fader
Crosstalk(@1kHz) Input GAIN=min	80dB adjacent input channels (CH1-16) 80dB input to output
Power requirements	Japan: AC100V 50/60Hz, 135W North America: AC120V, 60Hz, 135W Other Areas: AC220-240V, 50/60Hz, 135W
Dimensions (W x H x D)	DM1000: 436 x 200 x 585 mm (17.2" x 7.9" x 23.0") With MB and SP: 486 x 295 x 635 mm (19.1" x 11.6" x 25.0")
Weight	DM1000: 20.0 kg (44.1 lbs) With MB and SP: 23.6 kg (52.0 lbs)

*1 Total Harmonic Distortion are measured with a 6dB/octave filter @80kHz
*2 Hum&Noise are measured with 6dB/octave filter @12.7kHz;equivalent to a 20kHz filter with infinite dB/octave attenuation.

ANALOG INPUT SPECIFICATIONS

Input Terminal	Pad	Gain	Actual Load Impedance	For Use With Nominal	Input Level			Connector
					Sensitivity *1	Nominal	Max. before Clip	
CH INPUT 1-16	0	-60dB	3kΩ	50-600ohm Mics & 600ohm Lines	-70dBu	-60dBu	-40dBu	XLR3-31 type *2 (Balanced)
	0	-16dB			-26dBu	-16dBu	+4dBu	
	20	-16dB			-6dBu	+4dBu	+24dBu	
OMNI IN 1-4			10kΩ	600ohm Lines	+4dBu	+4dBu	+24dBu	

* 0dBu=0.775 Vrms.
* 0dBV=1.00 Vrms.
* +48V DC(phantom power) is supplied to CH INPUT(1-24) XLR type connector via each individual switch.
*1 Sensitivity is the lowest level that will produce an output of +4 dB (1.23 V) or the nominal output level when the unit is set to maximum gain. (All faders and level controls are maximum position.)
*2 XLR-3-31 type connectors are balanced (1=GND, 2=HOT, 3=COLD).
- In these specifications, 0 dBu = 0.775 Vrms.
- All input AD converters (INPUT 1~16, OMNI INPUT 1~4, TALKBACK) are 24-bit linear, 128-times oversampling. (@fs=44.1, 48 kHz)
- +48 V DC (phantom power) is supplied to CH INPUT (1~16) XLR type connectors via individual switches.

ANALOG OUTPUT SPECIFICATIONS

Output Terminal	Actual Source Impedance	For Use With Nominal	Gain SW	Output Level		Connector
				Nominal	Max. before Clip	
OMNI OUT 1-12	150Ω	600kΩ Lines	-	+4dBu	+24dBu	XLR3-32 type *1 (Balanced)
PHONES	100Ω	8Ω Lines	-	4mW	25mW	ST Phone jack *2 (Unbalanced)
		40Ω Lines	-	12mW	75mW	

* 0dBu=0.775 Vrms.
* 0dBV=1.00 Vrms.
*1 XLR-3-32 type connectors are balanced (1=GND, 2=HOT, 3=COLD).
*2 PHONES stereo phone jack is unbalanced (Tip=LEFT, Ring=RIGHT, Sleeve=GND).
- In these specifications, 0 dBu = 0.775 Vrms, 0 dBV=1.00 Vrms.
- All output DA converters (OMINI OUT 1~12, PHONES) are 24-bit, 128-times oversampling. (@fs=44.1, 48 kHz)

DIGITAL INPUT SPECIFICATIONS

Terminal		Format	Data Length	Level	Connector
2TR IN DIGITAL	1	AES/EBU	24bit	RS422	XLR3-31 type *1 (Balanced)
	2	IEC-60958	24bit	0.5Vpp/75Ω	RCA pin jack

*1 XLR-3-31 type connectors are balanced (1=GND, 2=HOT, 3=COLD).

DIGITAL OUTPUT SPECIFICATIONS

Terminal		Format	Data Length	Level	Connector
2TR OUT DIGITAL	1 *1	AES/EBU (Professional use)	24bit	RS422	XLR3-32 type *4 (Balanced)
	2 *2	IEC-60958 (Consumer Use)	24bit *3	0.5Vpp/75Ω	RCA pin jack

* 0dBu=0.775Vrms
* 0dBV=1.00Vrms
*1 channel status of 2TR OUT DIGITAL 1...type: linear PCM, emphasis: NO, sampling rate: depends on the internal configuration.
*2 channel status of 2TR OUT DIGITAL 2...type: linear PCM, category code: Digital signal mixer, copy prohibit: NO, emphasis: NO, clock accuracy: Level II (1000 ppm), sampling rate: depends on the internal configuration.
*3 dither: word length 16/20/24 bit
*4 XLR-3-32 type connectors are balanced. (1=GND,2=HOT,3=COLD)

CONTROL I/O SPECIFICATIONS

I/O Port		Format	Level	Connector in Console
TO HOST USB		USB	0V ~ 3.3V	B type USB Connector
MIDI	IN *1	MIDI	-	DIN Connector 5P
	OUT			
TIME CODE INPUT		SMPTE	Nominal - 10dB/10kΩ	XLR-3-31 type (Balanced) *2
WORD CLOCK	IN	-	TTL/75Ω	BNC Connector
	OUT			
CONTROL			C-MOS IN, Open Collector OUT 1pin: 150mA, 8pin total: 500mA	D-SUB Connector 25P (Female)
REMOTE			RS422	D-SUB Connector 9P (Male)
METER				D-SUB Connector 15P (Female)

*1 MIDI IN can use as TIME CODE IN MTC.
*2 XLR-3-31 type connectors are balanced. (1=GND, 2=HOT, 3=COLD).

02R96VCM Specifications

GENERAL SPECIFICATIONS

Internal processing	32bit (Accumulator=58bit)
Number of scene memories	99
Sampling frequency rate	Internal: 44.1kHz,48kHz,88.2kHz,96kHz External: Normal rate 44.1kHz(-10%) to 48kHz(+6%) Double rate 88.2kHz(-10%) to 96kHz(+6%)
Signal Delay	≤ 2.0 ms CH INPUT to STEREO OUT (fs=48 kHz) ≤ 1.1 ms CH INPUT to STEREO OUT (fs=96 kHz)
Total harmonic distortion *1 Input Gain=Min.	≤ 0.05%, 20Hz to 20 kHz @+14dBu into 600Ω ≤ 0.01%, 1kHz @+18dBu into 600Ω CH INPUT to STEREO OUT (@Sampling frequency = 96 kHz)
Frequency response	0.5, -1.5dB, 20Hz to 20 kHz @+4dBu into 600Ω (@Sampling frequency = 48 kHz) 0.5, -1.5dB, 20Hz to 40 kHz @+4dBu into 600Ω (@Sampling frequency = 96 kHz)
Dynamic range (maximum level to noise level)	110dB typ. DA Converter (STEREO OUT) 105dB typ. AD+DA (to STEREO OUT)
Hum & noise level *2 (20Hz to 20kHz) Rs=150Ω Input Gain=Max Input Pad=0dB Input Sensitivity=-60dB	-128dBu Equivalent Input Noise -92dBu residual output noise. STEREO OUT STEREO OUT off -92dBu(96dB S/N) STEREO OUT STEREO fader at nominal level and all CH INPUT faders at minimum level -64dBu(68dB S/N) STEREO OUT STEREO fader at nominal level and one CH INPUT fader at nominal level
Crosstalk(@1kHz) Input GAIN=min	80dB adjacent input channels (CH1 to 24) 80dB input to output
Power requirements	Japan: AC100V 50/60Hz, 200W North America: AC120V, 60Hz, 200W Other Areas: AC220-240V, 50/60Hz, 200W
Dimensions (W x H x D)	02R96: 667 x 239 x 697 mm (26.3" x 9.4" x 27.4") With MB and SP: 700 x 352 x 762 mm (27.6" x 13.9" x 30.0")
Weight	02R96: 34.0 kg (75 lbs) With MB and SP: 39.4 kg (86.9 lbs)

*1 Total Harmonic Distortion is measured with a 6dB/octave filter @80kHz
*2 Hum & Noise are measured with 6dB/octave filter @12.7 kHz ; equivalent to a 20 kHz filter with infinite dB/octave attenuation.

CONTROL I/O SPECIFICATIONS

I/O Port		Format	Level	Connector in Console
TO HOST	Serial	-	RS422	Mini DIN Connector 8P
	USB	USB 1.1	0V ~ 3.3V	B type USB Connector
MIDI	IN	MIDI	-	DIN Connector 5P
	OUT	MIDI	-	DIN Connector 5P
	THRU	MIDI	-	DIN Connector 5P
TIME CODE IN	MTC	MIDI	-	DIN Connector 5P
	SMPTE	SMPTE	Nominal - 10dB/10kΩ	XLR-3-31 type (Balanced)*1
WORD CLOCK	IN	-	TTL/75Ω (ON/OFF)*2	BNC Connector
	OUT	-	TTL/75Ω	BNC Connector
CONTROL		-	-	D-SUB Connector 25P (Female)
METER		-	RS422	D-SUB Connector 15P (Female)

*1 XLR-3-31 type connectors are balanced. (1=GND, 2=HOT, 3=COLD).
*2 This switch is on the rear panel.

ANALOG INPUT SPECIFICATIONS

Input Terminal	Pad	Gain	Actual Load Impedance	For Use With Nominal	Input Level: Sensitivity *1	Input Level: Nominal	Input Level: Max. before Clip	Connector
CH INPUT A/B 1-16	0	-60dB	3kΩ	50-600Ω Mics & 600Ω Lines	-70dBu	-60dBu	-46dBu	A:XLR3-31 type (Balanced) *2 B:Phone jack (TRS)(Balanced) *3
		-16dB			-26dBu	-16dBu	-2dBu	
	26				0dBu	+10dBu	+24dBu	
CH INPUT 17-24		-34dB	4kΩ	600Ω Lines	-44dBu	-34dBu	-20dBu	Phone jack (TRS)(Balanced) *3
		+10dB			0dBu	+10dBu	+24dBu	
CH INSERT IN 1-16			10kΩ	600Ω Lines	-6dBu	+4dBu	+18dBu	Phone jack (Unbalanced) *4
2TR IN ANALOG1(L,R)			10kΩ	600Ω Lines	+4dBu	+4dBu	+18dBu	Phone jack (TRS)(Balanced) *3
2TR IN ANALOG2(L,R)			10kΩ	600Ω Lines	-10dBV	-10dBV	+4dBV	RCA pin jack (Unbalanced)

* 0dBu=0.775 Vrms.
* 0dBV=1.00 Vrms.
* +48V DC(phantom power) is supplied to CH INPUT(1-24) XLR type connector via each individual switch.
*1 Sensitivity is the lowest level that will produce an output of +4 dB (1.23 V) or the nominal output level when the unit is set to maximum gain. (All faders and level controls are maximum position.)
*2 XLR-3-31 type connectors are balanced (1=GND, 2=HOT, 3=COLD).
*3 Phone jacks are balanced (Tip=HOT, Ring=COLD, Sleeve=GND).
*4 Phone jacks are wired: Tip=OUT, Ring=IN, Sleeve=GND
- In these specifications, 0 dBu = 0.775 Vrms, 0 dBV=1.00 Vrms.
- All input AD converters (except INSERT I/O 1~16) are 24-bit linear, 128-times oversampling.
- +48 V DC (phantom power) is supplied to CH INPUT (1~16) XLR type connectors via individual switches.

ANALOG OUTPUT SPECIFICATIONS

Output Terminal	Actual Source Impedance	For Use With Nominal	Gain SW *	Output Level: Nominal	Output Level: Max. before Clip	Connector
STEREO OUT(L,R)	600Ω	10kΩ Lines	-	-10dBV	+4dBV	RCA pin jack (Unbalanced)
	150Ω	600kΩ Lines	-	+4dBu	+18dBu	XLR3-32 type (Balanced) *5
STUDIO MONITOR OUT(L,R)	150Ω	10kΩ Lines	-	+4dBu	+18dBu	Phone jack (TRS)(Balanced) *3
C-R MONITOR OUT (L,R)	150Ω	600kΩ Lines	-	+4dBu	+18dBu	Phone jack (TRS)(Balanced) *3
OMNI OUT 1-8	150Ω	10kΩ Lines	+18dBu (default)	+4dBu	+18dBu	Phone jack (TRS)(Balanced) *3
			+4dBV	-10dBV	+4dBV	
INSERT OUT 1-16	600Ω	10kΩ Lines	-	+4dBu	+18dBu	Phone jack (Unbalanced) *4
PHONES	100Ω	8Ω Lines	-	4mW	25mW	Stero phone jack (TRS)(Unbalanced) *6
		40Ω Lines	-	12mW	75mW	

* +18dBu, +4dBV selectable (Internal SW)
0dBV=1.00 Vrms.
0dBu=0.775 Vrms.
*1 The maximum output level of each OMNI OUT can be set internally.
*2 XLR-3-32 type connectors are balanced (1=GND, 2=HOT, 3=COLD).
*3 Phone jacks are balanced (Tip=HOT, Ring=COLD, Sleeve=GND).
*4 Phone jacks are wired: Tip=OUT, Ring=IN, Sleeve=GND
*5 PHONES stereo phone jack is unbalanced (Tip=LEFT, Ring=RIGHT, Sleeve=GND).
- In these specifications, 0 dBu = 0.775 Vrms, 0 dBV=1.00 Vrms.
- All output DA converters (except INSERT OUT 1_16) are 24-bit, 128-times oversampling.

DIGITAL INPUT / OUTPUT SPECIFICATIONS

Terminal		Format	Data Length	Level	Connector
2TR IN DIGITAL	1	AES/EBU	24bit	RS422	XLR3-31 type (Balanced) *1
	2	IEC-60958	24bit	0.5Vpp/75Ω	RCA pin jack
	3	IEC-60958	24bit	0.5Vpp/75Ω	RCA pin jack
CASCADE IN		-	-	RS422	D-sub Half Pitch Connector 68P (female)

*1 XLR-3-31 type connectors are balanced. (1=GND, 2=HOT, 3=COLD)

DIGITAL INPUT / OUTPUT SPECIFICATIONS

Terminal		Format	Data Length	Level	Connector
2TR OUT DIGITAL	1	AES/EBU *1 (Professional use)	24bit *3	RS422	XLR3-32 type (Balanced) *4
	2	IEC-60958 *2 (Consumer Use)	24bit *3	0.5Vpp/75Ω	RCA pin jack
	3	IEC-60958 *2 (Consumer Use)	24bit *3	0.5Vpp/75Ω	RCA pin jack
CASCADE OUT		-	-	RS422	D-sub Half Pitch Connector 68P (female)

*1 channel status of 2TR OUT DIGITAL 1...type: 2 audio channels, emphasis: NO, sampling rate: depends on the internal configuration
*2 channel status of DIGITAL OUT 2, 3...type: 2 audio channels, category code: 2 channel PCM encoder/ decoder, copy prohibit: NO, emphasis: NO, clock accuracy: Level II (1000 ppm), sampling rate: depends on the internal configuration
*3 dither: word length 16 - 24 bit
*4 XLR-3-32 type connectors are balanced. (1=GND, 2=HOT, 3=COLD)

01V96VCM Specifications

GENERAL SPECIFICATIONS

Internal processing	32bit (Accumulator 58bit)
Number of scene memories	99
Sampling frequency	Internal: 44.1kHz,48kHz,88.2kHz,96kHz External: Normal rate: 44.1kHz-10% - 48kHz+6% Double rate: 88.2kHz-10% - 96kHz+6%
Signal delay	≤ 1.6ms CH INPUT to STEREO OUT (@Sampling frequency = 48kHz) ≤ 0.8ms CH INPUT to STEREO OUT (@Sampling frequency = 96kHz)
Total harmonic distortion *1 Input Gain=Min.	CH INPUT to STEREO OUT ≤ 0.05%, 20Hz to 20kHz @+14dBu into 600Ω ≤ 0.01%, 1kHz @+24dBu into 600Ω (@Sampling frequency = 48kHz) ≤ 0.05%, 20Hz to 40kHz @+14dBu into 600Ω ≤ 0.01%, 1kHz @+24dBu into 600Ω (@Sampling frequency = 96kHz)
Frequency response	CH INPUT to STEREO OUT 0.5, -1.5dB, 20Hz - 20kHz @+4dBu into 600Ω (@Sampling frequency = 48kHz) 0.5, -1.5dB, 20Hz - 40kHz @+4dBu into 600Ω (@Sampling frequency = 96kHz)
Dynamic range (maximum level to noise level)	110dB typ. DA Converter (STEREO OUT) 106dB typ. AD+DA (to STEREO OUT) @fs=48kHz 106dB typ. AD+DA (to STEREO OUT) @fs=96kHz
Hum & noise level *2 (20Hz to 20kHz) Rs=150Ω Input Gain=Max Input Pad=0dB Input Sensitivity=-60dB	-128dBu Equivalent Input Noise. -86dBu residual output noise. STEREO OUT STEREO OUT off. -86dBu (90dB S/N) STEREO OUT STEREO fader at nominal level and all CH INPUT faders at minimum level. -64dBu (68dB S/N) STEREO OUT STEREO fader at nominal level and one CH INPUT fader at nominal level.
Crosstalk(@1kHz) Input GAIN=min	80dB adjacent input channels (CH1-12) 80dB adjacent input channels (CH13-16) 80dB input to output
Power requirements	Japan: AC100V 50/60Hz, 90W North America: AC120V, 60Hz, 90W Other Areas: AC220-240V, 50/60Hz, 90W
Dimensions (W x H x D)	436 x 150 x 540 mm (17.2" x 5.9" x 21.3")
Weight	15.0 kg (33.1 lbs.)

*1 Total Harmonic Distortion is measured with a 6dB/octave filter @80kHz.
*2 Hum & Noise are measured with a 6dB/ octave filter @12.7kHz; equivalent to a 20kHz filter with infinite dB/octave attenuation.

ANALOG INPUT SPECIFICATIONS

Input Terminal	Pad	Gain	Actual Load Impedance	For Use With Nominal	Input Level: Sensitivity *1	Input Level: Nominal	Input Level: Max. before Clip	Connector
CH INPUT 1 to 12	0	-60dB	3kΩ	50-600Ω Mics & 600Ω Lines	-70dBu	-60dBu	-40dBu	A:XLR-3-31 type (Balanced) *2 B:Phone jack (TRS)(Balanced) *3
		-16dB			-26dBu	-16dBu	+4dBu	
	20				-6dBu	+4dBu	+24dBu	
CH INPUT 13 to 16		-26dB	10kΩ	600Ω Lines	-36dBu	-26dBu	-6dBu	Phone jack (TRS)(Balanced) *3
		+4dB			-6dBV	+4dBu	+24dBu	
CH INSERT IN 1 to 12			10kΩ	600Ω Lines	-6dBu	+4dBu	+24dBu	Phone jack (TRS) (Unbalanced) *4
2TR IN [L,R]			10kΩ	600Ω Lines	-10dBV	-10dBV	+10dBV	RCA pin jack (Unbalanced)

* 0dBu=0.775 Vrms.
* 0dBV=1.00 Vrms.
* +48V DC(phantom power) is supplied to CH INPUT(1-24) XLR type connector via each individual switch.
*1 Sensitivity is the lowest level that will produce an output of +4 dB (1.23 V) or the nominal output level when the unit is set to maximum gain. (All faders and level controls are maximum position.)
*2 XLR-3-31 type connectors are balanced (1=GND, 2=HOT, 3=COLD).
*3 Phone jacks are balanced (Tip=HOT, Ring=COLD, Sleeve=GND).
*4 CH INSERT IN/OUT phone jacks are unbalanced. (Tip=OUTPUT, Ring=INPUT, Sleeve=GND).
- In these specifications, 0 dBu = 0.775 Vrms, 0 dBV=1.00 Vrms.
- All input AD converters (CH INPUT 1~16) are 24-bit linear, 128-times oversampling. (@fs=44.1, 48 kHz)
- +48 V DC (phantom power) is supplied to CH INPUT (1~12) XLR type connectors via individual switches.
- Three PHANTOM +48V switches CH1~4, 5~8, 9~12 turn on the phantom power for inputs 1~4, 5~8, 9~12 respectively

ANALOG OUTPUT SPECIFICATIONS

Output Terminal	Actual Source Impedance	For Use With Nominal	Output Level: Nominal	Output Level: Max. before Clip	Connector
STEREO OUT (L,R)	150Ω	600Ω Lines	+4dBu	+24dBu	XLR-3-32 type *1 (Balanced)
OMNI OUT 1 to 4	150Ω	10kΩ Lines	+4dBu	+24dBu	Phone jack (TRS)(Balanced) *2
MONITOR OUT (L,R)	150Ω	10kΩ Lines	+4dBu	+24dBu	Phone jack (TRS)(Balanced) *2
CH INSERT OUT 1 to 12	600Ω	10kΩ Lines	+4dBu	+24dBu	Phone jack (TRS) (Unbalanced) *3
2TR OUT (L,R)	10KΩ	600Ω Lines	-10dBV	+10dBV	RCA pin jack (Unbalanced)
PHONES	100Ω	8Ω Lines	4mW	25mW	Stero phone jack (TRS)(Unbalanced) *4
		40Ω Lines	12mW	75mW	

* 0dBu=0.775 Vrms.
* 0dBV=1.00 Vrms.
*1 XLR-3-32 type connectors are balanced (1=GND, 2=HOT, 3=COLD).
*2 Phone jacks are balanced (Tip=HOT, Ring=COLD, Sleeve=GND)
*3 CH INSERT IN/OUT phone jacks are unbalanced. (Tip=OUTPUT, Ring=INPUT, Sleeve=GND).
*4 PHONES stereo phone jack is unbalanced (Tip=LEFT, Ring=RIGHT, Sleeve=GND).
- In these specifications, 0 dBu = 0.775 Vrms, 0 dBV=1.00 Vrms.
- All output DA converters are 24-bit, 128-times oversampling. (@fs=44.1, 48 kHz)

DIGITAL INPUT SPECIFICATIONS

Terminal	Format	Data Length	Level	Connector
2TR IN DIGITAL	IEC-60958	24bit	0.5Vpp/75Ω	RCA pin jack
ADAT IN	ADAT *1	24bit	-	OPTICAL

*1 ALESIS Proprietary Multichnnel Optical Digital Interface Format.

DIGITAL OUTPUT SPECIFICATIONS

Terminal	Format	Data Length	Level	Connector
2TR OUT DIGITAL	IEC-60958 Consumer use	24bit *2	0.5Vpp/75Ω	RCA pin jack
ADAT OUT	ADAT *3	24bit	-	OPTICAL

*1 channel status of 2TR OUT DIGITAL type: linear PCM, category code : Digital signal mixer, copy prohibit: NO, emphasis: NO, clock accuracy: Level II (1000 ppm), sampling rate: depends on the internal configuration.
*2 dither: word length 16/20/24 bit.
*3 ALESIS Proprietary Multichnnel Optical Digital Interface Format.

CONTROL I/O SPECIFICATIONS

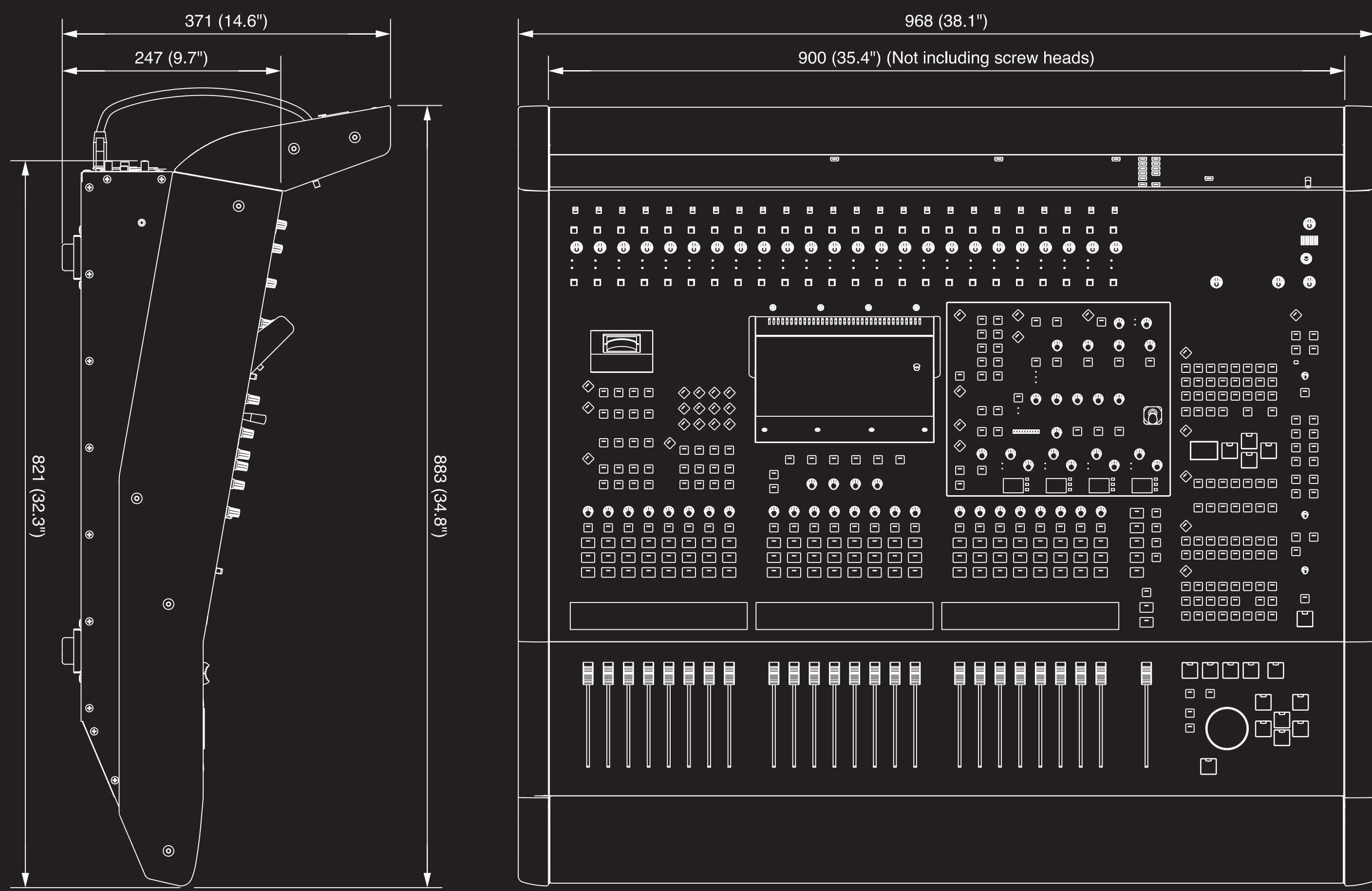
I/O Port		Format	Level	Connector in Console
TO HOST USB		USB	0V - 3.3V	B type USB connector
MIDI	IN *1	MIDI	-	DIN Connector 5P
	OUT	MIDI	-	DIN Connector 5P
	THRU	MIDI	-	DIN Connector 5P
WORD CLOCK	IN	-	TTL/75Ω	BNC Connector
	OUT	-	TTL/75Ω	BNC Connector

*1 MIDI IN can use as TIME CODE IN MTC.

Dimensions

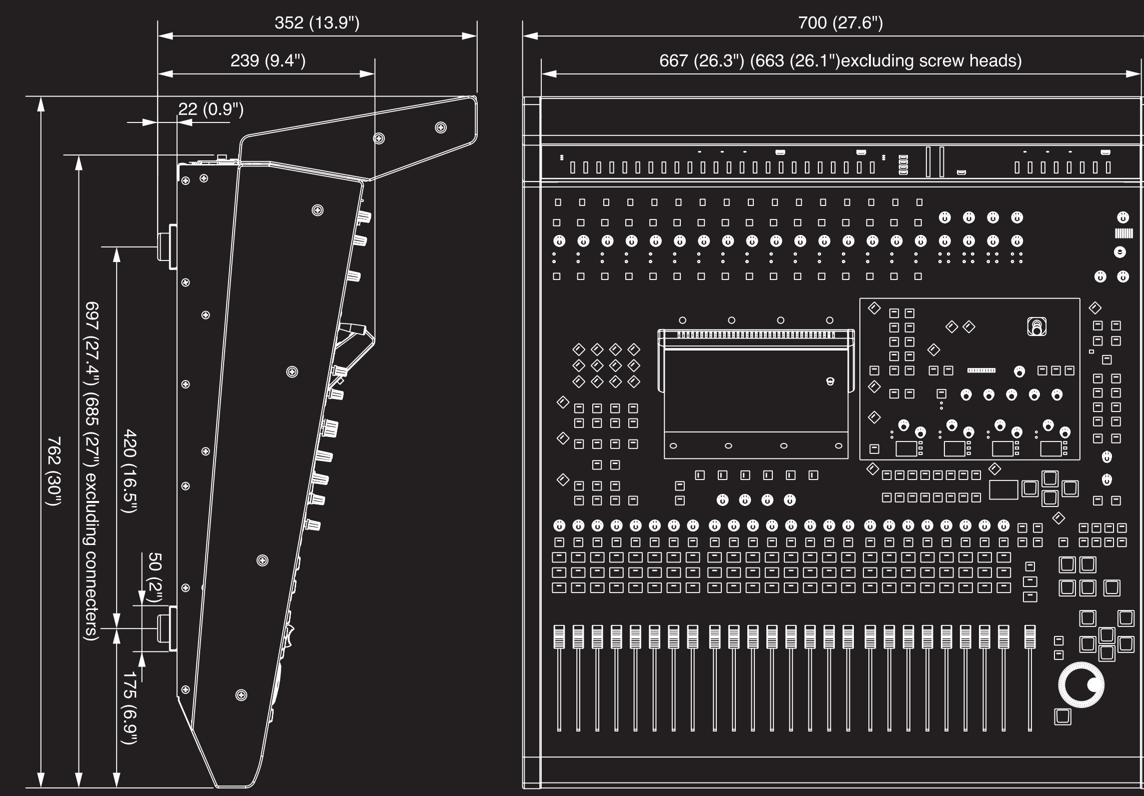
DM2000VCM

unit; mm (inch)



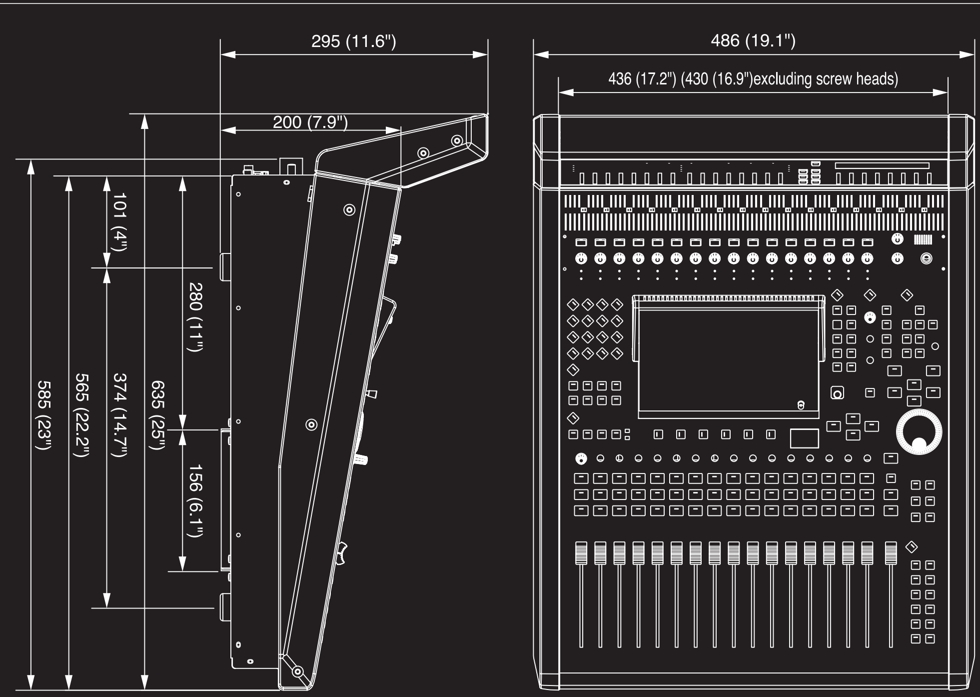
02R96VCM

unit; mm (inch)



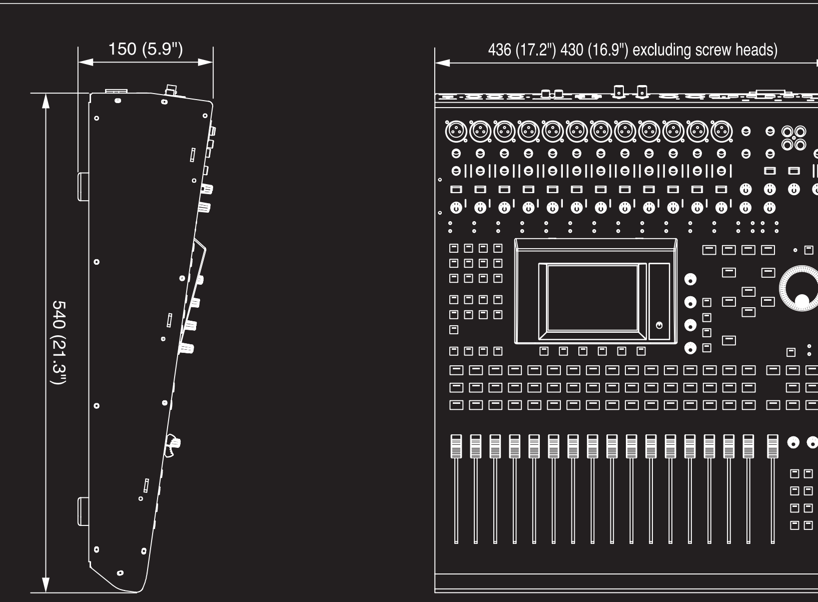
DM1000VCM

unit; mm (inch)



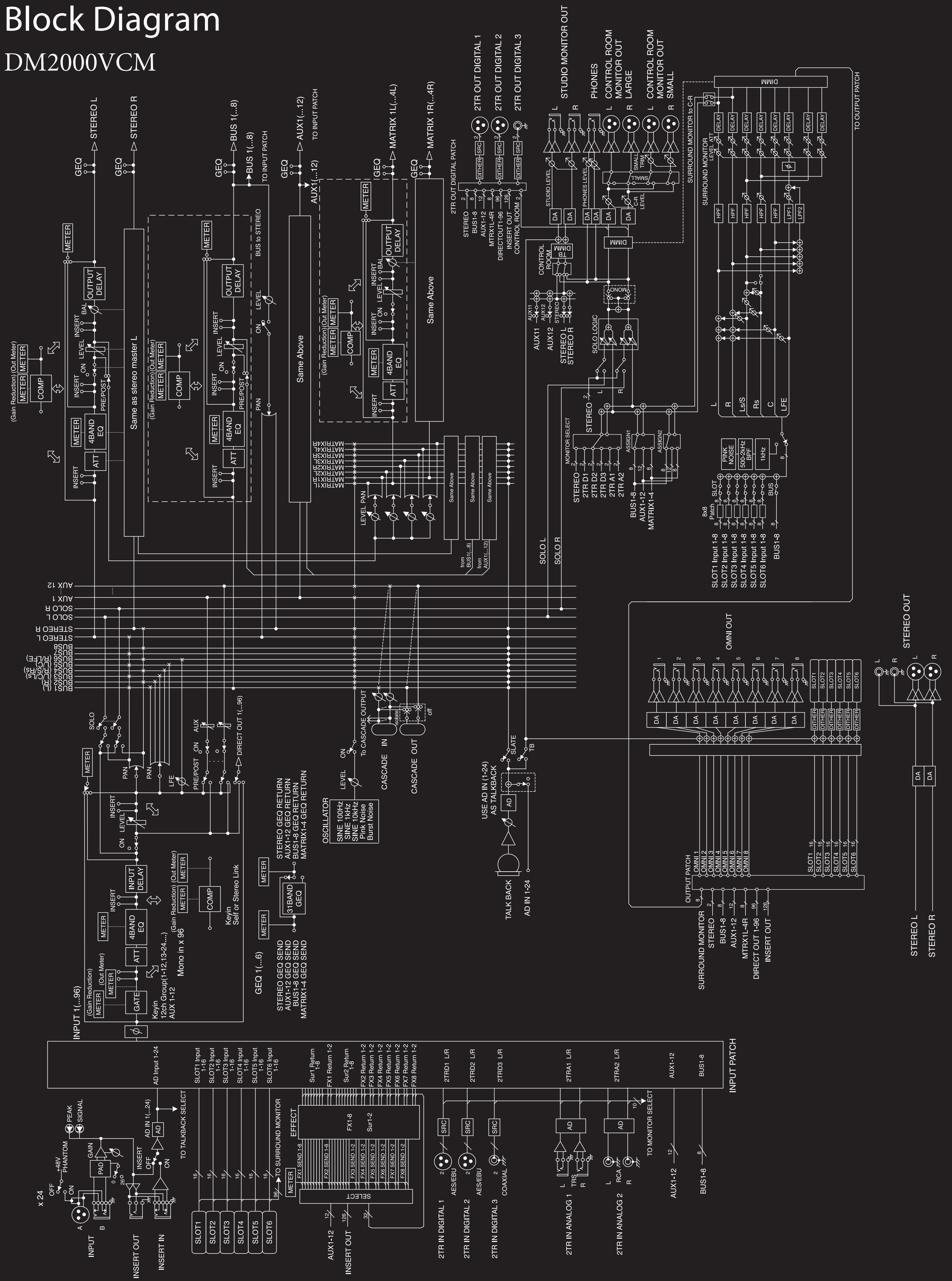
01V96VCM

unit; mm (inch)

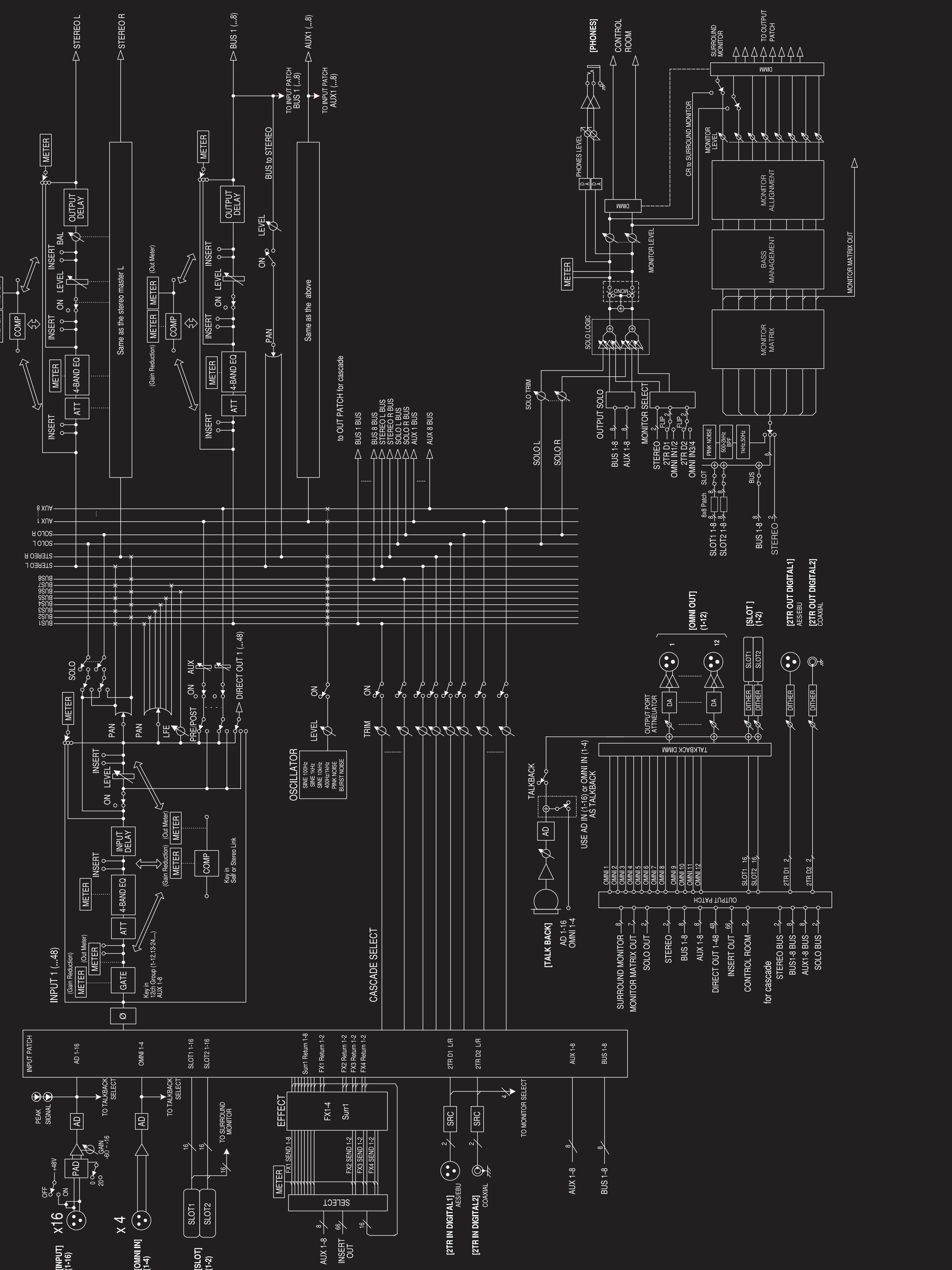


Block Diagram

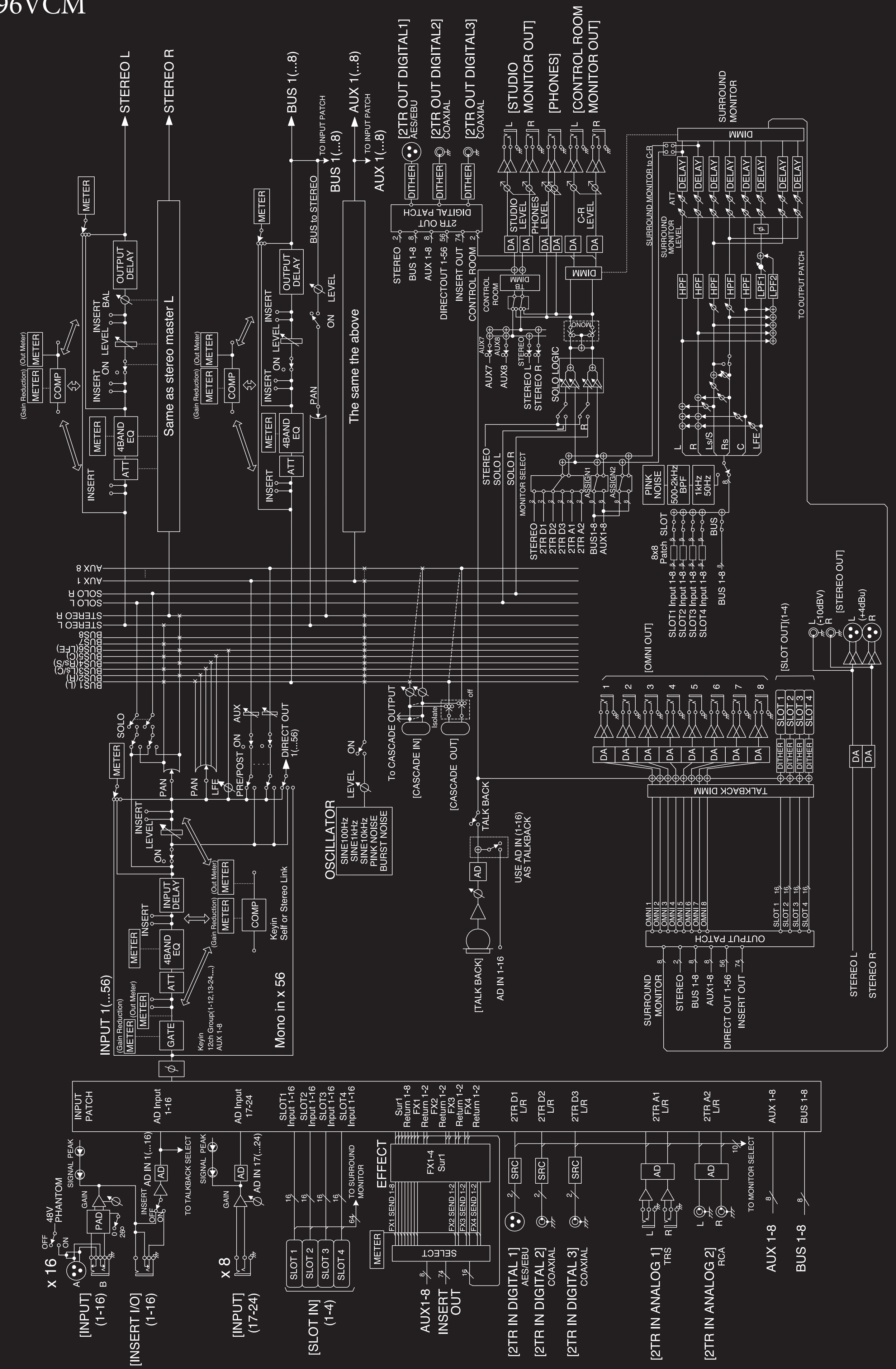
DM2000VCM



DM1000VCM



02R96VCM



01V96VCM