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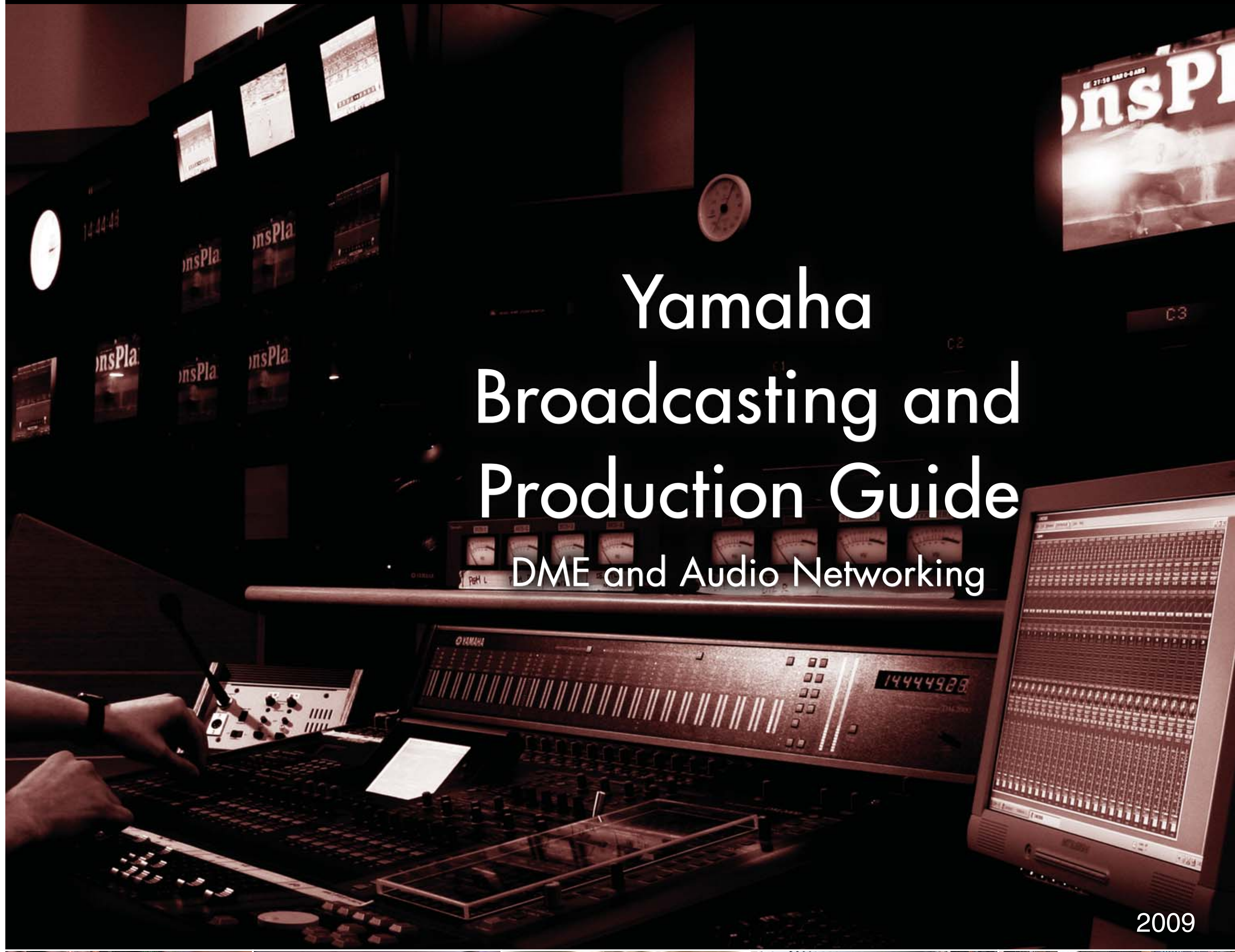
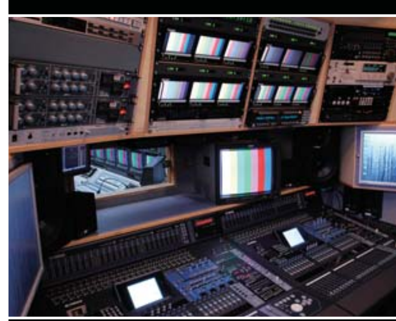
YAMAHA



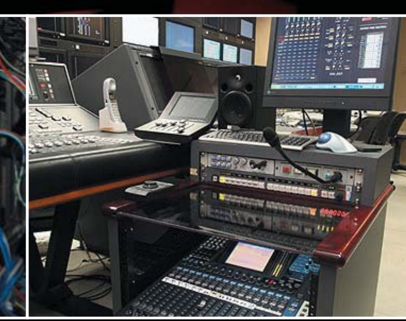
Yamaha Broadcasting and Production Guide

DME and Audio Networking

2009







DIGITAL MIXING ENGINE

DME64N / DME24N

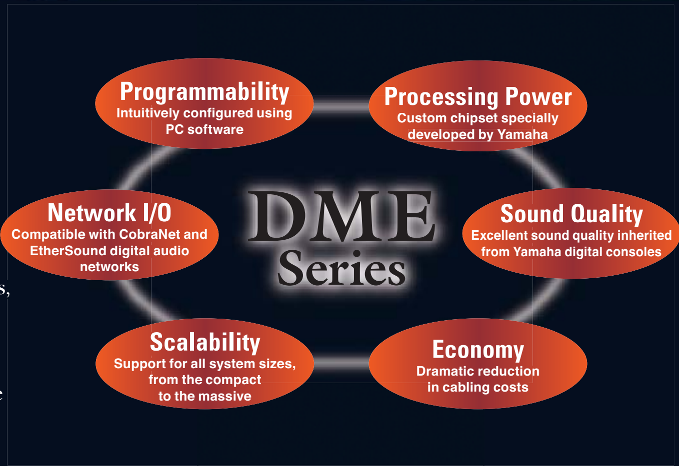
Totally reliable yet utterly flexible, in the fields of broadcasting and music production, Yamaha's DME Series is second to none.

While it goes without saying that mixing solutions for broadcasting stations and professional music-production studios must deliver excellent sound and performance, total reliability is the true mark of quality. The digital mixing engines of the Yamaha DME Series comfortably and efficiently satisfy this need. To achieve steadfast reliability, functional components are freely and flexibly assembled using a dedicated PC application in order to create systems where physical connections and wiring are reduced to the absolute minimum. And for even greater versatility, these digital mixing engines also boast impressive capabilities as digital transmission and networking devices. Recent developments in the field of digital audio transfer have led not only to the elimination of quality deterioration associated with long-distance transmission, but also to vastly enhanced levels of reliability through the integration of line redundancy and a diverse range of network topologies such as ring, star, and daisy-chain. The DME Series offers wide-ranging support for these innovative technologies. Within these pages, you will find examples of how this awe-inspiring series is being put to highly effective use in broadcasting stations and music production studios all over the world.

In the DME Series of digital mixing engines, Yamaha has realized a DME box that is simultaneously mighty and easy to use. The custom chipset lying at the heart of each mixing engine has been developed specifically for complex audio processing, and using a dedicated PC application, the awe-inspiring functionality that it provides can be freely configured to create a vast range of different sound systems. The ideal choice for any environment where audio is mixed, this series is renowned for its ability to deliver flexible solutions for applications of all shapes and sizes. Yet the DME Series offers even more: In broadcasting stations and production studios – where sound quality and processing power are crucial but reliability is paramount – the true potential of these digital mixing engines is fully realized. Systems can be freely and intuitively assembled from a vast array of different functional components to realize thoroughly flexible solutions for many different mixing environments. And thanks to stunning sound quality inherited from Yamaha digital consoles, which continue to deliver unrivalled performance in studio recording and live sound-reinforcement settings, a single unit can easily provide the functionality required for surround-sound and other high-level systems. Mini-YGDAI cards for audio input and output can be selected to suit many different system formats and channel-number requirements, and the DME Series also offers support for advanced digital-audio transmission solutions such as CobraNet, EtherSound, and MADI. In cases where distributed processing is desirable, furthermore, a solution is readily at hand in the form of the DME Satellite Series.

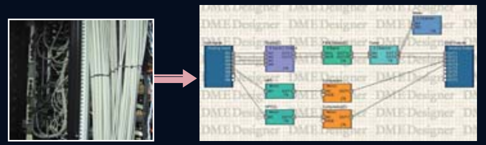
DME is ...

Integrating a chipset specifically developed for the processing of digital signals, the digital mixing engines of the Yamaha DME Series constitute DSP boxes with exceptional processing power. Using a dedicated PC application, systems can be intuitively and flexibly assembled from a myriad of components such as mixers, equalizers, and compressors in addition to the delays, limiters, de-essers, matrix mixers, routers, and other processors that are indispensable in the fields of broadcasting and music production. Components may also be combined as powerful functional blocks, which can then be arranged to create innovative solutions for downmixing, bass management, and other critical processes in surround sound systems. Of course, these configurations can be conveniently rearranged and customized whenever needed.



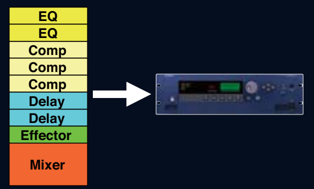
Advantage 1 Virtual wiring for steadfast reliability

Built within a PC application, DME sound systems require no actual wiring between functional components, and with physical connections and wiring therefore reduced to the absolute minimum, stunning levels of reliability are achieved. What's more, GPI devices and MIDI-based controllers can also be integrated whenever needed in order to realize the perfect working environment.



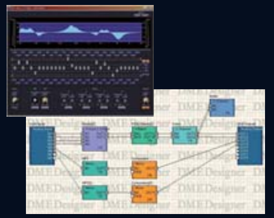
Advantage 2 Remarkable cost and space savings

With conventional mixing systems, a separate device is generally used for practically every different function. In stark contrast, however, the DME Series incorporates a dedicated PC application, DME Designer, to freely assemble a vast array of components contained within the hardware unit itself so that functions can be easily called up whenever needed. As such, this design is ideally matched to the field of broadcasting, where operations are often carried out in confined spaces or where equipment must be made available rapidly in response to changes in conditions.



Advantage 3 Irresistible flexibility and ease-of-use

DME Designer – the PC application forming an integral part of the DME Series – provides an extremely user-friendly environment in which configurations and systems can be intuitively setup using simple operations such as drag and drop. What's more, frequently-used parameters can be freely defined and assigned to external controllers. In this way, DME Designer makes it easy for anyone – regardless of their level of expertise – to construct highly practical and effective systems.



Broadcasting and Production Guide

— DME and Audio Networking —

Contents

Surround System

DME64N / DME24N

With digital broadcasting becoming more and more popular, ever-increasing use of surround sound formats is inevitable. What's more, continued growth of the use of DVDs, multichannel audio sources, and other products using these formats is also beyond doubt. It is no wonder then that systems capable of producing multichannel music are in extremely high demand. And while the high price of convention systems has been a constant obstacle, a remarkably flexible and reliable solution is now available in the DME64N and DME24N. Allowing an impressive array of audio processing components to be intuitively assembled in a user-friendly software environment, each of these mixing engines can be used to create indispensable tools for surround-sound music production. In addition, this impressive arsenal of mixing and processing functions has been neatly packaged into a compact hardware device taking up very little rack space.

DME Advantage

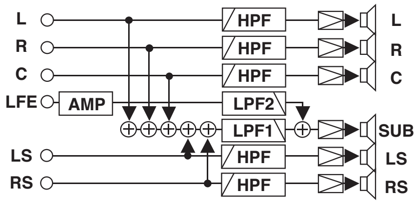
The true power of the DME Series is witnessed when freely configuring systems from a myriad of audio-processing components. Operating as a fully-customizable DSP box, either the DME64N or DME24N can create systems satisfying a wide variety of needs. Even 7.1-channel Dolby Digital EX can be easily supported by simply adding components.

Downmixing

The ability to freely configure a diverse range of audio-processing components offered by the DME Series is of immeasurable benefit when downmixing multiple audio channels for stereo playback. Even when time and budget are limited, a host of different mixes can be easily created, compared, and evaluated while still in the studio.

Bass Management

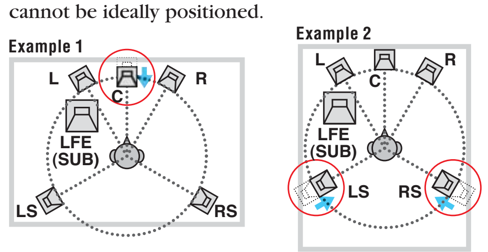
With the DME Series, bass management can be effortlessly achieved by combining a matrix mixer component with high-pass and low-pass filters. No additional hardware is needed. Moreover, groups of assembled components can be saved and recalled as component blocks, making it easy to monitor how mixes will sound on a variety of different bass-managed systems and to also compensate for poor low-end characteristics in mixing environments such as outside broadcasting vans and smaller studios.

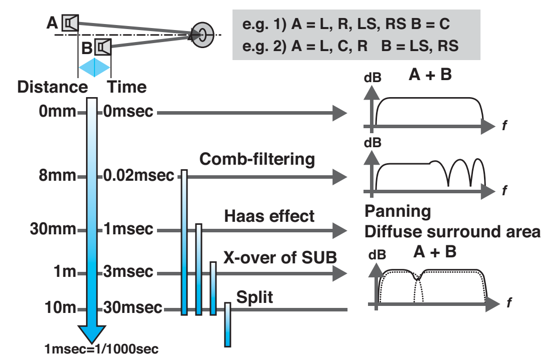


Bass management controller

Monitor Speaker Alignment

By providing each speaker with independent, finely-tuned delays and EQ, highly accurate monitoring environments can be conveniently setup, even when the speakers themselves cannot be ideally positioned.

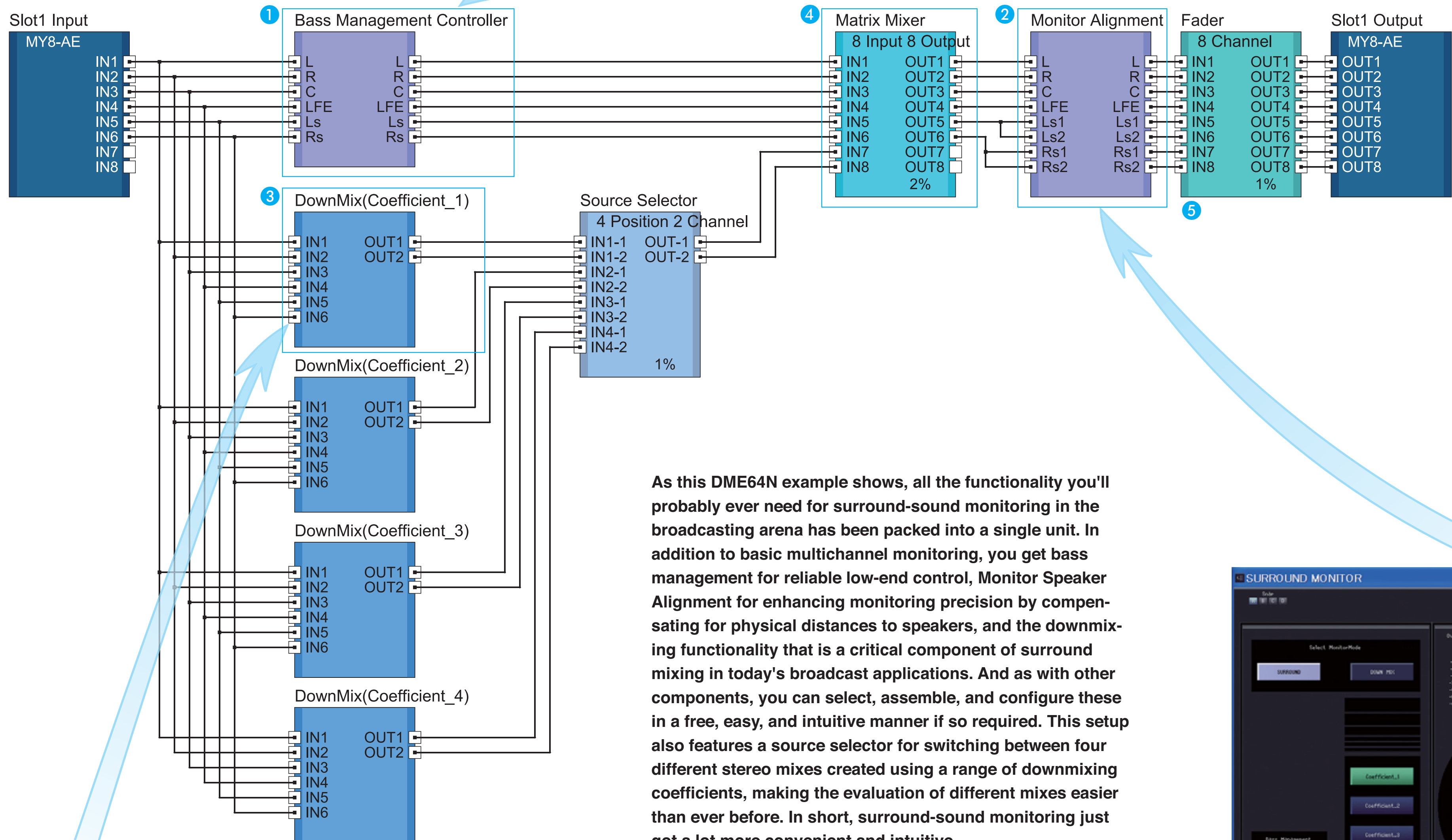




Monitoring errors caused by differences in monitoring distance

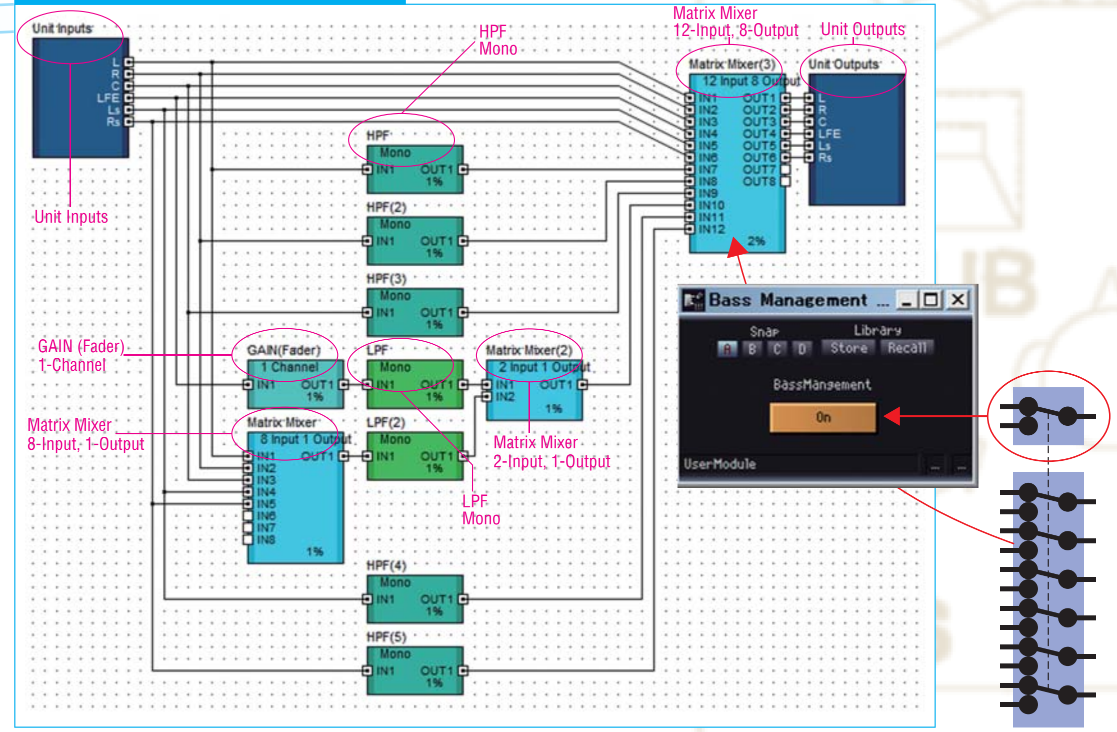
Surround System

DME64N / DME24N



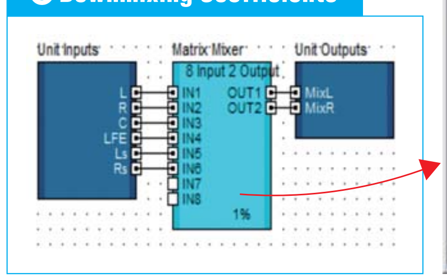
As this DME64N example shows, all the functionality you'll probably ever need for surround-sound monitoring in the broadcasting arena has been packed into a single unit. In addition to basic multichannel monitoring, you get bass management for reliable low-end control, Monitor Speaker Alignment for enhancing monitoring precision by compensating for physical distances to speakers, and the downmixing functionality that is a critical component of surround mixing in today's broadcast applications. And as with other components, you can select, assemble, and configure these in a free, easy, and intuitive manner if so required. This setup also features a source selector for switching between four different stereo mixes created using a range of downmixing coefficients, making the evaluation of different mixes easier than ever before. In short, surround-sound monitoring just got a lot more convenient and intuitive.

1 Bass Management Controller



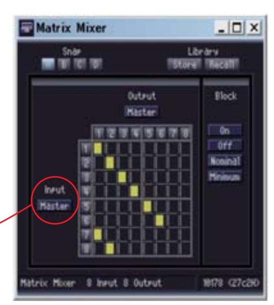
Not limited to preset configurations and components alone, the DME Series lets you create user-defined configurations by freely assembling components and saving them as user modules. For example, this original user module forms the core of the Bass Management Controller from the surround system on the left. Specifically, the audio-processing circuit splits up the six outputs of the 5.1 surround mix, extracts the low-frequency components of the five full-range speakers, and combines them with the LFE signal using a Matrix Mixer component. Individual and overall levels are adjusted and balanced within this mixer, and cross-over points are setup as required via the combination of high-pass and low-pass filters. Broadcast engineers know the importance of comparing the straight 5.1 surround sound and the bass-managed mix, and that's why another Matrix Mixer has been setup just in front of the output stage for highly convenient switching between the two at the click of a button. And thanks to the unprecedented flexibility offered by user modules, this same setup can even be modified to effortlessly handle 7.1 surround and other original specifications.

3 Downmixing Coefficients



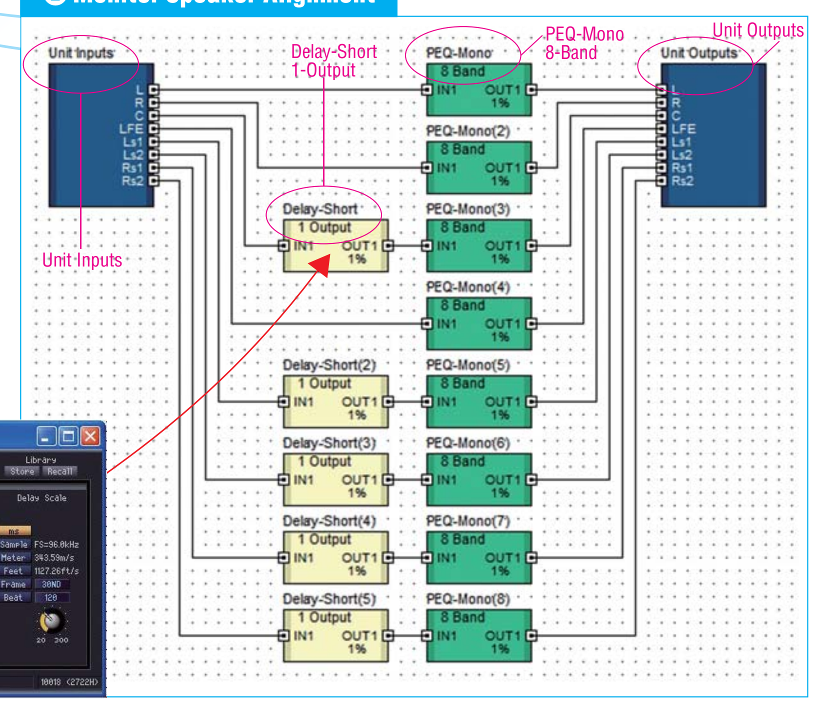
Broadcast engineers will really appreciate the downmixing capability offered by the DME Series. The above configuration, for example, is one of the sample system's four DownMix modules, each of which uses a different set of coefficients to instantly convert 5.1 surround sound to a stereo output. Having a range of different downmixing options readily available within a single unit is vastly more convenient than the conventional approach, where a single set of coefficients is adjusted for each individual application. Allowing different sets of coefficients to be evaluated quickly, conveniently, and objectively in order to select the best, Yamaha have made the task of downmixing considerably easier.

4 Matrix Mixer



The sample system's output-stage Matrix Mixer is fed by eight input signals – namely, the 5.1 surround signals processed by the Bass Management module and the stereo mix from the selected DownMix user module. The six surround channels can be grouped together using the Parameter Linking function for convenient mixing, while a single switch toggles instantaneously between the 5.1 mix on channels 1 to 6 and the stereo mix on 7 and 8.

2 Monitor Speaker Alignment



In the surround system example on the left, a wider monitoring area is needed, so a pair of back surround speakers is used on both the left and right for eight speakers in total. In an ideal world, surround-sound speakers would all be setup at the recommended angles and distances from the listener in order to facilitate highly accurate monitoring. Unfortunately, however, factors such as room shape often render such precise positioning impossible. Compensation for this using individual delays is known as "time alignment". Normally, the more speakers used, the greater the number of delay units required, and the more complex and expensive time alignment becomes. Not so with the Yamaha DME Series, where multiple delays can be easily setup and controlled within a single hardware device. Need precise time alignment for more monitors? Just add delays to the configuration and adjust.

In this surround system, time alignment is realized via the Monitor Speaker Alignment user module shown above. Of the monitoring environment's eight speakers, no alignment is required for the standard left and right or for the LFE, but highly-precise short delays are provided for each of the remaining five channels. Since fine tuning is so essential to multichannel monitoring, speaker positions can be virtually adjusted in units of one millimeter. Furthermore, another feature of advantage in surround-sound monitoring is the array of eight-band parametric EQs used to tune individual speaker outputs and to adjust for unique frequency characteristics in the monitoring environment.

Turn to page 49 for information on how to download and install DME Designer.

TV4
(Stockholm, Sweden)

Since its launch in 1990, the TV4 of Sweden has gone on to become the country's largest commercial TV channel. Recently, the channel's edit suites in Stockholm have undergone a massive multi-million kronor technical refit and – as further evidence that Yamaha Digital Mixing Engines (DMEs) are the perfect one-box solution for enhanced broadcast efficiency – six DME24Ns now form the heart of this cutting edge installation. Each DME unit is fitted with MY8-ADDA96 AD/DA cards, replacing a wide range of outboard studio devices, and in doing so, eliminating many of the problems experienced by broadcasters when using patch bays and equipment from a wide range of different manufacturers.

JMG Support AB of the Bromma borough of Stockholm installed the system. "We chose the DME solution because, as a Yamaha dealer, we were familiar with the DME range and the many different ways in which it can be controlled," says JMG Support's Johan Küller. "Of course we also knew that audio quality is excellent, so the customer would be very happy with the end result."

Each DME system at TV4 is setup to control two edit suites and a recording/speaker booth shared between them. In addition to all monitoring, the DME systems also handle all switching and level control of audio recording sources, which include a non-linear editing (NLE) system, videotape recorders (VTRs), and a microphone in the booth. The system also provides compression and limiting for the microphone.

Another input comes from telephone hybrids that TV4 uses to accept calls from viewers and to record or broadcast their messages. The DME systems allow the output from these units to be monitored in the booth and in both edit suites, as well as feeding it to the inputs of the NLEs and to a PC located in the booth.

"Our biggest challenge was to create a highly user-friendly interface," says Johan. "The client wanted one-button solutions for the most common scenarios and, as we looked at all the different combinations of inputs and outputs, it became clear how complex the switching could become. In order to simplify the switching procedures to the level desired by TV4, digital signal processing (DSP) was the only practical solution."

"Because of the client's requirement for simplicity of operation, we also realized that we would need to produce custom control panels. So the complexity of the system, plus the requirement for simple switching and custom panels meant that the Yamaha DME Series was by far the best solution."

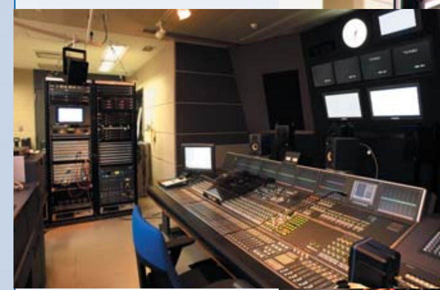
Custom control panels are located both in the booth and in each control room, which are connected to the serial ports of the DME24Ns. In the booth, the operator selects whether to record via microphone or telephone hybrid and whether to route the signal to be recorded within the booth or in one of the edit suites.

Within the edit suites, the operator has a control panel for selecting whether to monitor the output of the NLE system or the signal from the microphone or telephone hybrid. Additionally, he can select whether to record from the booth or from VTR and can also set the recording level.

"This was a challenging installation, particularly because the system needed to be very adaptable, with complex switching functions controlled in a very straightforward, intuitive way," says Johan. "The DME solution has provided TV4 with an easy-to-use, high fidelity network offering vastly increased audio fidelity."

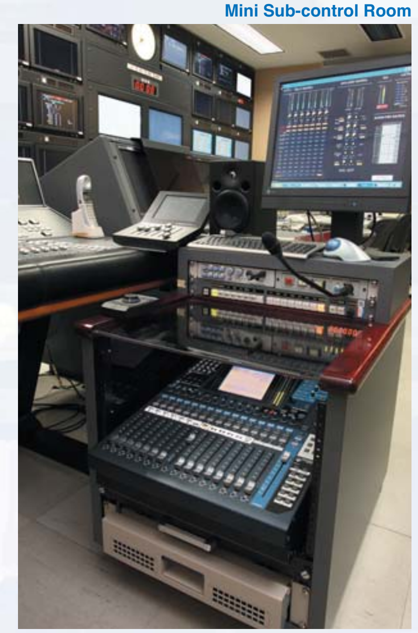
"But the best thing is the flexibility of the system. Using the DME software, we can easily change things that would – in a non-DME installation – have required physical re-wiring, taking much longer and, of course, causing TV4 plenty of headaches. Now, the potential for such problems simply doesn't exist."

Mainichi Broadcasting System, Inc.
(Osaka, Japan)



Galaxy Sub-control Room

Mini Sub-control Room

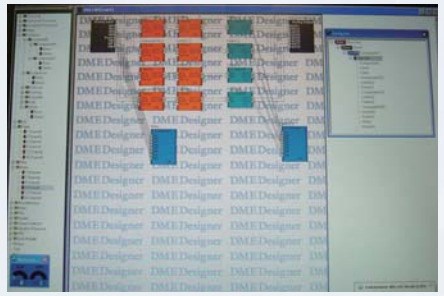


Mainichi Broadcasting System (MBS) uses a DME64N for both its Chichin Puipui show (a four-hour live broadcast) and Galaxy Sub program. Two other DME64Ns are employed as "mini subs" for news and sports programs – one being used for staff and commentator monitoring, while the other has been setup with surround-monitoring configurations for use as a monitoring processor. And setup in an outside-broadcasting van, a further DME64N functions as a multi-purpose processor for golf and auto-racing broadcasts.

Case Study #3

Processor

TeleModena
(Modena, Italy)

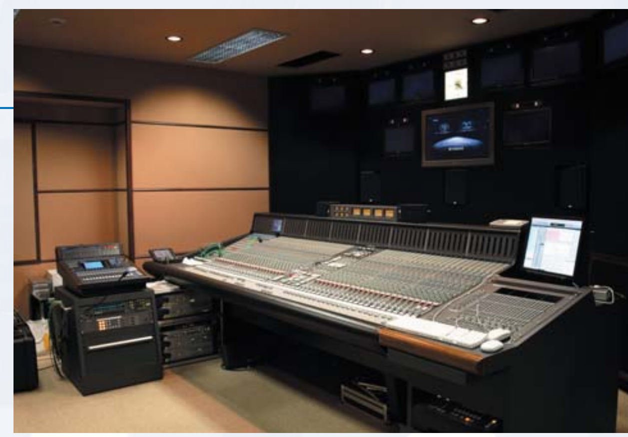


Comprising four shopping and local news stations, Telemodena is a small commercial TV network based in the city of Modena in northern Italy. At this channel, audio signals from the four control rooms are routed to a DME24N for dynamic control, limiting, and leveling before the high-frequency stage. Furthermore, technicians can conveniently monitor the state of signal processing using a PC constantly synchronized to the DME unit.

Case Study #4

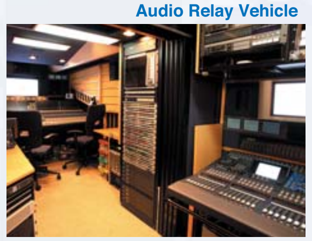
Surround & Multi-purpose Processor

Yomiuri Telecasting Corp.
(Osaka, Japan)



1 Sub

Audio Relay Vehicle



At Yomiuri Telecasting, a DME64N is used for surround monitoring at Sub-Control Room 1.
In addition, this unit also serves as a monitoring processor in an outside-broadcasting van, which incidentally, was fitted with a Yamaha DM2000 Digital Production Console in 2005.

Case Study #5

Multi-purpose Processor

Rakuonsha (Studio 2001)
(Tokyo, Japan)

Rakuonsha, an audio editing and sound production company that works on movies and animated films, opened their new Studio 2001 dubbing stage in March of 2006. Theirs is only the second dubbing stage to have been THX-certified in Japan, and Rakuonsha's THX engineers work primarily on content for cinema.

In order to gain THX certification as a dubbing stage, finely-tuned delay and EQ are required for each speaker. Normally, this would require the installation of dedicated audio processors for each individual speaker in the system; however, a single DME64N can easily meet this challenge and can even facilitate fine-tuning of system parameters via DME Manager running on a PC.

Receiving edited audio from a mixer, the DME64N at Rakuonsha adjusts delay and EQ for each of the studio's 23 speakers and controls these speakers so as to maintain the quality required in a THX-certified environment. Whenever needed, furthermore, the DME unit can also be operated from its front panel to instantly switch between different monitoring environments.

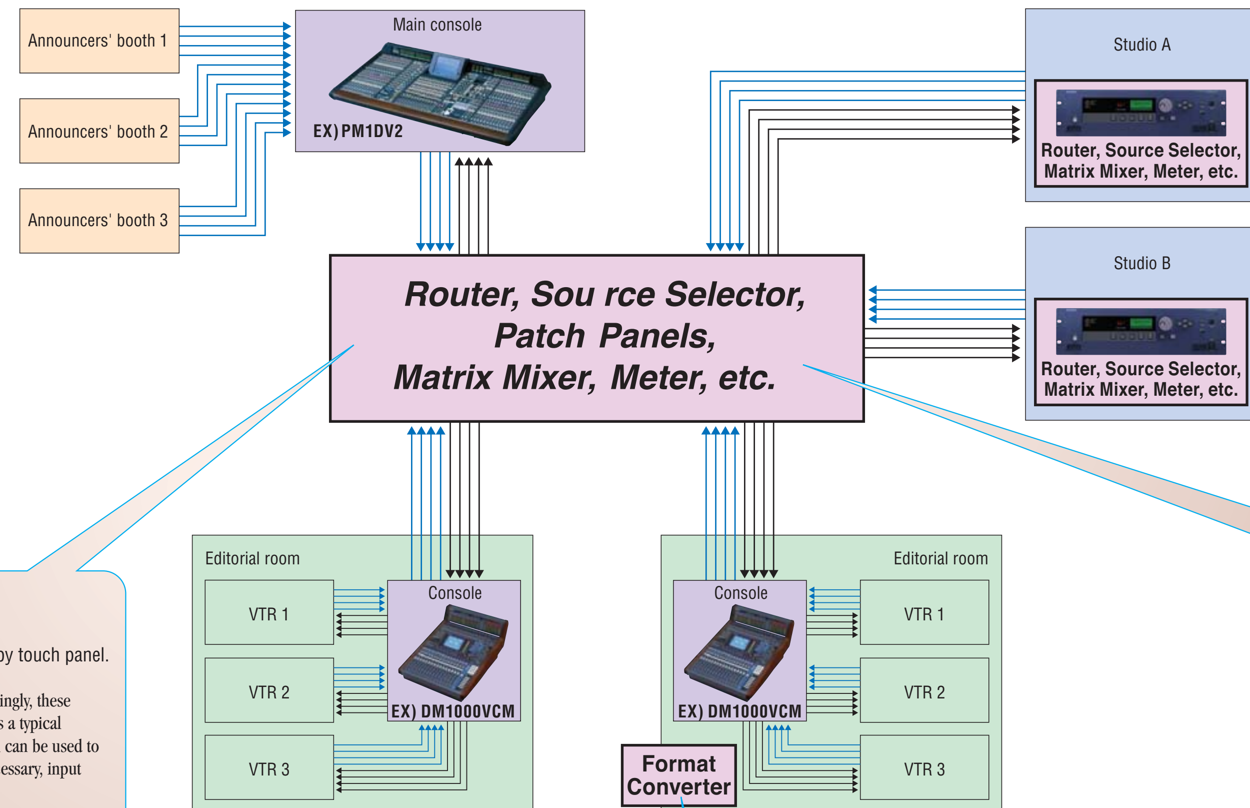
Network Control

Router, Source Selector, Matrix Mixer, and Audio Format Converter

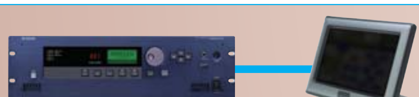
The vast array of studios, auxiliary control rooms, announcer's booths, video editing suites, and other facilities making up today's broadcasting stations must all be interconnected via a network, usually taking the form of analog cables. And although a standard procedure within stations, the reconfiguration of network connections for different applications is highly critical in nature and mistakes are simply unforgivable. Nevertheless, analog cables can be huge, often leading to extreme difficulty and complexity in connection, routing, and installation.

When working simultaneously with multiple lines in a conventional analog network, broadcast engineers have to install dedicated routers, source selectors, matrix mixers, and other expensive gear. Using a Yamaha digital mixing engine, however, you can freely and flexibly call up practically any number of built-in routers, source selectors, and matrix mixers for integration into a network. Where necessary, these configurations can also be stored and later recalled using your PC or a wide assortment of intuitive controllers. Thanks to this sophisticated approach, it is little wonder that the DME Series allows network operations to be carried out with superior levels of efficiency, practically no potential for error, and ease-of-use that is second to none, all without compromising its stunning sonic performance.

With four slots for mini YGDAI (MY) cards on the rear panel of the DME64N and one on the DME24N, this series offers a wholly-flexible open structure for the input and output of audio signals. You can freely select MY cards based not only on the required number of input and output channels, but also on the audio formats needed for the application in question. As such, Yamaha's digital mixing engines make easy work of conversion between numerous digital and analog formats.



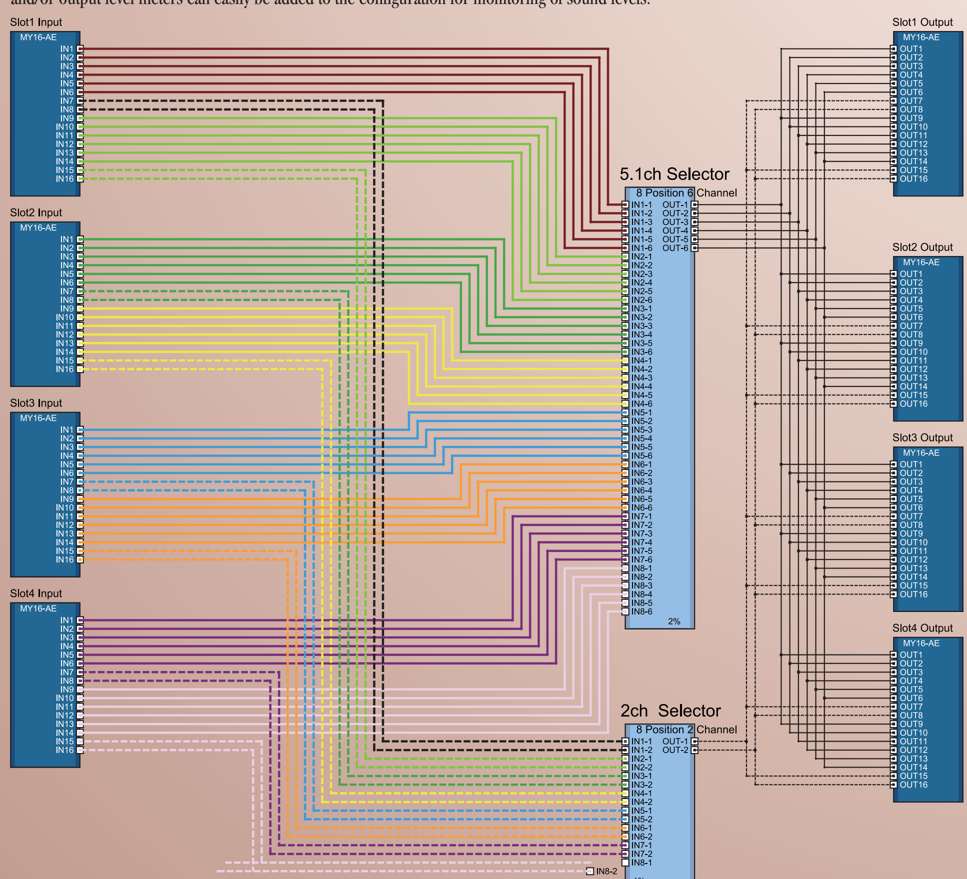
It is customary in broadcasting applications that audio be exchanged between individual processing sections in multichannel format, often employing 4 channels. And if surround sound must also be supported, the number of network channels will increase accordingly. As a solution for fail-safe, efficient, and inherently reliable control of these multichannel networks, nothing beats configurations setup on a DME64N.

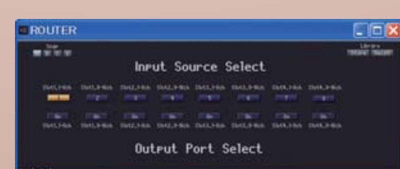


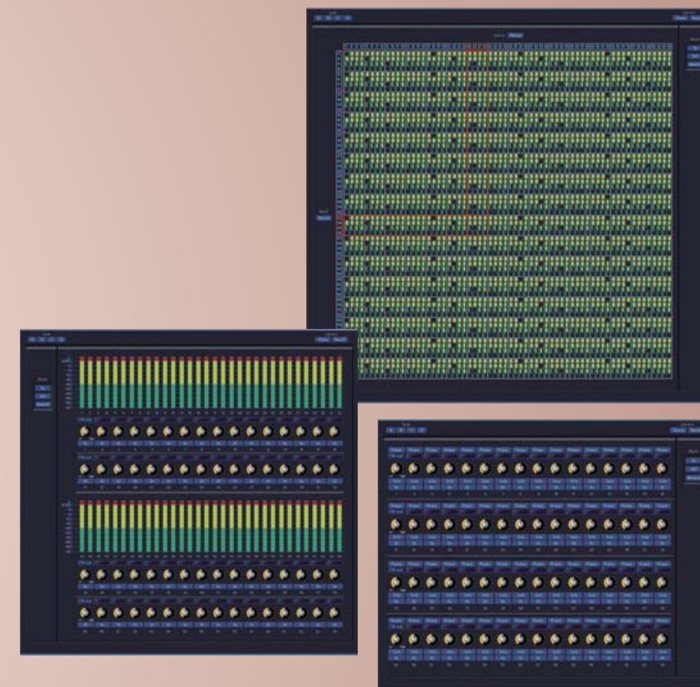
Multichannel Router

DME's User Control Panel allows routing operation intuitively by touch panel.

High-definition video recorders, which have become increasingly popular in recent years, can also record eight tracks of audio; accordingly, these devices are often used to record the six channels of standard 5.1 surround sound together with a stereo mix. The configuration below is a typical example of how the DME Series can route eight channels as one for just such an application. As shown, a simple, intuitive control panel can be used to select a source from among many groups of eight channels and to also specify the ultimate destination of these audio signals. And if necessary, input and/or output level meters can easily be added to the configuration for monitoring of sound levels.







Matrix Mixer

With a 64x64 channel matrix mixer as shown above, you can freely assign any source to any output channel and also adjust their levels as required. Even the most complex of configurations can, therefore, be easily and efficiently setup via an easy-to-understand graphic interface on your PC. What's more, numerous configurations created in this way can be conveniently stored in memory and instantly recalled whenever the situation demands.

Format Converter

A vast assortment of digital devices is regularly put to use in today's broadcasting and music production studios. While exchange between devices is never an issue with analog signals, the diverse range of formats that exist for digital audio can give rise to serious compatibility problems, and it's not unusual to see multiple format converters used side-by-side in order to resolve incompatibilities. Rather than using expensive, mono-functional format converters in this role, however, it is far more efficient and economical to take advantage of the open configuration for signal input and output afforded by the DME Series. The DME64N, for example, has four slots for mini-YGDAI (MY) cards, while the DME24N has one. MY cards can thus be freely selected to convert between different audio formats and channel numbers with a high degree of precision and flexibility. Aside from high-quality 96-kHz/24-bit AD/DA conversion, the MY card series also provides support for ADAT, TASCAM, AES/EBU, MADI, CobraNetTM. In order to cater for a wide variety of possible applications, these digital I/O cards have been made available in both 8- and 16-channel versions. With its four MY slots, therefore, the DME64N can be provided with as many as 64 digital I/O channels.

Example: AES/EBU 8chI/O(75 BNC) <—> AES/EBU 8chI/O (with sampling rate converter) <—> a-dat 16chI/O <—> AES/EBU 16chI/O

Example: AES/EBU 40chI/O <—> a-dat 16chI/O

Example: MADI 16chI/O <—> AES/EBU 16chI/O <—> a-dat 16chI/O <—> TDIF 16chI/O

Turn to page 49 for information on how to download and install DME Designer.

#6 Case Study

Multi-purpose Processor

Rainbow Network Communications
(New York, USA)

Rainbow Television Network (RNC) a subsidiary of Rainbow Network Communications (Bethpage, NY) is a supplier of SD and HD program origination and distribution services to the multimedia industry. With a complex array of live and prerecorded programming to handle, Rainbow recognized the need for a new generation of live video multichannel master control rooms and entrusted the systems challenge to engineering firm Communications Engineering Inc. (CEI) as well as the RNC engineering team.

The new facility provides broadcast capabilities for RNC's entire broadcast "assembly line," which covers the gamut from signal ingestion, program editing, promo creation, quality control, and live and recorded playback through automated multichannel master control and satellite uplink.

The RNC facility originates 16 individual program channels presently, with three types of programming: automated delivery of previously recorded programming, which includes long-play server-based programs; live television events such as sports programming; and promos. These different operational processes led to alternative master control room designs including the DME systems.

The design team catalogued each channel's unique requirements and categorized the key similarities and differences in order to design an overall facility approach. The infrastructure was developed to provide ample capability to each channel in the form of router matrices, audio and video I/O, physical space for operations, pre-production activities, and other parameters.

Ten Yamaha DME64N units were installed on the premises with intended use for audio/video control between edit rooms and for end of television programming rolling credits. In SD/HD SDI On-Air Master Control the DME64N is used under automation control to perform an A-B-A Fade / Mix commercial/promotional segments and program segments in parallel with a DVE squeeze back during credit rolls. In this application the Group1 Pair 1 Audio is an AES/EBU Dolby E 5.1 Channel Surround Mix and the Group1 Pair 2 Audio is a PCM Left SAP / Right/SAP Mix.
In Tape Ingest workstations the DME64N is used to synthesize a 5.1 Channel Surround Mix from an AES/EBU PCM Left Front / Right Front Mix in preparation for use in On-Air Master Control.

Multi-purpose Processor

#7 Case Study

Sun Television Co.
(Hyogo, Japan)

Sun Television, an independent TV channel based in Japan's Hyogo Prefecture, uses a DME24N in their outside-broadcast van to handle over 100 live broadcasts per year, mainly of local Hanshin Tigers baseball games and other sports events. The DME24N is used as a multi-purpose processor to conveniently change the configuration for each show, thus providing invaluable support for local programming.

Multi-purpose Processor

#8 Case Study

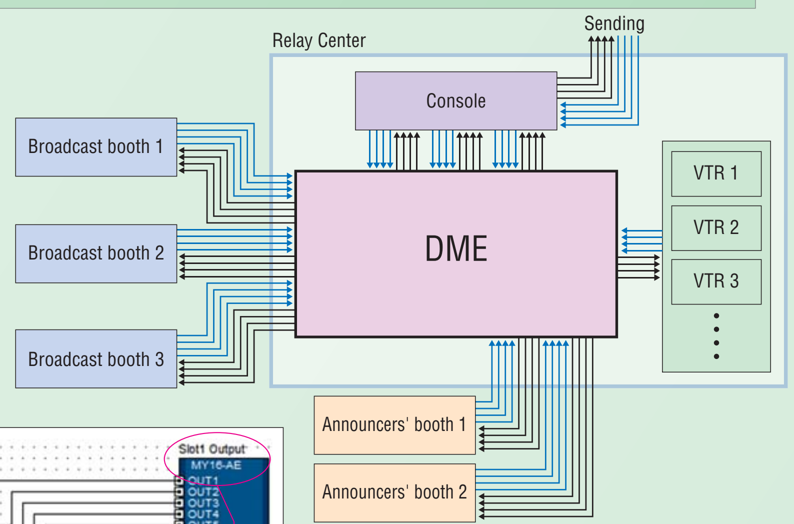
Sala Medusa
(Rome, Italy)

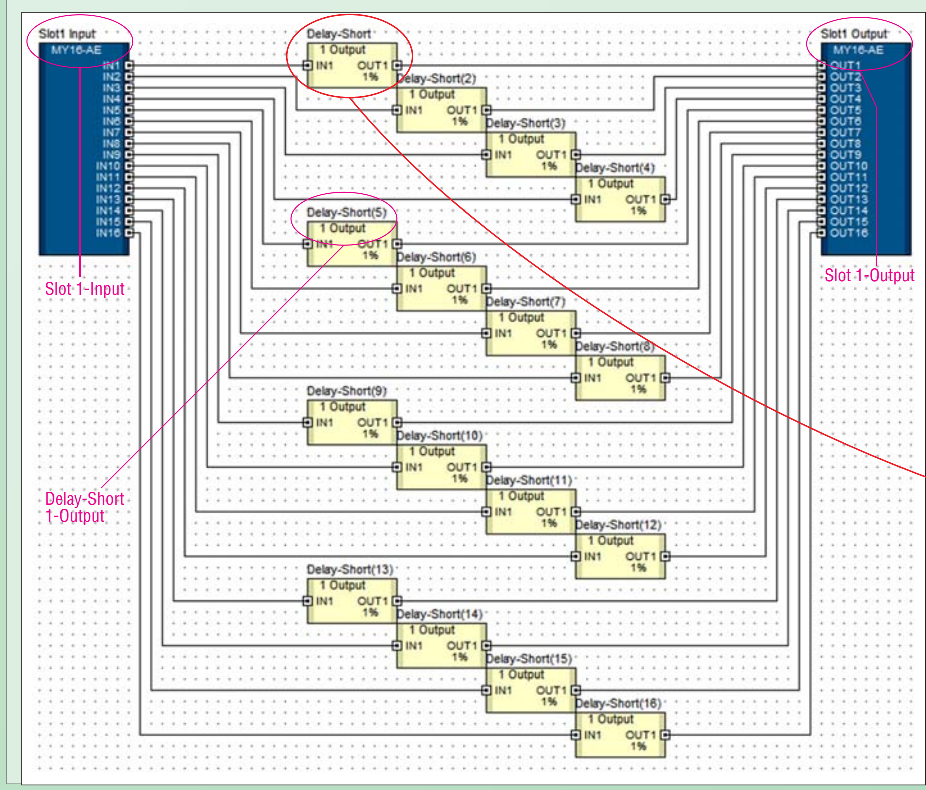
Medusa Film is one of the most important movie production companies in Italy, and Sala Medusa is a small private cinema located within the Rome premises of Mediaset (Italy's largest private broadcaster). A DME64N has been installed at Sala Medusa to provide source selection, dynamic control, and surround matrix functionality. Audio signals from a DVD player, various video-tape machines, and an old film projector are input via MY8-ADDA96 analog cards into the DME unit, where they are mixed, dynamically controlled, and equalized. In addition, the processed signals are then routed by the DME64N to the venue's power amplifiers and speakers.

Live Broadcasting and Digital Audio Networks

Notwithstanding limited hardware resources, live broadcast engineers must be ready to respond instantly and effectively to any changes in conditions. Add to this the fact that mistakes can never be tolerated and it becomes clear why live broadcasting is probably the most merciless environment in which audio is processed. Yet even here, the DME Series has much to offer.
What will impress live broadcast engineers most about Yamaha's digital mixing engines is how easily they can be setup as needed in practically any location to provide a multifunctional, highly flexible DSP box. Each hardware unit includes an expansive array of advanced digital components such as routers and matrix mixers for managing signal flow, multiple delays to compensate for time lag, and equalizers, compressors, and limiters for further processing of audio signals. And with highly accurate speaker processors, Yamaha SPX effects, and a host of other powerful components also built-in, you have at your disposal a vast arsenal of highly potent tools for tactful, resourceful, and intuitive deployment whenever the situation demands.
All components for routing, mixing, or processing of audio are accessed and configured by programming the digital mixing engine itself, so no longer will you need stack up and connect vast banks of cumbersome effect units. Setup couldn't be easier, and convenient memory functions allow configurations to be saved and instantly recalled to respond to changes in conditions; furthermore, as problems associated with connections and wiring between multiple physical processors are no longer relevant, unprecedented levels of reliability are achieved. In the DME64N, for example, Yamaha have packed an incredible palette of functions into a mere 3U of rack space, allowing you to make the most efficient use of the space available and also to relocated with unprecedented ease. In the realm of live broadcasting, where resources are limited yet flexibility is paramount, surely there can be no better solution.
When it comes to live broadcasting of sporting events, furthermore, the transfer of audio signals over long distances is a major headache. Recent years have seen the emergence of a wide range of digital audio formats and network protocols offering sophisticated solutions to this problem. As you would expect from Yamaha digital audio devices, the DME Series provides full support for these technologies, integrating seamlessly with CobraNet™, EtherSound™, MADI, and more.

When broadcasting golf and other large-scale sporting events, broadcast and production booths must be setup in multiple locations. Each of the broadcast booths returns numerous audio and monitor signals, and when outputs from announcers' booths and feeds of pre-recorded material are also taken into consideration, it's not hard to see how communication networks become so large and complex. And as if this wasn't enough, changes in conditions often demand reconfiguration at the drop of a hat. Offering a speedy, efficient, and inherently reliable solution for managing these highly complicated systems, the DME Series will allow you go live – and stay live – with utmost confidence.

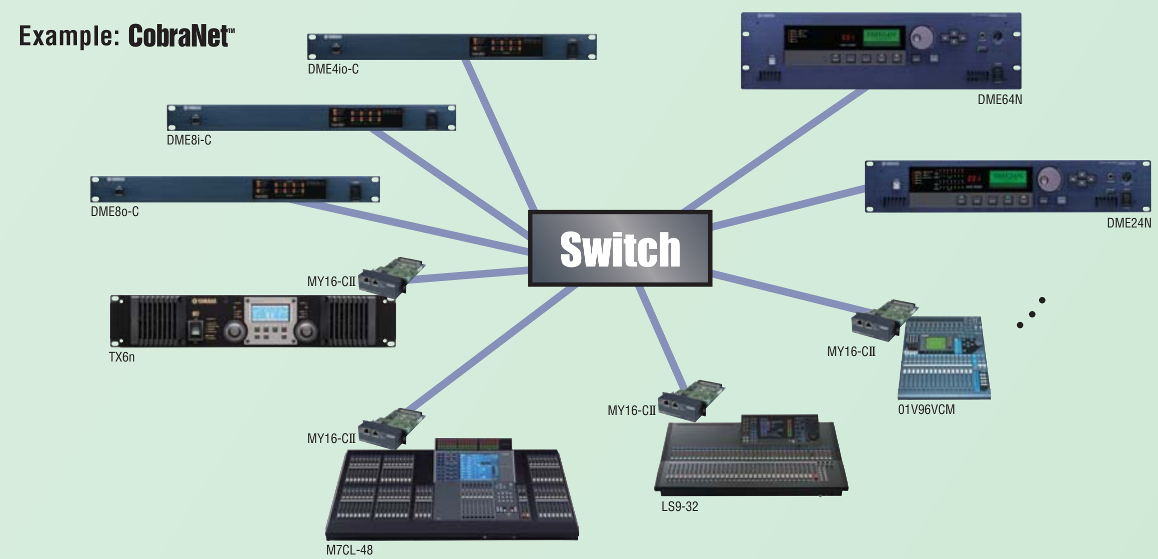




As audio lines are often processed individually in live broadcasting applications, time lag can easily occur. Added to this, an even bigger problem is the loss of synchronization with video signals, resulting from the significant latency generally experienced in the processing of video for broadcasting. Large numbers of delays are normally used to compensate for these time shifts, and with Yamaha digital mixing engines, the number of built-in delay components that can be used is limited only by processing performance. Take, for example, the multi-delay configuration on the left: Despite the fact that 16 short delays are used, it requires but 1% of the DME's total processing power. And as shown below, you can intuitively configure and adjust each delay component from your PC via an easy-to-use control panel.

Digital Audio Networks

The immense scale of arenas, stadiums, courses, and other sporting venues presents a major challenge for broadcast engineers. Laying of mile after mile of analog cable is not only extremely hard work and expensive, but it also comes with increased risk of disconnection, damage, and other associated problems. Further, compromised sound quality is an unavoidable consequence of long-distance transfer over analog networks. Today, a range of innovative digital-audio technologies offering outstanding advantage are increasingly being put to use in long-distance signal transfer. Immune to degradation of audio quality, technologies such as CobraNet™ and EtherSound™ allow networks to be configured at low cost, and network routing can also be freely and intuitively modified using a PC. What this all means is that highly developed, inexpensive solutions for long-distance transfer and networking can be realized more easily than ever before.



CobraNet™

CobraNet™ is a sophisticated technology that facilitates bidirectional transfer of up to 64 audio and control channels via Ethernet cable. As a fully integrated system, it allows digital data to be carried up to 100 meters* between devices by common CAT-5/100Base-TX cabling, or up to 2 kilometers by optical fiber. Utilizing a star topology, CobraNet is ideal for configurations featuring one production booth in combination with multiple broadcast booths, and it's fast, with a minimum latency of 1.33 ms**. Thanks to redundant systems included as standard, this technology is inherently reliable, and using commonly available cables and connections, it is remarkably economical and easy to use. CobraNet supports Ethernet variants. It uses standard Ethernet packets and network infrastructure, and therefore, is compatible with commonly available, inexpensive controllers, switches, and cabling. CobraNet provides transparent digital audio transmission with no degradation of the audio signal, and no digital distortion or artifacts are introduced during transmission. This technology is capable of transmitting 24-bit audio with a dynamic range of 146.24 dB, and distortion is a mere 0.0000049% at full level. Frequency response is 0 to 24 kHz ±0 dB. In short, the performance of CobraNet is significantly better than that provided by today's A/D and D/A conversion technology.
CobraNet capability can be conveniently added to a Yamaha digital mixing engine by installing an MY16-CII digital network card into one of the device's expansion slots. Additionally, the activity status and conditions of devices connected to the CobraNet network can be monitored using the Yamaha software package NetworkAmp Manager.

** The maximum length of cables may differ depending on the quality of the Ethernet equipment or the particular settings used.*
*** CobraNet™ low latency mode is now supported, allowing you to choose 5.33, 2.66, or 1.33 ms.*

MY16-CII

EtherSound

Implementing the highly reliable ring topology, EtherSound networking technology is more suited to relatively simple sound systems with direct connections. Networks are easy to connect and allow multiple channels of digital audio to be transferred over standard Ethernet cable with extremely low latency. This advanced, easy-to-manage protocol is designed to handle up to 64 channels of digital audio and will easily transfer 48 channels of 24-bit, 48-kHz audio in both directions over distances up to 100 meters with appropriate high-performance cables. As with CobraNet networks, you can use standard Ethernet switches to create any network configuration that suits your needs.
The most convenient way to add EtherSound capability to a DME Series device is to install one or more MY16-ES64 digital network cards into one or more expansion slots according to the performance and channel capacity required.

MY16-ES64

MADI: the First Choice for Production, Broadcast, and Live Recording

An industry-standard communication protocol for multi-channel digital audio, Multichannel Audio Digital Interface (MADI) allows transmission of up to 64 channels of 24-bit, 48-kHz audio via a single coaxial or optical cable. Coaxial transmission lines can be up to 100 meters in length, while optical lines provide error-free transmission up to 2,000 meters. Using 75-ohm coaxial cable with BNC connectors or standard 62.5/125μ network optical cables, MADI is the simplest means of economically transmitting sample-accurate audio data over long distances.

MY16-MD64

#9 Case Study

Live Broadcasting

2006 Mitsui Sumitomo VISA Taiheiyo Masters

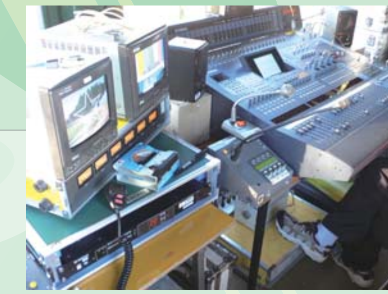
Surround-Sound Golf Broadcast

Broadcasting System Overview

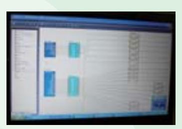
- Tokyo Broadcasting System (TBS) used surround sound in live and delayed broadcast of the 2006 Mitsui Sumitomo VISA Taiheiyo Masters – one of the biggest events on Japan's golfing calendar.
- Rather than having every individual audio signal converge on one mixing center, multiple sound booths were setup to recreate the natural surround sound of the corresponding holes, and from here, surround mixes were fed to a central broadcast center.
- In line with this approach, mix engineers at each sound booth carried out surround-sound monitoring of signals returned from the broadcast center.
- Surround sound from each sound booth was mixed in 4.0 format. With commentary and the like added at the broadcast center and the LFE channel not used, the final surround-sound product was in 5.0 format.
- Each sound booth was provided with a Yamaha DM1000 or DM2000 Digital Production Console.
- As a direct result of broadcasting in surround sound, the system comprised a huge number of channels; however, smooth and efficient compensation for time lag was made possible by a Yamaha DME64N Digital Mixing Engine at the broadcast center.
- Audio signals were transferred digitally using a combination of CobraNet™ and fiber-optic cables, with the Yamaha NHB32-C Network Hub and Bridge used as an interface at each sound booth.

Why Combine CobraNet™ and Fiber-optic Cables?

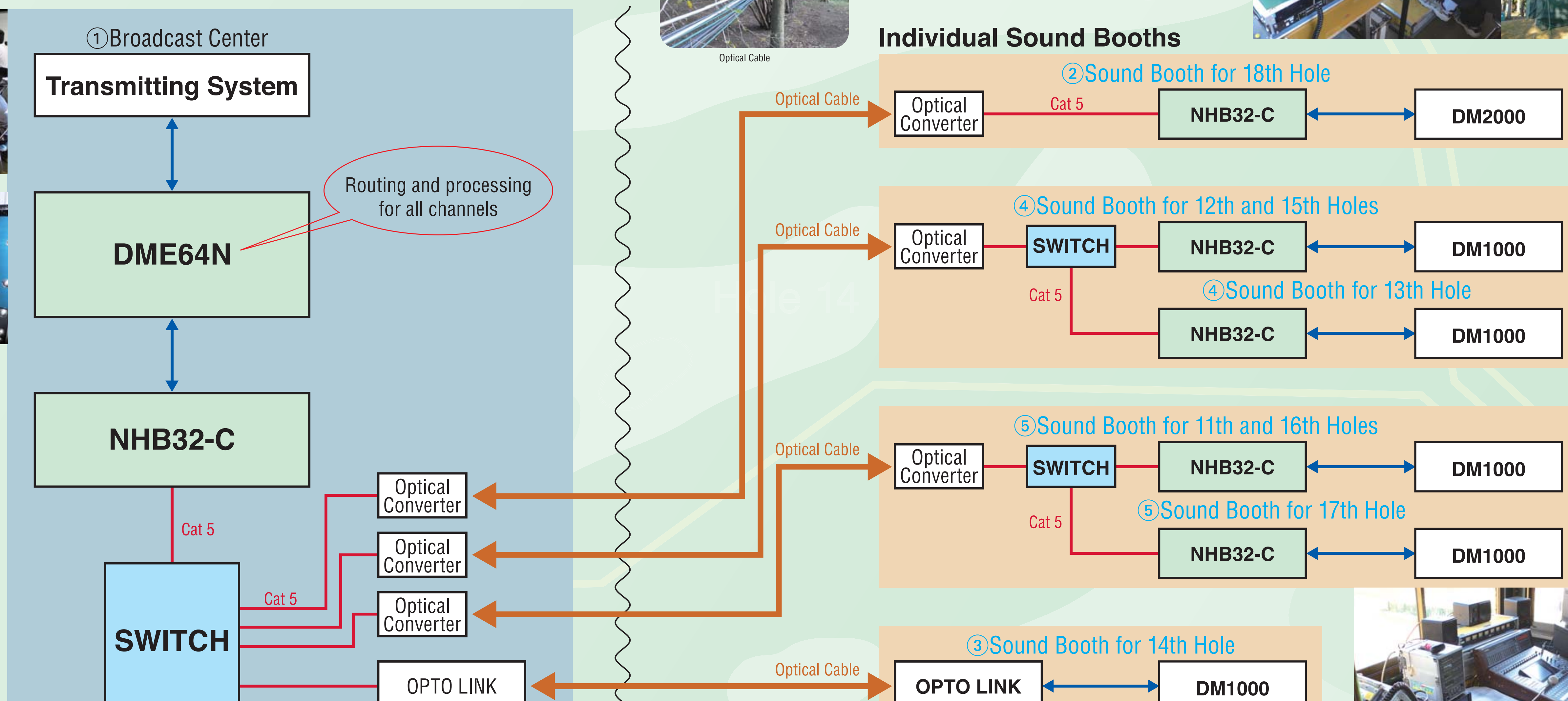
- With sound booths scattered around the golf course and a single broadcast center at the heart of the network, the star topology employed by the CobraNet™ protocol was ideal.
- As the NHB32-Cs allowed patch changes from anywhere on the network, patch processes and NHB32-C status monitoring for each booth could be conveniently carried out from the broadcast center.
- CobraNet™ audio signals, which are exchanged over standard Ethernet networks, can be converted into optical signals using general-purpose media converters; and therefore, the multi-fiber optical network needed to transfer video signals could also be used for long-distance transfer of audio.
- The redundant line functionality provided by CobraNet™ could be easily used to protect the integrity of signals in case anything went wrong.
- When compared with the time lag resulting from encoding, transfer, and decoding of video signals, latency in CobraNet™ networks is relatively small; accordingly, compensation for this latency was easily achieved with no adverse effect whatsoever on system operation.



Block Diagram of Golf Tournament Broadcasting System



Optical Cable



2006 Mitsui Sumitomo VISA Taiheiyo Masters
Surround-Sound Golf Broadcast

Of the golf events broadcast live at TBS, the Mitsui Sumitomo VISA Taiheiyo Masters is unquestionably the largest. In terms of spectators, the VISA Masters has incredible drawing power, and every year without exception, the passionate applause with which top class players are greeted to the 18th green proves quite moving. At Tokyo Broadcasting System (TBS), Mr. Ikuo Hirai of the Program Engineering Department wondered whether it would be possible to replicate this emotive scene in surround sound and deliver it to living rooms all over Japan. Around June of 2006, he and his team started to give serious consideration to the design of just such a system for the VISA Masters to be held later that year. After many meetings with upper level management, approval was finally given under the condition that surround sound would enrich the viewing experience, and in doing so, would also help to boost the popularity of golf, particularly among Japanese men. The following is a brief introduction to the system subsequently designed, setup, and successfully operated by TBS at the 2006 Mitsui Sumitomo VISA Taiheiyo Masters.

1. System Configuration

Instead of bringing all signals together at a central location, live broadcasting of golf at TBS uses an approach whereby audio and video for the holes being broadcast are packaged together and only then sent to the broadcast center. The natural sound of each hole is recreated within dedicated, localized sound booths, and commentary, reports, interviews, and other speech are added to this multichannel audio within the broadcast center in order to realize the finished audio product. In configuring a system in line with this approach, the most difficult issue to be resolved was whether or not the return feed to each of the sound booths should be in surround or stereo format. While Mr. Hirai's team did consider returning downmixed stereo, they ultimately came to the conclusion that mixing at the individual booths – where the original surround mixes would actually be created – was more critical in terms of overall sound than mixing at the broadcast center. For the VISA Masters, therefore, it was decided that each sound booth should be provided with functionality for reliable surround-sound monitoring.

- DME64N for Delay Processing

In the macro-system also handling video, the audio and video data from broadcast holes was, after first being adjusted for time lag using delays, split between a router and a main-line mixer. The feed sent to the router was used for recording on a VTR, while the feed to the main-line mixer was used for live takes. As the recording equipment used for this project could only support four channels of audio, each sound booth created a multichannel mix for the corresponding hole using four surround channels – L, R, Ls, and Rs. Ultimately, commentary and other speech were added to this 4.0 feed at the broadcast center, resulting in 5.0 surround sound; furthermore, the LFE channel was deemed unnecessary for this particular event.

With the number of audio channels greatly increased as a result of handling surround sound, compensation for time lag in the video signals upon arrival at the broadcast center using a vast bank of delay units would have been completely impractical. Instead, the team used a Yamaha DME64N fitted with digital I/O cards to perform this multi-delay processing. Although the digital mixing engine was used only for time-lag compensation at the VISA Masters, Mr. Hirai and his team are fully aware of the flexibility and power it affords when used to intuitively configure compressors, DDAs, ADAs, routers, and a rich selection of other components. At TBS, the DME Series is not seen as being suited only to domestic broadcasting of the largest-scale events. On the contrary, it is also considered an ideal solution in cases where overseas broadcasts must be carried out with compact gear.

- System for Transfer of Audio to Broadcast Holes

In terms of the actual transfer of audio within the system, four channels were used to carry signals from sound booth to broadcast center, but twenty channels were required for the return leg, and this marked increase in the number of return channels is a direct result of using surround sound. Having determined that the same monitoring signals should be returned to each hole and having also concluded that transmission thereof via a network would be most efficient, Mr. Hirai decided to use Yamaha's NHB32-C Network Hub and Bridge, which facilitates integration of CobraNet™. Thanks to the adoption of these devices at the sound booths, all required channels could be transferred smoothly and without any difficulty.

Yamaha's NHB32-Cs are controlled using a network protocol, and up to eight units can be included within a system. Connection to other network devices is carried out via Cat 5 cables and hubs, allowing devices to be distributed in much the same way as networked PCs. For example, in situations where a pair of sound booths shared a common direction from the broadcast center, they could be served by a single master cable, with signals then split at a suitable intermediate point using a hub. This approach was extremely efficient in terms of cabling work.

When it comes to I/O support, the NHB32-C provides 16 input/16 output AES/EBU channels (or when converted to analog, 32 input/ 32 output); accordingly, each individual unit could handle the VISA Masters signal flow with ease. And while the Cat 5 cables used by CobraNet™ cannot in principle be extended for more than about 100 meters, it is important to bear in mind that the TBS system differed significantly from others setup specifically for transmission over long distances. To elaborate, CobraNet™ allows signals fed into the network at any point to be extracted at any other point, and with excellent overall cost performance also taken into consideration, this was instrumental in the decision to employ the technology.

As a maximum transmission length of 100 meters is of absolutely no use in terms of golf broadcasts, TBS utilized fiber-optic media converters in order to extend this length significantly, and fiber optic cables were thus used instead of Cat 5 cables to cover the majority of the distance to each broadcast hole. When fiber-optic media converters are introduced into a system in this way, it has been reported that transmission distances of several kilometers are possible with multimode optical fiber, while this increases to several dozen kilometers with singlemode fiber. This singlemode variety of optical fiber can also be used for camera cables in television stations, so Mr. Hirai's team decided to use singlemode media converters. Thanks to this approach, the considerable transmission distances to be covered at the tournament posed no problems whatsoever, and in addition, the required amount of analog cabling could also be reduced by a great deal.

Incidentally, 1 kilometer of optical fiber and 100 meters of Cat 5 were needed to cover the longest distance in the VISA Masters network – namely, from the broadcast center to the commentator's box at the 17th hole – and everything worked perfectly. In fact, pre-event testing back at the television center identified no problems even at a distance of 1.5 kilometers, so it would seem that the maximum distance can be extended even further still.

Two different types of fiber-optic media converter are available – one converting signals from a pair of optical fibers into signals for one Cat 5 cable; the other converting from one optical fiber to one Cat 5. Just because two fibers are used by the former does not guarantee increased reliability, however, and communication will be lost if either optical fiber were to be damaged. So, with no significant difference in terms of risk, it was decided at TBS to use the single-fiber type. This approach is also highly efficient on another front, as it allows construction of both primary and secondary circuits.

It should be noted that switching of transmission systems over to digital did have some disadvantages. For example, as all inputs and outputs would obviously be digital, the sound booths at each hole either needed banks of AD/DA converters or digital mixers for processing of all of these signals. In addition, synchronization becomes a much more complex task with digital. When synchrony between signals is lost, you run the risk of introducing noise in the form of regular clicks or completely loosing all sound. At this event in particular, Mr. Hirai's team became keenly aware of the value of digital transmission devices that can be conveniently and reliably synchronized.

- DM1000 and DM2000 Mixers for Sound Booths

TBS setup six individual sound booths at the VISA Masters: one for each of the 13th, 14th, 17th, and 18th holes, with a further two handling the 11th and 16th holes and the 12th and 15th holes, respectively. Each of these booths was provided with either a DM1000 or DM2000 Digital Production Console. As the transmission network was completely digital and only a limited amount of space was available inside each booth, it was decided that the AD/DA conversion functionality of a digital mixer such as the Yamaha DM1000 would be a far better option than racks of individual hardware converters. In principle, all NHB32-C input and output signals were passed through the digital mixer; accordingly, the returned surround-sound audio, in addition to other audio signals in the network, could be converted from digital within the mixer and output as analog audio via the console's OMNI OUT ports. Furthermore, as the sound booths for the 17th hole and for the 11th and 16th holes were relatively close to each other, they shared a common optical media converter, with a hub splitting and combining signals as required. Naturally, this approach produced further increases in signal transfer efficiency.

2. Creating Surround Sound

The first hurdle to be cleared in the creation of surround sound mixes was directionality. While the position of the listener is obvious in the case of soccer broadcasts, there are a number of sports where selecting this virtual position is not so easy, and golf is one of them. Simply put, the greater the number of potential camera positions, the more difficult creating believable surround sound becomes. Ideally, Mr. Hirai's team would have loved to adjust the surround-sound directionality to match the camera angle, so that the listener would, for example, hear shots onto the green slightly off with crowd noise also in the distance, or hear the audience all around in the case of back shots. However, upon careful consideration of the network complexity and operational difficulties that would accompany such kaleidoscopic changes in the sonic environment, they decided against it on this particular occasion. Instead, the team identified audio characteristics unique to each hole and made every effort to reproduce them in surround sound, so for example, one hole had traffic running along a side of the audio field, while another had carts coming and going at the rear.

- Surround Monitoring Environment at Sound Booths

TBS took some bold steps when setting up the surround-sound monitoring environment at sound booths. First of all, in order to provide maximum support to decision-making by the mix engineers responsible for creating the surround-sound mixes, they discarded the conventional headphone-based approach and installed five monitoring speakers in each of the sound booths. This change was well received by the engineers, as it allowed them to easily evaluate directionality and to check for any differences in the monitoring mix sent back from the broadcast center. And perhaps because of the excellent soundproofing of the booths, sound leakage presented no major difficulties at the VISA Masters. Mr. Hirai also elected to use a large acoustic space for central monitoring, with the mixer, switcher, and director all surrounded by the speakers. The policy for broadcast-center operations at TBS has always been to keep audio and video under one roof, and in addition, the approach adopted for the VISA Masters was expected to eliminate any sense that the event was focused primarily on audio. With the project's technical director convinced that this could be highly effective, the team went about procuring an empty equipment truck to provide the necessary space for accurate surround-sound monitoring. Rather than have a central engineer busily blend the mixes, a single person issuing surround-related instructions to each sound booth turned out to be surprisingly effective, and there was a real sense of the individual surround mixes gradually converging.

- Downmixing at Auxiliary Control Room

Back at TBS television center, the auxiliary control room team worked in line with the event's sound design to finalize the surround sound. Broadcasting over the event's four days took the following format.

- Day 1: Delayed broadcasts on the BS-i satellite channel
- Day 2: Live BS-i broadcasts
- Day 3: Live terrestrial broadcasts
- Day 4: Delayed terrestrial broadcasts

In terms of broadcasting modes, dual surround and stereo was used for digital terrestrial, stereo only for analog terrestrial, and surround only for BS-i. Generally speaking, live broadcasts were encoded with Dolby E, while SRS was used with recordings for delayed broadcasts. Background music and inserts were remixed for surround-sound broadcasting, and all audio material for stereo broadcasts was prepared by downmixing at the auxiliary control room. Ultimately, no material recorded in stereo at the event was broadcast.

As is often the case with monitoring within an auxiliary control room, it was not possible to setup the same equipment and environment as being used on-location; accordingly, an audio-only connection was established between the golf course and the auxiliary control room in order to ensure close communication. Surround-sound phases were handled carefully during monitoring; special consideration was given to linking of the front and rear audio; and similar care and attention were also given to downmixing. Of course, the team also carefully monitored stereo quality, and both 5.0 surround and stereo were prepared simultaneously. For syndication purposes and the like, TBS also had to cater for a wide range of other requirements, such as passing the natural sound alone through the on-air auxiliary control room. With so much to take care of – including the exchange of line signals, master feed, forwarding of raw material to the recording center, addition of background music and inserts, and downmix management – they were certainly kept busy.

Slovak Radio, MediaTech Ltd., Slovakia
(Bratislava, Slovakia)

In order to satisfy its need for outside broadcasting via ISDN or satellite and the recording of music and other radio content off-site, the Slovakian national broadcaster, Slovak Radio, commissioned a pair of outside-broadcast vans from Mediatech Limited. While these mobile studios would have to be full featured and easy to use, Slovak Radio also required them to be small, fast, flexible, and relatively inexpensive to operate. Furthermore, in consideration of driving license regulations, road taxes, and urban traffic restrictions, the vehicles themselves were limited to a maximum weight of 3.5 tons.

Because of the focus on live events and the compact design, light weight, and ease-of-use that this dictates, Mediatech decided to install a Yamaha M7CL-32 Digital Mixing Console in each of the vans. Furthermore, a number of processing devices designed and manufactured by that company were added to provide functionality required for studio applications. Thanks to this configuration, the system provides plenty of two-track returns; independent talkback and intercom functionality; monitoring features such as mono, monitor dim, mute, and monitor speaker switching; and various on-air features like automatic green/red lights, talkback blocking, and fader starts. With proven track records in a large number of installations, each of the additional devices (JZP16, PIS8-Icom8, MFS 08-M7CL, and MT-ESC SBC2) affords the same levels of convenience and user comfort as expected from non-mobile recording and broadcast studios; furthermore, they also make the mixing console fast, flexible, and easy-to-use in live mixing situations.

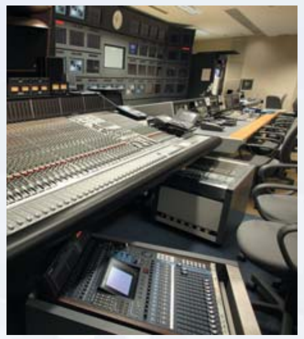
All audio signal transmission and recording within the outside-broadcast vans is based on state-of-the-art EtherSound technology, with Yamaha's MY16-ES64 and MY16-EX Mini-YGDAI cards providing the mixing console with the necessary interfacing functionality. Simultaneous recording of up to 60 tracks (28 direct from stage rack, 32 from console) is made possible using Digigram's LX6464ES sound card. Furthermore, the stage rack in each van features an NAI48-ES Network Audio Interface for conversion between EtherSound and AES/EBU in addition to Yamaha AD8HR units for AD/DA conversion. The AD8HR's microphone preamps can be remotely controlled from the mixing console, and a pair of DME4io-ES DSP expansion units allows three channels of intercom, independent talkback, auxiliary line, and GPI data to be transferred via the EtherSound network for signaling from studio to stage.

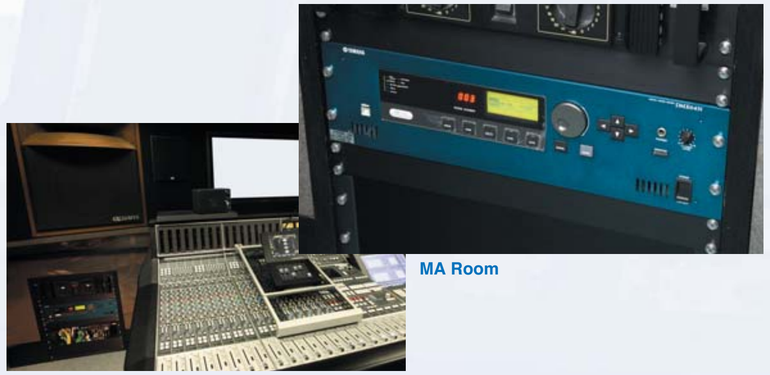
The extensive functionality of the DME4io-ES expansion units is used for signal processing on the audio lines, and EtherSound connection between the studio and stage rack is achieved with heavy-duty Fiberco optical cables. In addition to audio, these cables also carry signals for the intercom, light control, CCTV cameras, and the like. Finally, a UPS provides power for up to two hours of independent operation.

Surround & Multi-purpose Processor

Asahi Broadcasting Corp.
(Osaka, Japan)

A Sub-control Room



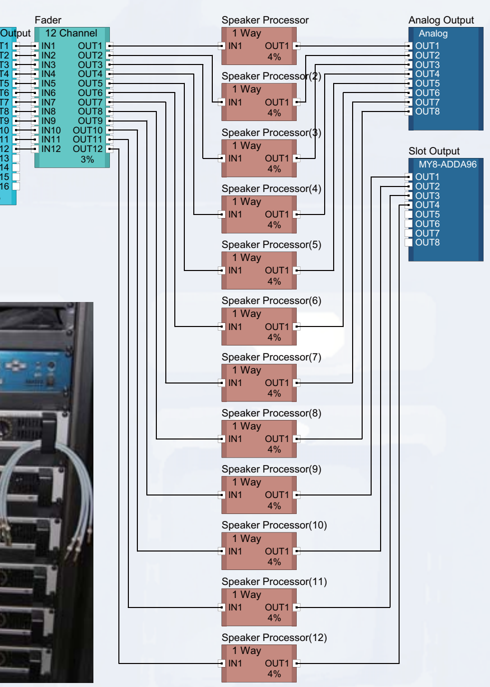


MA Room

At Japan's Asahi Broadcasting, a DME64N is used as an A Sub-control room for *Gokigen Brand New* – a popular variety show. With its surround-sound mixing capabilities, this unit is used as a monitoring processor. The DME64N is also used in the MA room as a monitoring processor for the TV station's Symphony Hall classical music program. Further, the Asahi Broadcasting outside-broadcast van is also equipped with a DME64N and a DME32 for use as a surround processor and a multi-purpose processor, respectively.

Multi-purpose processor

Telesanterno- Gruppo GTV Studio
(Bologna, Italy)



Telesanterno- Gruppo GTV studio is a television production center located in Bologna and was successfully designed by ONDA TELEELECTRONIC. 8 x IF2108 and 4 x IF2208 Installation Series loudspeakers are installed for one of its studio.
Those are powered by six of XP series amplifiers and processed by a DME24N as a matrix router and speaker processor.
All the signals go into M7CL console in the control room and are mixed for live-broadcasting. and television production.

OB Van / Optocore Network

TVV Television Valenciana
(Valencia, Spain)

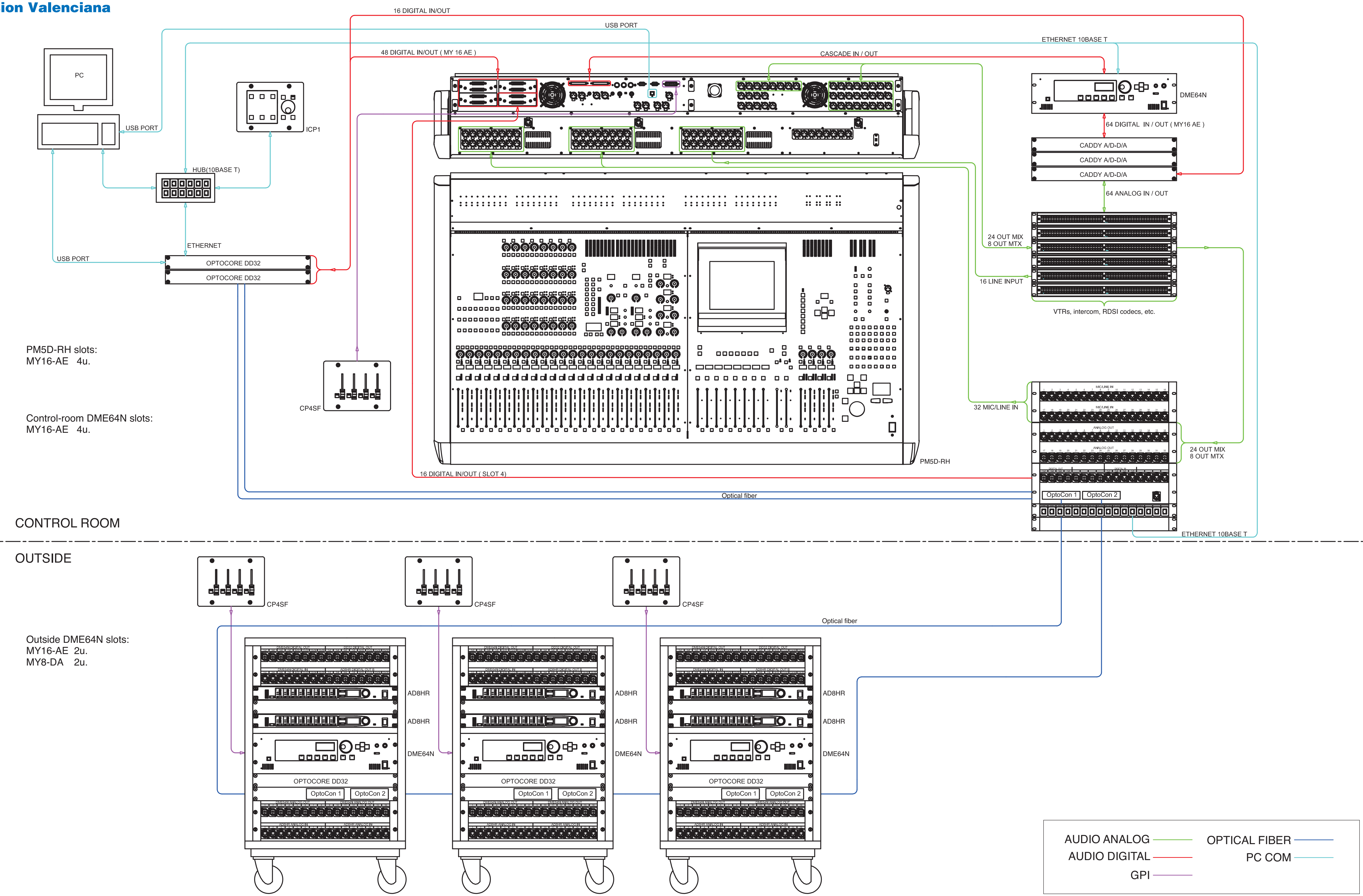
Ràdio Televisió Valenciana is one of the biggest regional radio and TV broadcasters in Spain, and its television arm, Televisió Valenciana (TVV), offers a pair of TV channels – Canal 9 and Punt 2 – to the almost five million people of the Valencia region. On these channels, TVV broadcasts a variety of local programming in the Valencian and Spanish languages, in addition to a broad selection of foreign-language content that they dub for local consumption.

At Canal 9, the sound system for the UM1 outside-broadcasting van was conceived to provide a range of features for TV broadcasting, and using this system, it is possible to perform A/D conversion, audio processing, and signal distribution in several formats simultaneously. Sound is captured and converted using Yamaha AD8HRs, and for long distances in excess of 200 meters, optical fibers are used to send audio together with control and synchronization signals. A PM5D-RH Digital Mixing Console lying at the heart of the system provides four Mini-YGDAI expansion slots that are used to interface with an Optocore network and to cascade with a Yamaha DME64N. This control-room DME serves as a switching matrix for in-house signals (from high-definition digital video, ISDN, and intercom), and it also converts these signals into the required output formats for analog or digital transmission or, for example, recording on VTRs with different dynamic ranges.

A number of racks located at different points within the system include a DME64N for processing of sound in local mode, and this is of particular benefit in broadcast applications where commentators are required to input speech at various points for different TV transmissions. With Yamaha CP4SF remote faders used for GPI control of the DME units, the processing and routing of monitoring signals, feedback, audio sources, pre-mixes, and dynamic processing signals can be easily achieved.

These TVV outside-broadcast vans are used to cover important international sporting events, with feeds provided to all syndicated TV networks. In addition, they are employed in the production of several shows for the Canal 9 and Punt 2 channels, and the system has proved highly effective in the broadcasting of concerts from major venues such as the Palau de la Música and the newly inaugurated Palau de les Arts. Thanks to the powerful DSP and signal-transport functionality realized by integrating the system's PM5D-RH, DME64Ns, and AD8HRs with Optocore technology, outside broadcasting from sporting, music, political, and all other types of event can be handled with confidence and ease.

TVV Television Valenciana
(Valencia, Spain)



Case Study #13

OB Van / Processor

White Mobile - Amek & Vanis
(Modena, Italy)

The Amek & Vanis White Mobile is a large outside-broadcast van optimized for the recording and broadcasting of music events. Within this van, a Yamaha DME24N is used for audio processing and the selection of a range of monitoring signals in 5.1, 3.1, Stereo+Sub, 2.0, and other formats.

A bank of twelve AD824s receives signals from outside sources, converts them into ADAT format, and then routes them to a pair of DM2000V2s configured in cascade mode. From these digital production consoles' OMNI OUTs, the signals then go to the DME24N, where they are processed and distributed to the van's monitor speakers. And using a homemade GPI box integrated with the DME, stored scenes and configurations can be easily and rapidly recalled whenever needed.

Case Study #14

Interface / CobraNet Network

Mizuno Classic, Mainichi Broadcasting System, Inc.
(Osaka, Japan)

Using a number of NHB32-C interface units, Mainichi Broadcasting System (MBS) of Osaka, Japan digitally transfers audio via CobraNet™ as part of its relay broadcasting of golf, road races, and other sporting events. For example, at the Mizuno Classic, held in November 2007, the broadcaster used CobraNet™ to transfer no less than 96 channels of audio between an outside audio broadcasting van and a relay center.

Case Study #15

Surround & Processor

Le Cercle Rouge (TSF)
(Paris, France)

ShowMax Screen and DME64N

On its recent move to larger premises, the French cinema image and lighting-equipment hire group TSF built a large "no compromise" projection room with the aim of full compatibility with traditional film and high-definition video. The room – known as Le Cercle Rouge – features an 11.2-meter screen and a 16-kW ShowMax sound-reproduction system built around a Yamaha DME64N Digital Mixing Engine.

The TSF group, founded by Thierry de Ségonzac, is the European leader in its field, supplying cinema and television production companies with everything they need for production. In fact, seventeen of the films at this year's Cannes Film Festival were shot using TSF equipment.

When, early in 2007, TSF moved into a new, customized building, Thierry de Ségonzac decided to build a state-of-art projection room for 35-mm and 2K (soon to be 4K) projectors. In Le Cercle Rouge, de Ségonzac successfully achieved the "no compromise" room that he had been seeking. Located on the second floor of the TSF building, it features 126 Club Class seats and an 11.2-meter wide screen in 2.35:1 format. To obtain the best possible sound and image quality, de Ségonzac placed his trust in the ShowMax process approved by acoustics engineer Pierre Vincent. Image quality is already impressive in 2K digital cinema, and the even higher quality 4K format is just around the corner.

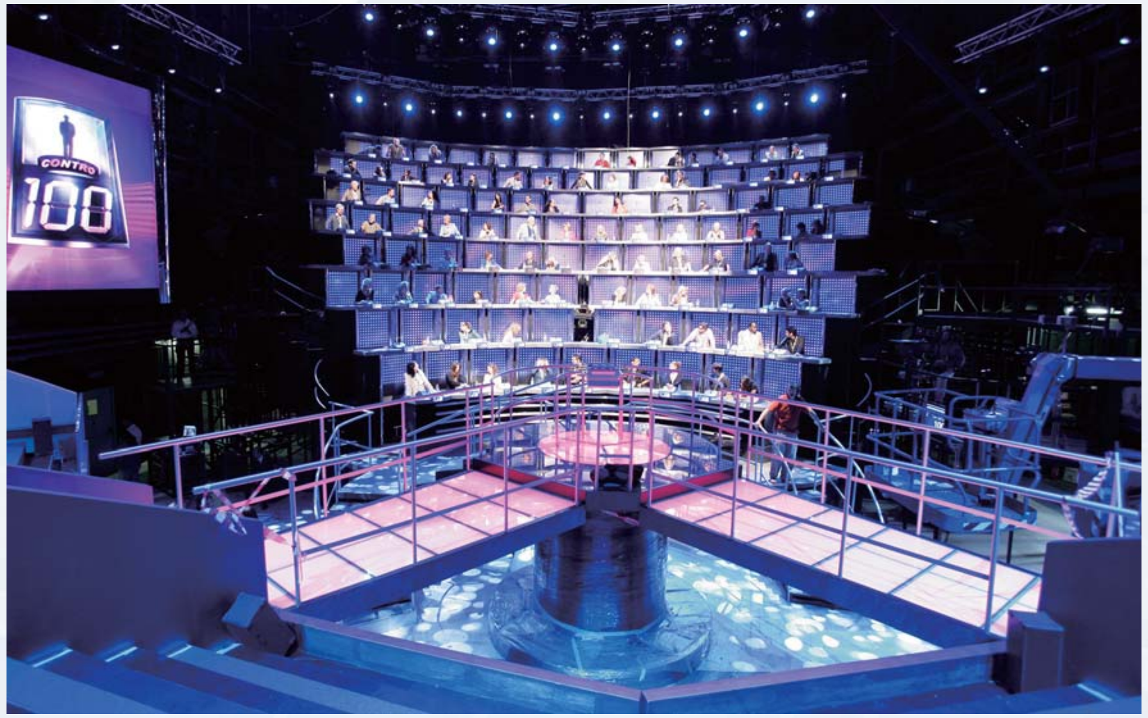
At the heart of the audio installation is a Yamaha DME64N processor operating at 24 bits/96 kHz to guarantee uncompromising audio quality. "In my first systems", explains Pierre Vincent, "I had used a different digital processor from a highly respectable manufacturer, but as soon as I changed to the DME64N, the improvement in audio quality was obvious. As far as I know, the DME is the only HD compliant digital audio processor available today that's capable of processing eight channels at 24/96 in accordance with the recommendations of the Digital Cinema Initiative (DCI)."

The DME64N at Le Cercle Rouge works continuously at more than 90% of its processing capacity. It is responsible for 32x32 grid functions, room compensation (X curve), and active three-way bass/mid-range/treble filtering for all the front channels and the delay (essential for digital cinema). The digital mixing engine is fitted with two digital input boards providing sixteen channels, and one of these is allocated to the HD projector's digital server. In addition, sixteen analog input channels are provided by two further boards, one of which receives the output from the Dolby Digital CP650 processor.

As the room must adapt to different sound-mixing standards (5.1, 6.1, etc.) and the ten surround enclosures must switch their sources accordingly, the DME64N also manages sixteen different outputs. This has been easily realized by storing the corresponding configurations within the DME, and a conference preset has even been setup for times when Le Cercle Rouge is used for this purpose. Furthermore, Vincent has also configured the DME such that the room's overall sound level can be controlled using the data wheel on the front panel.

As a result of this uncompromising approach, sound reproduction in the room is amazing – far better than that in many commercial cinemas. The bass seems to be unlimited, dialogue has a rare precision and intelligibility, and the spatialization is faultless. Even so, Pierre Vincent has more ideas for improvement: "I am currently testing Yamaha amps in the Tn range as they are said to be particularly suitable for this type of installation".

1 contro 100, Canale 5
(Milano, Italy)



Originally created by Dutch TV company Endemol, the television game show 1 vs. 100 is now screened successfully in all five continents. From Australia to the United States, Germany to Bulgaria, and Vietnam to Honk Hong, this quiz phenomenon is aired in no less than 22 countries worldwide. In the Italian version, 1 contro 100, which is produced by Mediaset television station Canale 5 and hosted by Amadeus, a lone challenger faces a wall of one hundred other contestants in a combat arena and must eliminate each and every one by correctly answering questions in order to win a 200,000 euro cash prize.

The game consists of a series of questions, each with a cash prize that gradually increases in value. The first to answer the questions are the "mob" of other contestants, and the lone contestant then takes his or her turn to answer, facing the challenge from a rostrum in the centre of the studio, right in front of the wall. If the contestant answers correctly, all members of the mob who got it wrong are eliminated.

In order to realize the audio system for this game, which above all must ensure faultless, rapid opening of over one hundred microphones, as well as excellent sound for the participants, Mediaset called in Backstage PA from the northern Italian town of Mariano Comense.

The PA company's Alberto Mariani explains, "The system was designed to meet the studio's strict scenographic require-ments and the need to use one hundred plus microphones, which are opened and closed depending on how the game develops. Nothing is preset – we know that only there will be groups of microphones that can vary at any time and that we must be able to pick out individual contestants in each group. We therefore decided to design the system around a single control centre."

The microphone signals from the wall of contestants are converted under the stage by a Yamaha AD824 – an eight-channel analogue/digital converter with high-quality 24-bit conversion and providing a 110-dB dynamic range. For connection with other devices such as digital mixing consoles and MTRs, the AD824 features a Mini-YGDAI card slot, and thanks to the variety of Mini-YGDAI cards currently available, support is available for all of the most-popular digital audio connection formats. In addition, the AD824's inputs have remote-controllable mic/line preamplifiers, and the unit can also be used as a stand-alone converter.

After A/D conversion, the signals are routed to the broadcaster's control room via a Yamaha NHB32-C Network Hub & Bridge, which features 32 I/O AES/EBU channels and uses CobraNet™ technology to transfer high-quality 24-bit/48-kHz digital audio in real time with a simple Cat5 cable link. As well as being able to handle up to 64 channels of digital audio, the NHB32-C can also be used with control data.

The system designed by Backstage takes the AES/EBU signals from the NHB32-C outputs and sends them to a pair of Yamaha DME64N Digital Mixing Engines, which group, equalize, and compress the microphone signals, before sending them on to the Mediaset play-out console.

In addition to this microphone-signal conversion, transmis-sion, and processing, another critical factor for the success of the game – particularly as far as the contestants are concerned – is sound reinforcement on the studio set. This role was entrusted to a series of four Yamaha XM4180 Multi-channel Power Amplifiers that – in addition to the enclosures used by Amadeus and the solo contestant – power all sixty-eight mini loudspeakers installed behind the players in the wall. The XM series combines lightweight, compact lines with high-quality performance and power, while the XM4180 amplifiers, which have been designed and manufactured for the most demanding live applications, provide four channels of 180 W on 8 ohms and ensure compatibility with various types of installation and sound control system.

Federico Farina of Backstage concludes, "Our software development department also designed a control interface to handle the job of opening and closing the microphones during the game and sending them to the broadcast control room on a single channel, as well as switching the luminous mic rings on the wall on and off. This interface enables us to form groups of microphones to be opened together, and also to store and recall the required scenes. Each individual microphone can be recalled in various ways – via mouse, touch screen or numerical keyboard, the last of which is the fastest, most practical method."

Backstage PA

A leading audio contractor for the television broadcast sector, Backstage recently celebrated twenty years in the industry. During this time, the company's partners – namely, Pino Di Costanzo, a veteran of the music show-production scene, and Daniele Mascheroni, a digital audio wizard – have not only set up steady long-lasting relationships with clients such as Canale 5, but have also participated in important projects for the Italian state broadcaster, RAI.

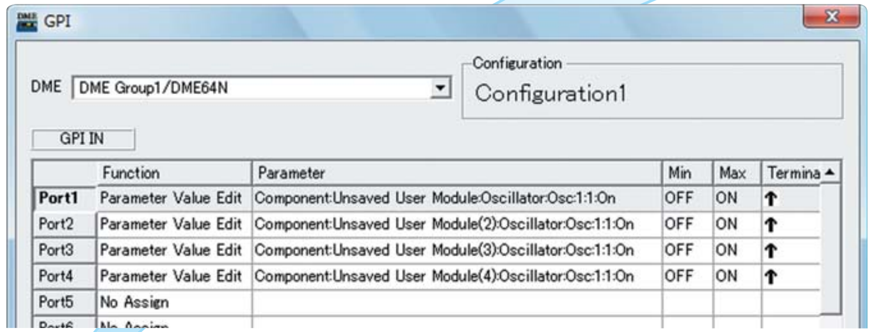
Since purchasing its first digital mixer in 1990, Backstage has had a constantly increasing stock of Yamaha products, as Mascheroni explains: "At present, we have nine PM1D digital mixing systems, five PM5D digital mixing consoles, and about twenty DM2000 digital production consoles, in addition to approximately fifty units from the Digital Mixing Engine series."

Di Costanzo adds, "As well as specializing in the audio aspects of television productions, we also have a mobile studio, designed primarily as a pre-control room for television productions, but also used for important live multi-track recordings and post-production work. Thanks to its Yamaha PM1D digital mixing system, the studio is ideal for this type of work."

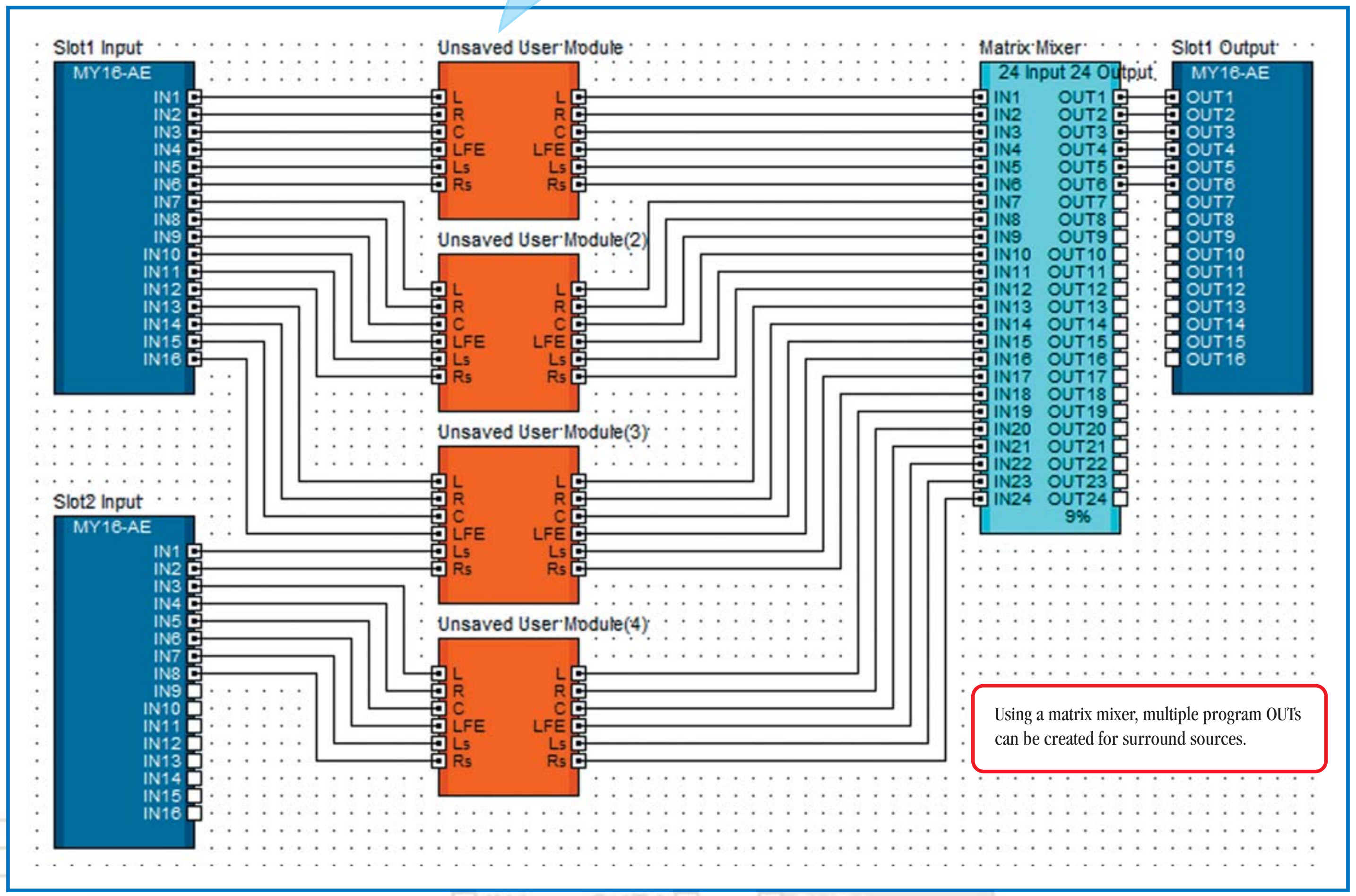
Audio source switching linked with active camera angles

Using DME New Componet "Program Ducker"

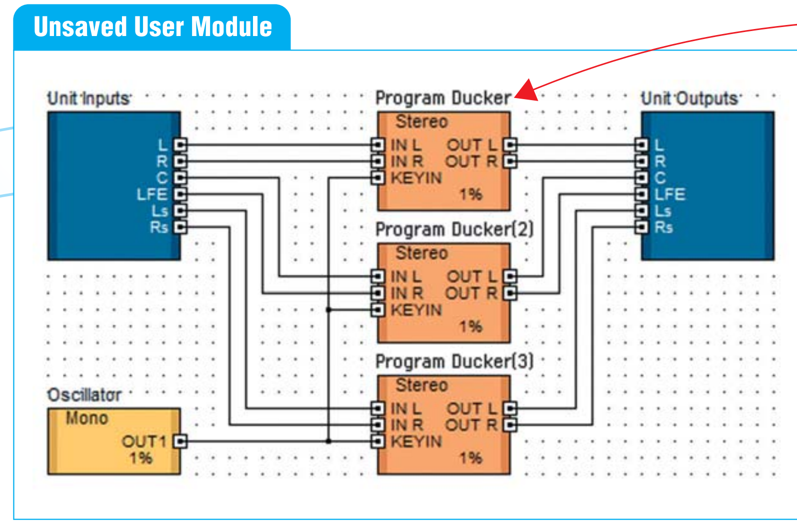
The new Program Ducker component features significantly longer attack and release times than the original Ducker component. Mono and stereo types can be flexibly configured to accommodate a wide range of applications.
The Program Ducker can be triggered from an oscillator. This makes it possible to use the output of a single oscillator to simultaneously trigger multiple channels for 5.1 surround, for example.
The new component "Program Ducker" is available on DME firmware version 3.5.



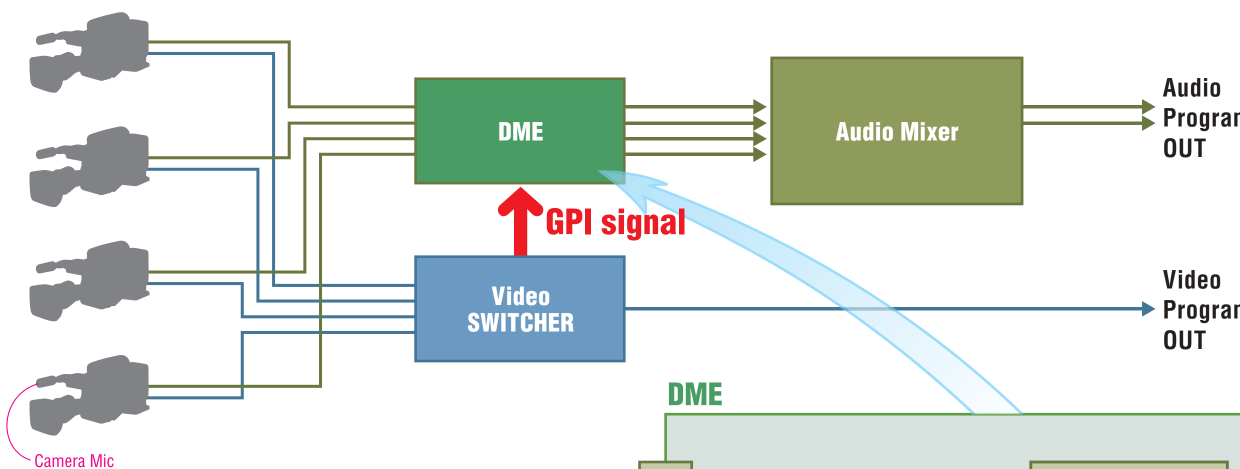
Assign the GPI outputs of the video switcher to DME's GPI inputs to control an oscillator by an external trigger. By distributing the oscillator output in parallel to the KEY IN inputs of multiple Program Duckers, it is possible to fade-in/ fade-out several audio channels at the same time.



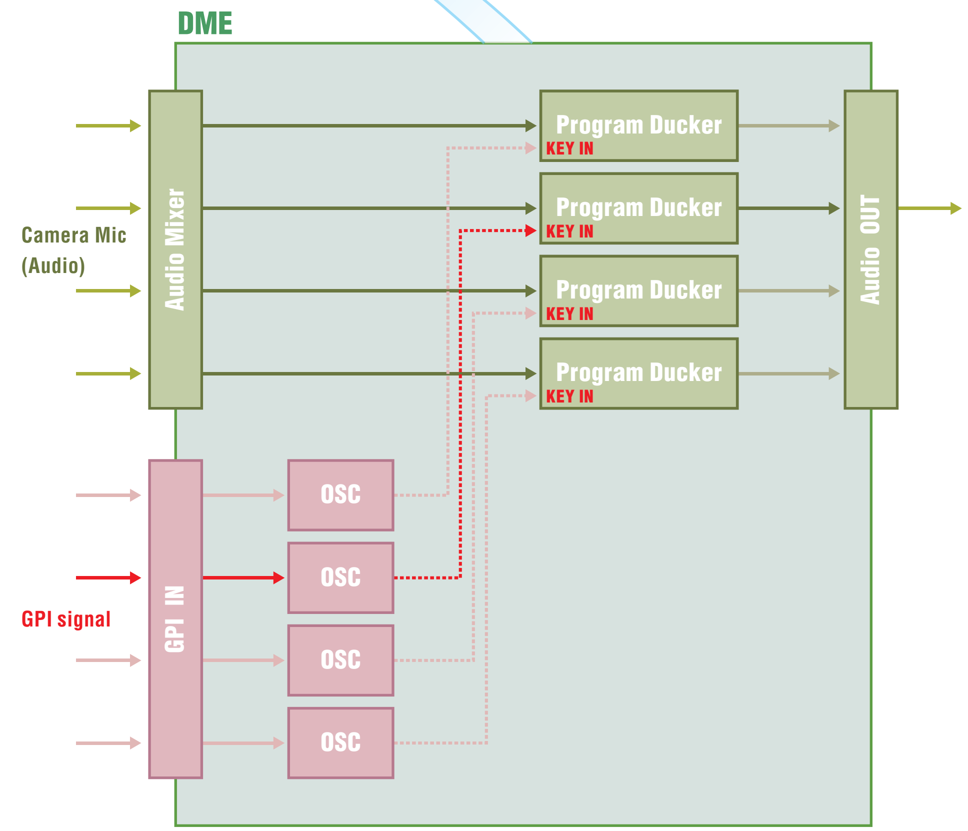
Using a matrix mixer, multiple program OUTs can be created for surround sources.



Example: Source level control using an oscillator



The flexibility of DME's new program ducker, allows you to create solutions for complex applications such as synchronized switching of audio source with video source in live sports broadcasts.
The program ducker uses an oscillator signal as its trigger, and multichannel signals such as stereo or dolby 5.1 may be controlled by a single oscillator.
This allows you to create a more flexible and cost efficient system than is possible using conventional dedicated equipment.



Turn to page 49 for information on how to download and install DME Designer.

Digital Mixing Engine DME series

Programmable, Networkable Mixing Engines for a Range of Audio Processing Applications

DME64N

96 MIDI DIGITAL MY16 3U

Processing power that is on a par with Yamaha's DM1000 Digital Mixing Console allows complex audio systems to be built around a single DME64N unit.

Rear Panel

DME24N

96 MIDI DIGITAL MY16 2U

Substantial processing power plus built-in analog I/O for fast, easy system implementation.

Rear Panel

DSP Power that Rivals Advanced Digital Mixers

The DME64N and DME24N employ Yamaha's original DSP6 and DSP7 signal-processing LSIs for extraordinary audio processing power and quality. The DME64N offers processing power that is on a par with Yamaha's DM1000 Digital Mixing Console, and the smaller DME24N has about half that processing capability while offering built-in analog I/O for simple, more compact systems. In either case you get a remarkable amount of processing capability that allows complex systems to be built around a single DME unit and fine tuned for optimum performance in concert halls, multi-purpose halls, event spaces, institutions, and a wide range of other applications. Plenty of DSP power also means that advanced configurations that previously required multiple hardware units can now run comfortably on just one DME64N or DME24N. All of this translates into significant time, energy, and cost savings for the design, installation, and operation of DME-based systems.

24-bit 96-kHz Processing and Circuitry Designed for Superior Sound

It's no secret that Yamaha digital mixing consoles are favored by leading professionals for live sound and studio recording applications. Yamaha DME mixing engines benefit from the same cutting-edge audio technology and attention to details that make a real difference in sound and performance. Both the DME64N and DME24N deliver faultless precision and reproduction fidelity with optimally-tuned 24-bit, 96-kHz digital processing, and the DME24N also features high-performance analog head amplifiers that equal the sound and quality of those found in top-line mixing consoles.

Note: Configurations created for 48-kHz operation will require half the processing power of configurations created for 96-kHz operation.

Exceptional I/O Flexibility and Expandability - Cascade Up to 8 DME64N Units for 512 Inputs and 512 Outputs

The DME64N has four rear-panel expansion slots that accommodate Mini-YGDAI I/O cards, and the DME24N has one expansion slot. Each slot supports up to 8 analog inputs or outputs when fitted with precision A/D or D/A cards, or up to 16 channels of digital I/O in AES/EBU, ADAT, or TASCAM format. With it's four expansion slots the DME64N can support up to 64 channels of I/O, while the DME24N has a single slot plus 8 channels of built-in analog I/O for a total of 24 channels. For even larger systems up to eight DME64N units can be cascaded to provided a massive 512 digital inputs and 512 digital outputs, or 128 analog inputs and 128 analog outputs.

Eight Analog Inputs and Outputs on the DME24N

The DME24N provides eight built-in analog inputs and outputs via Euroblock terminals on the rear panel. The analog inputs and outputs feature precision 24-bit, 96-kHz A/D and D/A converters that deliver top class audio performance.

Ethernet Control

A network of up to 16 DME64N, DME24N and ICP1 Intelligent Control Panel units can be interconnected via their RJ45 connectors and 100Base-T CAT5 type Ethernet cables. Ethernet connection makes it possible to set up a control network quickly and easily, and with minimum cost.

GPI, RS232C/RS422, USB, and MIDI Interfaces

Both the DME64N and DME24N offer a comprehensive selection of control interfaces for use with a wide variety of equipment. The DME64N has 16 GPI input and output terminals while the DME24N has eight to facilitate interfacing with other GPI-equipped devices. RS232C/RS422 ports allow direct connection to remote control units and computers, USB ports allow direct hookup with most modern computers, and MIDI terminals can be used for synchronization and control with musical instruments, sequencers, and lighting controllers.

Scene and Function Names in 5 Languages

The DME64N and DME24N, as well as the ICP1 Intelligent Control Panel, can display scene and function names in 5 languages: English, Japanese, French, German, and Spanish. All three models offer a user-friendly interface for smooth operation regardless of the user's level of skill or experience.

Audio I/O Distribution and DSP Expansion Units DME Satellite series

CobraNet™

DME Satellite Models for CobraNet Networking

CobraNet has become the choice for audio networking in complex, large-scale sound systems. Up to 64 channels of audio data can be carried via a single CAT5 Ethernet cable. CobraNet offers high reliability via its redundancy system with primary and secondary ports offered as standard.

DME8i-C 96 DIGITAL 1U

DME8o-C 96 DIGITAL 1U

DME4io-C 96 DIGITAL 1U

Rear Panel (DME8i-C)

Ether ES Sound

DME Satellite Models for EtherSound Networking

EtherSound offers extremely low latency, and has become choice paticularly for the temporary live applications. Up to 64 channels of audio data can be carried via a single CAT5 Ethernet cable so that the connection is easy and cost-effective.

DME8i-ES 96 DIGITAL 1U

DME8o-ES 96 DIGITAL 1U

DME4io-ES 96 DIGITAL 1U

Rear Panel (DME8i-ES)

- Vastly expand the capabilities and capacity of a DME-based sound system, or any other networked audio devices that use CobraNet™ or EtherSound™ protocol.*
- Controllable remote I/O plus powerful DSP processing capability allow distributed processing for unprecedented system design flexibility and power.
- Reduce system cabling costs while maximizing overall reliability.
- Also usable as stand-alone processors in smaller systems.
- Full 24-bit 96-kHz audio processing, plus the same highly-acclaimed analog circuitry used in the DME24N.
- Supplied DME Designer software application can be used to control, monitor, and create complete processing"configurations" in the same way as with the DME64N or DME24N.
- 8-in/4-out GPI terminals allows direct, easy connection to wall-mountable CP4SF control panels featuring four switches and four faders.

* CobraNet™ models have a "-C" suffix. EtherSound™ models have an "-ES" suffix.

Wall-Mount Remote Control Panels

ICP1

Intelligent Control Panel

The most sophisticated of the DME series remotes, the ICP1 connects via Ethernet. Functions include scene recall and six user-defined keys at the top and bottom of the LCD screen, which can be assigned to DME parameters such as microphone and music source levels. Up to 4 sets of "pages" are available - giving up to 24 parameters. LCD display shows names and scenes and function keys in five languages - English, German, French, Spanish and Japanese.

Maximum Connection of ICP1
In a DME system including one or more DME units and ICP1 units, up to 16 units can be connected in a Zone totally. For example, in a Zone, ICP1 can be used up to 15 units when one DME unit is used, or ICP1 can be used up to 14 units when two DME units are used.

CP4SF

Four switches and four faders control panel
Wall-mountable remote control panel for GPI control. Uses a US-type 3 gang wall box.

CP4SW

Four switches control panel
Wall-mountable remote control panel for GPI control. Uses a US-type 1 gang wall box.

CP1SF

One switch and one fader control panel
Wall-mountable remote control panel for GPI control. Uses a US-type 1 gang wall box.

Please refer the column "GPI on DME units" below about connection number for control panels including CS4SF, CP4SW and CP1SF.

Note: Use a standard (US-type) wall box: 3-gang with depth 44mm for ICP1 and CP4SF, 1-gang with depth 44mm for CP4SW and CP1SF. It is necessary to use the included frame plate to install these remote control panels in standard wall boxes.

GPI on DME units

What is GPI

GPI stands for General Purpose Interface. This is external control interface for digital equipment and allows creating simple and affordable control system. DME64N has 16 GPI input and output terminals (Euroblock) while the DME24N has 8 input and output GPI terminal (Euroblock). And DME Satellite has 8 input and 4 output GPI terminals (Euroblock).

Simple and low cost

Structure of simple and low cost control system is one of the biggest advantages of GPI using only DC 5 volts and resistor without requiring complicated parts and programming. Only preparing general switches and faders compatible to 5 volts are necessary to make original controller for meeting installation facility size.

Parameter Control via GPI

GPI can handle on/off type switch and fader type control. Typical usage by switch is changing on/off and scene (program) change, and usage by fader is master volume control and etc. Any parameter required to be controlled via GPI can be assigned on "DME Designer" software and controlled externally by CP series control panel of DME options as well as any GPI devices.

Connection number of GPI device

DME64N has 16 input and output, DME24N has 8 input and output, and DME Satellite has 8 input and 4 output GPI terminals. GPI devices on option control panel including switches, faders, and LEDs can be connected in total 16, 8, and 12 respectively for DME64N, DME24N and DME Satellite.

	GPIOutput	GPI Input
CP4SF	8	4
CP4SW	4	4
CP1SF	2	1

Remote control capability up to 200 meter*

There is a report that cable for GPI could be extended up to 200 meters by using shielded cable proofed electrical interfere noise. This offers big advantage for designing installation of sound system.
(* Distance may vary depending on cable performance and environmental conditions.)

16-channel DA Converter and Monitoring / Control Unit

ACU16-C

CobraNet MIDI DIGITAL 1U

Rear Panel

High-performance 16-channel D/A converter and monitoring/control unit.

- The ACU16-C converts up to 16 channels of digital audio (48 kHz, 20-bit or 24-bit) to analog using CobraNet technology, and delivers analog audio to Tn series and PC-1N series power amplifiers with a dynamic range of 110dB.
- A single ACU16-C unit can control up to 32 amplifiers.
- Amplifier settings and status can be monitored and controlled using the supplied NetworkAmp Manager II software and a computer: power on/standby, attenuation, out/in levels, temperature, load impedance, and more.
- A computer can be connected to any NHB32-C or ACU16-C in the network to provide full monitoring and control capability for all amplifiers on the CobraNet network.
- Automatic troubleshooting and logging.
- Redundant connections for exceptional reliability.

32 IN / 32 OUT Channel Audio and Control Interface

NHB32-C

CobraNet MIDI DIGITAL 1U

Rear Panel

32-in/32-out audio and control interface sends and receives digital audio and control signals via CobraNet.

- A single NHB32-C unit can support up to 32-channels of digital audio I/O, and one control signal I/O.
- Four separate 8-channel AES/EBU input/output terminals.
- CobraNet™ transfers high-quality 20-bit or 24-bit 48 kHz digital audio over four bundle paths in real time.
- MIDI, head amp remote (AD8HR, AD824) and amplifier control.
- Network Amp Manager II software allows central control of both audio and control signals.
- Up to eight NHB32-C units can be networked, with each unit transferring and receiving four bundles, and 64 channels of digital audio can be received from any ACU16-C or NHB32-C on the network via CobraNet™.
- A networked system with ACU16-C and NHB32-C units is capable of handling up to 64 channels of digital audio.
- Multicast and unicast modes can be used in combination to optimize bandwidth.
- Primary and secondary CobraNet™ ports are built-in, providing a redundant network for reliable operation.

Network Audio Interface

NAI48-ES

EtherSound 100 96 DIGITAL 1U

Rear Panel

Connect Yamaha digital mixing consoles or DME processors to remote head amps or AD/DA converters for analog interfacing wherever needed.

- Transfer up to 48 channels of digital audio between console and stage or any other location via a single Cat-5 Ethernet cable at sampling rates up to 96 kHz.
- Eliminates noise problems that are unavoidable with conventional analog gear and multi-cable type setups, and dramatically reduces analog cable length for superior sound quality.
- 48 AES/EBU inputs and 48 AES/EBU outputs are provided via 25-pin D-sub connectors.
- Designed with Yamaha professional sound gear in mind: minimum setup hassle, trouble-free operation.
- An external power supply can be used in addition to the internal power supply for redundant failsafe operation.
- Use standard Ethernet hubs and routers to create any network configuration you need.

AD Converter with Remote Preamplifier

AD8HR

96 DIGITAL 1U

Rear Panel

Remotely controllable 8-channel AD converter and preamplifier with 96 kHz processing.

- Microphone preamplifier technology inherited from the PM5000 analog live sound console for unsurpassed sound quality.
- Microphone preamplifier gain can be remotely controlled in steps of 1 dB from compatible Yamaha digital mixing consoles.
- High-pass filter with remotely controllable cutoff frequency on each channel.
- Remotely switchable phantom power supply.
- Up to 255 AD8HRs can be daisy-chained for massive input capacity.
- AES/EBU digital connection to digital mixing console minimizes the need for analog cabling.
- Eight XLR connectors and D-Sub AES/EBU terminals in a compact 1U design.
- Dual output connectors enable 2 x 8-channel digital audio output in the AES/EBU format.
- Remote control can be implemented via RS422 or switchable PC/RS422 nine-pin terminals.

EtherSound Compliant 16in/8out Stage box

SB168-ES

EtherSound 100 3U

Rear Panel

Versatile 16-in/8-out EtherSound stage box for various Yamaha digital mixers.

- An affordable 3U-size stage box that utilizes reliable, low-latency EtherSound technology for digital audio signal transmission.
- 16 channels of sonically superb remote analog input — each with its own mic/line head amp — plus 8 channels of analog output.
- Audio can be transferred over distances up to 100 meters via standard CAT5e Ethernet cables (maximum distance may depend on cable performance).
- The SB168-ES can be used as a general-purpose analog-EtherSound I/O box.
- Handles uncompressed 24-bit audio at 44.1 kHz and 48 kHz sampling rates.
- Internal head amplifier gain and +48V phantom power switching can be remotely controlled from a compatible digital mixing console or from the AuviTran™ AVS-ESMonitor software.
- Up to four SB168-ES units can be linked to provide a total of 64 inputs and 32 outputs (maximum number may depend on the mixing console used).
- An ideal choice for use with popular digital consoles such as the Yamaha PM5D, M7CL or LS9.
- Compared to conventional analog console + analog multi-core systems the SB168-ES provides exceptionally high noise resistance and makes it possible to keep microphone cables short for optimum signal quality.
- Easy set-up reduces the time, effort, and cost of installation.

Yamaha Mini-YGDAI cards

Each expansion slot – DME64N, DME24N and Yamaha Digital Mixers – can be used to add up to 16 analog or digital I/O channels in a variety of formats by simply plugging in the appropriate mini-YGDAI expansion card including CobraNet and EtherSound audio network interface cards.

Digital I/O Series

Digital Network Cards

MY16-CII
16-Channel Audio CobraNet format I/O and Control I/O

MY16-ES64
16-Channel EtherSound format I/O and Control I/O

MY16-MD64
16-Channel Audio MADI format I/O

MY16-EX
16-Channel Expansion Card for MY16-ES64 and MY16-MD64

AES/EBU Format

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

MY8-AE96S
8-Channel AES/EBU format I/O (with sample rate converter)

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

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

MY8-AEB
8-Channel AES 3id-1995 format I/O

ADAT Format

MY16-AT
16-Channel ADAT format I/O

MY8-AT
8-Channel ADAT format I/O

TDIF Format

MY16-TD
16-Channel TDIF format I/O

MY8-TD
8-Channel TDIF format I/O

HD-SDI

MY8-SDI-D NEW
8-Channel Demultiplexer Card

Analog I/O Series

AD/DA Card

MY8-ADDA96
8-Channel Analog Input/Output Card

AD Cards

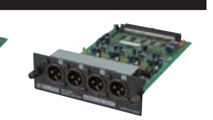
MY8-AD96
8-Channel Analog Input Card

MY8-AD24
8-Channel Analog Input Card

MY4-AD
4-Channel Analog Input Card

DA Cards

MY8-DA96
8-Channel Analog Output Card

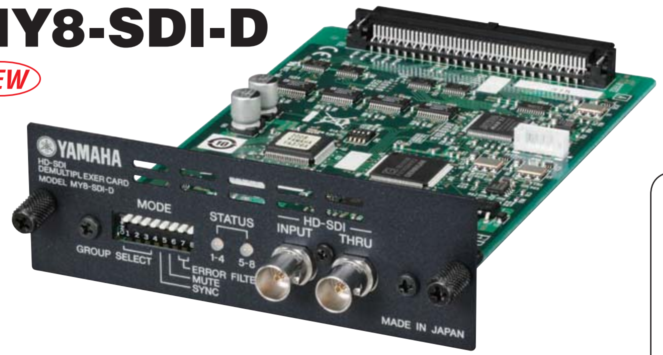


MY4-DA
4-Channel Analog Output Card

HD-SDI Demultiplexer Card

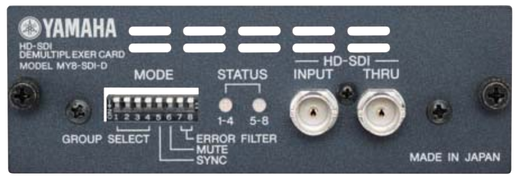
MY8-SDI-D

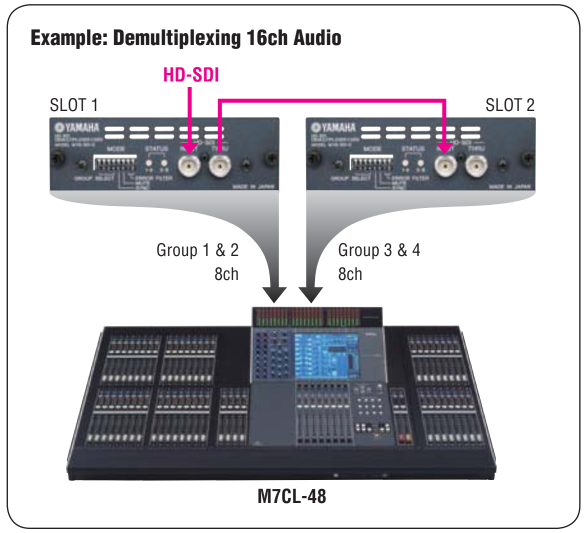
NEW



An Interface Card that Brings HD-SDI Embedded Audio Signals to Yamaha Digital Consoles

- 1 x HD-SDI input, 1 x thru-output (non-reclocked).
- Group selection allows demultiplexing of two 4-channel embedded audio groups.
- Switchable audio synchronization mode: sync with embedded audio or video signals.
- Embedded audio status display.





Specifications

HD-SDI INPUT CHARACTERISTICS

SDI Input Signal	SMPTE 292M (1.485 Gbps, 1.485 / 1.001 Gbps)
Embedded Audio	SMPTE 292M
Resolution	24 bit
Fs	48 kHz
Input Impedance	75 Ω
Connector	BNC x 1

DIGITAL AUDIO CHARACTERISTICS

Resolution	24 bit
Fs	48 kHz
Lines	4 lines (8 channels), through group selection

HD-SDI THRU OUTPUT CHARACTERISTICS

SDI Output Signal	Non-reclocked thru output
Output Impedance	75 Ω
Connector	BNC x 1

COMPATIBLE FORMAT

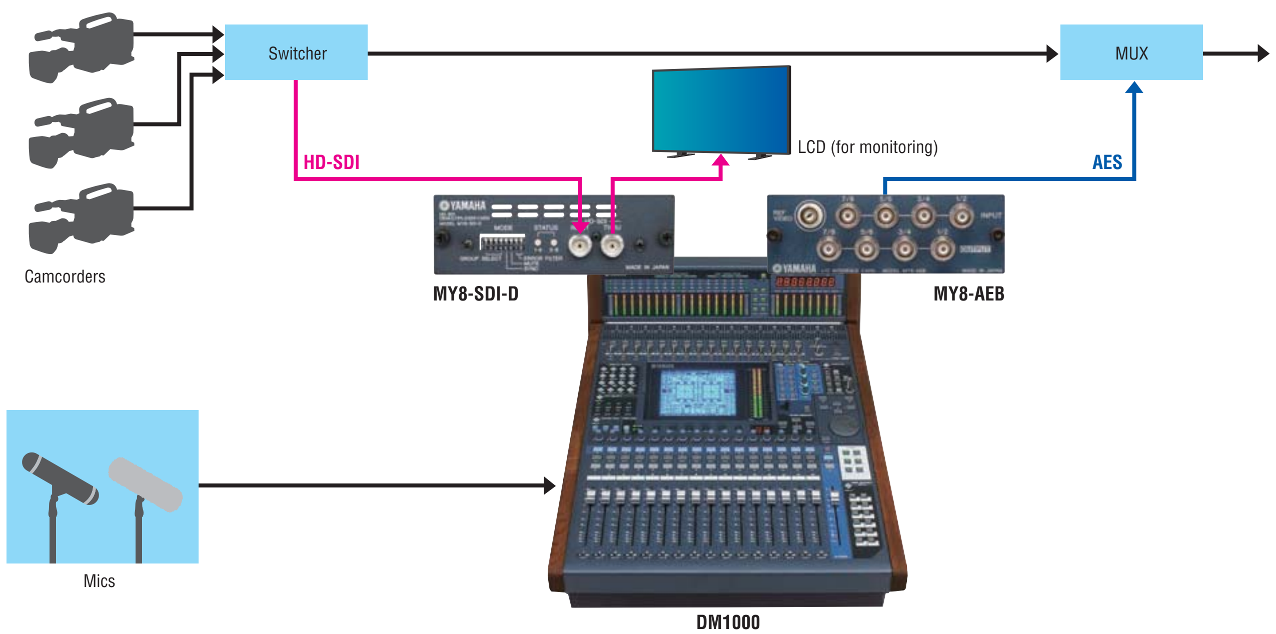
720p 50	720p 59.94	720p 60		
1035i 59.94	1035i 60			
1080i 50	1080i 59.94	1080i 60		
1080p 23.976	1080p 24	1080p 25	1080p 29.97	1080p 30
1080sF 23.976	1080sF 24			

COMPATIBLE HOST MODELS

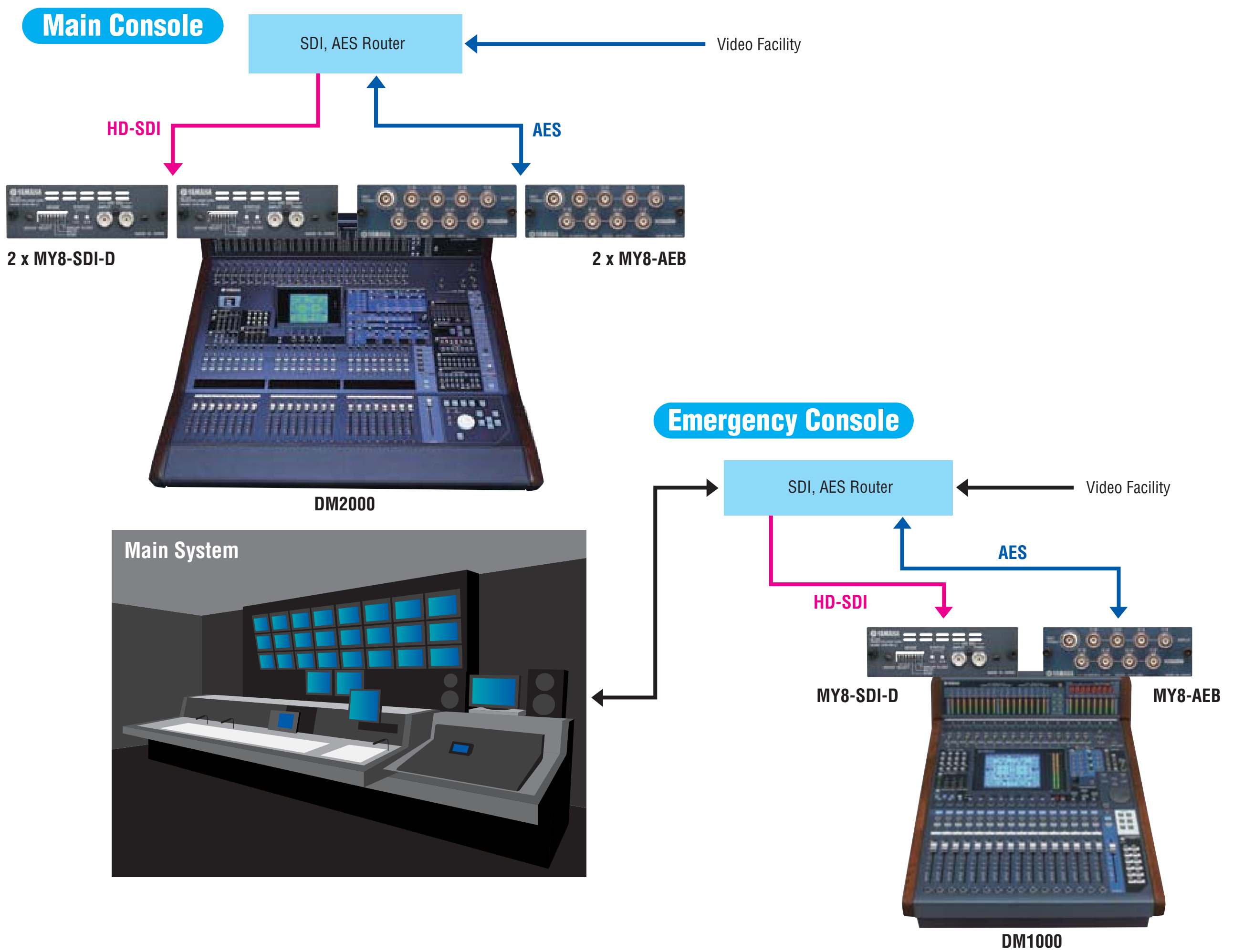
Model	Max No. of Cards	Total No. of Input Channels
DM2000	6	48
02R96	4	32
DM1000	2	16
01V96	1	8
PM5D	4	32
DSP5D	2	16

Model	Max No. of Cards	Total No. of Input Channels
DME64N	4	32
DME24N	1	8
M7CL	3	24
LS9-16	1	8
LS9-32	2	16

Use Case 1 : Live Broadcasting



Use Case 2 : Console in Broadcast Station



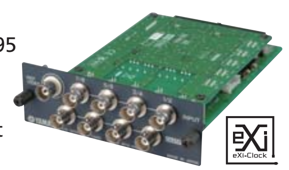
I/O Interface Card

MY8-AEB

8 channel AES/EBU format I/O (w/REF VIDEO Input)

Main Features

- 8-in/8-out AES-3id-1995 standard with 75Ω BNC connectors.
- Reference video input connector.
- eXi-Clock synchronization maintenance function.



8-channel Digital Audio I/O with "eXi-Clock" Sync Capability

The MY8-AEB is one of Yamaha's MY-series plug-in interface cards, providing expanded I/O functionality for compatible Yamaha digital audio products. Specifically, the MY8-AEB provides 8 channels of AES-3id-1995 standard (75Ω BNC connectors) digital audio input and output with advanced video synchronization capability.

In addition to 8 BNC connectors – 4 stereo inputs and 4 stereo outputs – the MY8-AEB has a REF VIDEO input that can be used to receive a PAL or NTSC format Reference video (Black Burst) signal that will provide the synchronization word clock for the device into which the card is installed. For maximum system reliability the MY8-AEB features a unique Yamaha **"eXi-Clock"** function that is capable of seamlessly continuing word clock generation even if the Reference video signal is interrupted.

As shown in the system diagram below, the MY8-AEB makes it easy and convenient to use high-performance Yamaha digital mixers for video post production, making it possible to maintain the highest audio quality with reliable synchronization to a wide range of video sources.

Specifications

I/O	Input x 8, Output x 8, REF VIDEO In x 1
Connector	BNC (Input x 4/Output x 4) based on AES-3id
Fs	48 kHz, 44.1 kHz (Pull up/down 0.1%, 4%)

Digital Mixing Console

PM5D Version2
PM5D-RH Version2

PM5D Version2

PM5D-RH Version2

The PM5D and PM5D-RH Digital Mixing Consoles take the digital revolution to the next level.

- The PM5D features standard high-performance head amps, while the PM5D-RH adds head-amp recall capability that allows head amp gain settings to be recalled along with the other console scene data.
- 48 mono and 4 stereo inputs, 24 mix buses and 2 stereo outputs, and 8 matrix outputs (expandable).
- Custom "DSP7" LSI for ultra-high-speed 96-kHz/32-bit processing.
- I/O capacity and functionality can be doubled or tripled by adding one or two rack-mountable DSP5D Digital Mixing Systems.
- Easy "virtual soundcheck" with individually assignable channels does not require complex re-patching.
- Built-in VCM (Virtual Circuit Modeling) effects offer impeccable simulations of classic signal processing gear.
- 8 high-performance multi-effect processors and 12 graphic equalizers built in.
- Enhanced security features keep the system operating flawlessly in any application.

Power Supply Unit

PW800W

Rear Panel

The PM5D is reliably powered by an external power supply unit. The PW800W is extremely compact and lightweight (3U, 10kg). Thanks to its high efficiency, the low speed cooling fans are extremely quiet. Two PW800W units can be serially connected using optional PSL120 cable for failsafe operation. PW800W accepts 100 - 240 volts so it can be used anywhere.

Digital Mixing System

DSP5D

DIGITAL MY16 11U

Rear Panel

The DSP5D provides essentially all of the functionality of a PM5D-RH minus the control surface, in a rack-mountable unit that can be seamlessly controlled from a PM5D (RH) console.

- 48 microphone/line and four stereo inputs.
- 24 mix buses that can be cascade-connected to a PM5D V2 console.
- Comprehensive control from a computer running the DSP5D Editor.
- Remote control from a PM5D V2 via the DCU5D Digital Cabling Unit.

Digital Cabling Unit

DCU5D

1U

Rear Panel

The DCU5D lets you locate the DSP5D as far as 120 meters from the PM5D console, connected by a single CAT5e Ethernet cable.

DSP5D and DCU5D utilize EtherSound technology, but because Bandwidth is fixed exclusively for DSP5D and DCU5D, they cannot be controlled by ES (EtherSound) monitor.

** Maximum cable length will depend on the quality and performance of the cables used.*

Digital Mixing Console

M7CL Version2

Centralogic™ MY16

M7CL-32 Version2

M7CL-48 Version2

Centralogic interface delivers digital live sound mixing with the comfort and efficiency of analog.

- Yamaha Centralogic™ interface with touch-screen allows total control from an easily accessible central area.
- The M7CL-48 provides a total of 56 inputs – 48 mono mic/line inputs and 4 stereo line inputs. The M7CL-32 has a total of 40 inputs – 32 mono mic/line inputs and 4 stereo line inputs.
- 27 buses in the form of 16 mix buses, an LCR bus, and eight matrixes that can be used with inputs as well as buses.
- Straightforward connection via analog input connectors for every input channel.
- 16 analog "omni" outputs as well as a 2TR digital output.
- Three Mini-YGDAI I/O card slots can accommodate up to 16 channels of digital or analog I/O each, for up to an addition 48 I/O channels.
- Eight DCA groups and eight mute groups.
- Virtual 8-unit rack for effects and graphic EQ.
- Advanced access management functions provide multi-level control over user access.
- Recallable right down to head amp gain, plus safe and focus functions.
- M7CL Editor software application supplied.

Power Supply Unit

PW800W

Rear Panel

For many applications you can simply plug the M7CL directly into a convenient AC outlet and use the built-in power supply. But you should consider the external PW800W Power Supply Unit in situations where maximum regulation and reliability are required. When a PW800W unit is added the internal power supply and the PW800W provide redundant failsafe operation.

Version2 New features:
- Global Paste, which allows simultaneous editing of multiple scenes
- Matrix Sends on Fader feature to give access to 24 mix buses
- Channel Library that stores channel parameter settings such as dynamics and EQ …and more.

Digital Mixing Console

LS9

MY16

LS9-32

LS9-16

Lightweight, compact, all-in-one digital mixers with advanced features and outstanding sound quality.

- 16 or 32 mono mic/line input channels plus 4 stereo input channels, expandable up to 32 or 64 channels in two layers.
- 16 mix buses, 8 matrix buses, plus stereo and mono buses with LCR mode.
- Top-performance analog mic/line preamplifiers.
- Compact and light enough for one person to move and set up easily.
- Yamaha Selected Channel interface allows smooth, intuitive access to detailed channel functions via a color LCD display and logically arranged encoders.
- Extensive gating, compression, and equalization facilities.
- Built-in USB memory recorder/player for recording or BGM playback.
- Virtual Rack packed with effects and EQ for just about any processing requirements.
- Scene memory for instant store and recall of all console parameters, including head amp gain.
- Advanced access management includes user keys (standard USB memory devices) that can be issued with different access levels for different users.
- LS9 Editor Software for enhanced operability and programmability.

Digital Production Console

DM2000VCM

DIGITAL MY16

The top choice for professional audio production, now with advanced features for audio production, broadcast, and live applications.

- Precise 24-bit/96-kHz audio and high-performance head amps.
- 96-input 24-bus mix capacity at 96 kHz.
- Powerful channel functions with flexible control and digital patching capability.
- Eight advanced multi-effect processors plus six 31-band GEQs.
- Scene memory and auto-mix functions for efficient workflow.
- Versatile channel pairing and grouping functions enhance mixing efficiency.
- Comprehensive interface with touch-sensitive 100-mm motor faders.
- Six mini-YGDAI expansion slot for easy I/O expansion in a variety of formats.
- Advanced studio manager software supplied.
- Easy integration with computer-based DAWs (Digital Audio Workstations) or digital recorders to create an advanced digital production environment.
- A comprehensive range of features for surround production, including an enhanced surround monitoring environment with bass management.
- A new dimension of production power with the addition of Yamaha VCM effects and processing.
- Included in the THX pm3 TM Studio Certification Program Approved Equipment List.

Digital Mixing Console

02R96VCM

DIGITAL MY16

A comprehensive update of the legendary 02R.

- Precise 24-bit/96-kHz audio and high-performance head amps.
- Generous mixing capacity with up to 56 simultaneous inputs and 20 mix buses in the same compact desk-top dimensions as the original 02R.
- Powerful channel functions with flexible control and digital patching.
- Four advanced multi-effect processors include surround effects.
- Scene memory and auto-mix functions for efficient workflow.
- Versatile channel pairing and grouping functions enhance mixing efficiency.
- Comprehensive interface with 25 touch-sensitive 100-mm motor faders.
- 16 microphone/line inputs with balanced XLR/TRS jacks that feature top-performance head amplifiers for outstanding audio quality.
- Four mini-YGDAI expansion slots for easy I/O expansion in a variety of formats.
- Advanced Studio Manager application for Windows or Macintosh computers supplied.
- A new dimension of production power with the addition of Yamaha VCM effects and processing.
- Included in the THX pm3 TM Studio Certification Program Approved Equipment List.

Digital Production Console

DM1000VCM

DIGITAL MY16 DM1000 13U MB1000 3U

19-inch Digital Production Console for professionals in the studio or on the road.

- Precise 24-bit/96-kHz audio and high-performance head amps.
- Generous mixing capacity with up to 48 simultaneous inputs and 20 mix buses in a compact rack-mount size console.
- Powerful channel functions with flexible control and digital patching capability.
- Four advanced multi-effect processors include surround effects.
- Scene memory and auto-mix functions for efficient workflow.
- Versatile pairing and grouping functions enhance mixing efficiency.
- Comprehensive interface with 17 touch-sensitive 100-mm motor faders.
- A generous selection of control interfaces: MIDI, USB, REMOTE, SMPTE, and word clock.
- Two mini-YGDAI expansion slots for easy I/O expansion in a variety of formats.
- Advanced Studio Manager application for Windows or Macintosh computers supplied.
- Easy integration with computer-based DAWs (Digital Audio Workstations) or digital recorders to create an advanced digital production environment.
- A comprehensive range of features for surround production, including an enhanced surround monitoring environment with bass management.
- A new dimension of production power with the addition of Yamaha VCM effects and processing.
- Included in the THX pm3 TM Studio Certification Program Approved Equipment List.

Virtual Circuitry Modeling

VCM technology is responsible for the classic compressor, EQ, analog tape deck, and stompbox effect simulations in the DM2000VCM, 02R96VCM and DM1000VCM.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, while actually creating ideal examples of sought-after vintage gear.

Powered Near-field Monitor MSP STUDIO series

MSP7 STUDIO

Powered Monitor Speaker

- 2-way bas-reflex bi-amplified near field studio monitor 6.5" cone woofer and 1" titanium dome high-frequency unit delivers 45Hz- 40kHz frequency response.
- 130 watts (LF 80W + HF 50W) dynamic bi-amplified power.
- XLR balanced input.
- Advanced Magnetic Structure Design
- One-piece Molded Enclosure with Rounded Baffle
- 31 positions Level Control facilitates precise overall system level matching.
- Low Cut switch and TRIM Control (High/Low).
- Full magnetic shielding.

MSP5 STUDIO

Powered Monitor Speaker

- 2-way bas-reflex bi-amplified near field studio monitor 5" cone woofer and 1" titanium dome high-frequency unit delivers 50Hz- 40kHz frequency response.
- 67 watts (LF 40W + HF 27W) dynamic bi-amplified power.
- XLR balanced input and 1/4" unbalanced input.
- Advanced Magnetic Structure Design
- One-piece Molded Enclosure with Rounded Baffle 31 positions Level Control facilitates precise overall system level matching.
- TRIM Control(High/Low).
- Full magnetic shielding.

SW10 STUDIO

Powered Subwoofer

- 180 watts dynamic power.
- XLR balanced inputs (L/R/SUBWOOFER) .
- XLR balanced outputs(L/R/SUBWOOFER) parallel connection with input signals.
- Level control facilitates precise system level controls.
- 40-120 Hz, 80 Hz at Center Click LPF controls.
- Phase switch simplifies phase alignment.
- Full magnetic shielding.

Powered Near-field Monitor HS series

HS80M

Powered Monitor Speaker

- 2-way bass-reflex bi-amplified near-field studio monitor with 8" cone woofer and 1" dome high-frequency unit delivers 42 Hz – 20 kHz frequency response.
- 70 watts dynamic bi-amplified power.
- XLR and TRS phone jack inputs accept balanced or unbalanced signals.
- Large Magnets in an Advanced Magnetic Circuit Design.
- Level control facilitates precise overall system level matching.
- MID EQ, ROOM CONTROL, and HIGH TRIM response control switches.
- LOW CUT switch.
- Full magnetic shielding.

HS50M

Powered Monitor Speaker

- 2-way bass-reflex bi-amplified near-field studio monitor with 5" cone woofer and 3/4" dome high-frequency unit delivers 55 Hz – 20 kHz frequency response.
- 120 watts dynamic bi-amplified power.
- XLR and TRS phone jack inputs accept balanced or unbalanced signals.
- Large Magnets in an Advanced Magnetic Circuit Design.
- Level control facilitates precise overall system level matching.
- MID EQ, ROOM CONTROL, and HIGH TRIM response control switches.
- LOW CUT switch.
- Full magnetic shielding.

HS10W

Powered Subwoofer

- 8" bass-reflex powered subwoofer delivers solid 30 Hz – 120 Hz frequency response.
- 150 watts dynamic power.
- XLR and TRS phone jack inputs accept balanced or unbalanced signals.
- Balanced XLR L and R outputs connect to the main left and right speakers. L/R mix output connects to a second subwoofer if required.
- Level control facilitates precise overall system level matching.
- Phase switch simplifies phase alignment.
- Low-pass filter control and high-pass filter control with ON/OFF switch.
- Full magnetic shielding.

ADD-ON EFFECTS

The Yamaha ADD-ON EFFECTS series is a range of unique and valuable plug-in effects specially designed and provided in the 96kHz Audio DSP of the Yamaha Digital Consoles. You can freely assign these new plug-in effects in the same way as the current internal effect programs.
In order to use the ADD-ON EFFECTS, you will need to have the Version 2 system software installed in your digital console.

Channel Strip Package (AE-011)

Preinstalled PM1DV2 / PM5DV2 DM2000VCM / 02R96VCM / DM1000VCM

Virtual Circuitry Modeling — Developed by K's Lab

The AE-011 Channel Strip 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. Not only do these models faithfully capture the unique saturation of analog circuitry, 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 276S (stereo)

Compressor 260S (stereo)

Equalizer 601

Master Strip Package (AE-021)

Preinstalled PM1DV2 / PM5DV2 DM2000VCM / 02R96VCM / DM1000VCM

Virtual Circuitry Modeling — Developed by K's Lab

The AE-021 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. Because of their ability to smooth out peak levels and tidy up the response, many high-end recording studios still maintain openreel recorders such as the Studer A80mk1, A80mk4 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. 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. Now you can easily take advantage of top-end sound-shaping techniques in real time using Yamaha digital consoles.

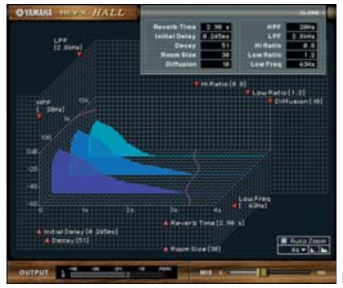
Open Deck

Reverb Package (AE-031)

Preinstalled PM1DV2 / PM5DV2 / M7CL / LS9 / DM2000VCM / 02R96VCM / DM1000VCM

REV-X

These reverb Add-on Effects employ 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. 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.



REV-X Hall

Vintage Stomp Package (AE-051)

Preinstalled DM2000VCM / 02R96VCM / DM1000VCM

Virtual Circuitry Modeling — Developed by K's Lab

The Surround post package takes full advantage of Yamaha's iSSP (The Vintage Stomp Pack utilizes VCM (Virtual Circuit Modeling) Technology to emulate a number of vintage guitar stomp box effects that are highly valued for their rich, warm sound.

MAX100

Dual Phase

Vintage Phaser

Surround Post Package (AE-041)

Preinstalled DM2000VCM / 02R96VCM / DM1000VCM

iSSP

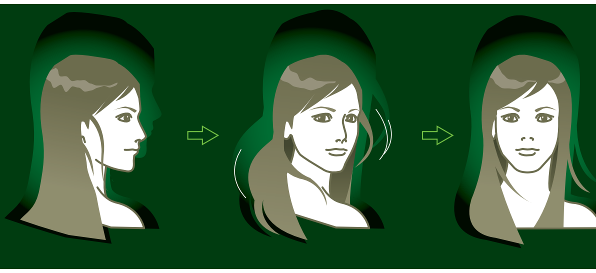
The Surround post package takes full advantage of Yamaha's iSSP(Interactive Spatial Sound Processing) technology to deliver precision controlled spatial processing capabilities well suited for cinema or television sound post-production and mixing facilities. Effects are compatible with a range of surround formats and deliver unprec-edented precision in matching visual motion with sound, and vast creative control in creating incredible sonic environments.

Room-ER

Ideal for placing a mono source in a precisely controllable surround environment, this effect simulates the acoustic properties of a 30-meter long room and accurately reproduces the direct sound and early reflections as affected by distance from the source, source motion, motion speed, and room surface characteristics.

■ 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

Besides recreating the effect of objects moving past the listener in a linear path, like the siren of an emergency vehicle as it draws near and passes by, Auto Doppler can move objects toward and then away from the listener 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

This effect rotates the percspective of entire 5.1 surround fields with either the listener at the center of rotation, or the listener rotating or being moved around a sound source. Parameters such as rotation axis, movement amount, distance from the center of rotation, and speed can be specified and controlled automatically or with a joystick like the one on the DM2000VCM console.

■ 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.

ADD-ON EFFECTS

An Interview with the Creators of Yamaha's Add-on Effects

Part 1: The VCM Compressor, Equalizer, Open Deck, and Stomp Box Effects

Yamaha's Add-on Effects are marvels of digital signal processing design and engineering. The VCM (Virtual Circuitry Modeling) effects discussed in part one of this interview deliver the sonic advantages of the finest analog vintage gear with digital convenience and control, without the noise and reliability issues that plague traditional outboard sound gear. Although initially made available only as add-on software packages for compatible Yamaha digital mixers, they have become "essential equipment" for a great many professionals in both the audio production and live sound fields, and are now included in varying combinations in Yamaha's latest VCM-series digital consoles.

We were very fortunate to have an opportunity to sit down with the creators of these remarkable effects packages and spend a couple of hours discussing how the Add-on Effects came about, and the impact they are having on the pro audio scene.

Channel Strip
Vintage analog compression and EQ brought to life in the digital domain through innovative VCM technology. Includes COMP276, COMP260, and EQ601 effects.

Master Strip
VCM technology recreates classic tape compression and saturation with extraordinary realism. Includes the OPEN DECK effect.

Vintage Stomp
VCM technology merges minute circuit analysis and simulation of individual circuit component characteristics to deliver the best in vintage analog sound.

Mr. Masaaki Okabayashi

Q: To begin, can you tell us how the Add-on Effects were initially conceived?

Mr. Okabayashi : One of Yamaha's strengths in producing digital mixers was the ability to provide built-in effects. Our SPX multi-effect units and reverb units have earned an excellent reputation over the years, and we've been building digital effects into our digital mixing console since the very beginning, 20 years ago. While giving our customers the best possible quality in the analog input and output stages, we also wanted to provide something new that would take maximum advantage of the digital technology involved.
The idea for the Add-on Effects actually came up when we were beginning development of the DM series mixers. It was fortunate that in addition to our hardware product development resources we already had a team devoted to algorithm development, and at the time they just happened to be looking for a platform to direct their efforts toward. It was in 2001, I think, that the decision was officially made to bring these strengths together in an actual product lineup.

Q: What was the market like at the time? What was the competition?

Mr. Okabayashi : In the professional digital live sound arena Yamaha was only offering the PM1D at the time. We had started with the DM series production consoles. Of course there were already TDM and VST plug-ins available for the major computer based audio workstations at the time. But we thought too much of that genre was aimed at hobby users, and were more interested in creating and refining technology that worked seamless with hardware designed for professional use. We decided to be very serious about our signal processing algorithms, and only develop and deliver the very best available.

Q: So does that mean that when the mixer hardware was initially developed the software wasn't yet in the pipeline?

Mr. Okabayashi : Mr. Kunimoto had already been working on physical modeling technology for musical instruments for a long time, so the foundation already existed, and we could clearly see the potential for that technology in a professional audio context.

Q: Physical modeling in the VL and VP synthesizers? That was quite a while ago.

Mr. Kunimoto : Yes. That was around 1993. The VL70m (tone module) is actually still being made and is quite popular. But in around 2000 we were looking for a new direction in which to take that technology. We already had reverb simulations and other effects, and believed that we could create effects that would be ideal for incorporation into digital mixers. By 2001 we had started doing research all around the country in order to pinpoint the possibilities and solidify our goals. It all came together in 2002.

Q. That's really interesting. VL modeling was basically acoustic musical instrument modeling, but it included technology that became the seeds of what we have today.

Mr. Kunimoto : That's right. And by around 2001 or 2002 we had elevated those seminal effects to a level that would be more than acceptable for use in Yamaha digital mixers. If you consider the fact that development of physical modeling technology was begun in around 1990, the history of the Add-on Effects is actually quite long. But since our team was originally part of the musical instrument division, we didn't have algorithms dedicated to audio processing, such as compressors or equalizers, at the time we merged with the pro audio division in 2000. That was a real challenge for us. It's easy enough to simply produce a compressor or equalizer, but creating a good sounding compressor or equalizer is a different story altogether. In fact, we weren't actually sure what constituted "good sound" in a professional audio context. What exactly were the sonic characteristics that professional sound engineers valued?

Q: Was there a specific point at which the direction became clear?

Mr. Kunimoto : Yes. We were very fortunate in being able to recruit the assistance of several outstanding sound engineers who helped us to achieve the excellent performance we are able to offer today. We made many, many prototypes that these talented sound specialists would analyze aurally, easily pinpointing deficiencies that needed to be addressed as well as positive aspects that could be enhanced. Those specialists are so familiar with the makings of "good sound" that they were able to demonstrate the type of performance they wanted by combining existing, traditional equipment and adjusting various parameters while explaining the merits or demerits of the end result. This approach made a lot of sense to us, and meant that we were designing by sound rather than numbers on a spreadsheet or waveforms on test instruments. The moment we realized just how well this type of cooperation was working was the moment we became totally confident that we could deliver something very special. Of course it required many rounds of prototyping and analysis, but we knew we were headed in the right direction. We supply the algorithms, but the final sound comes from extensive cooperation with some of the best sound engineers in the business. When it comes to sound, the most advanced software and hardware is nothing without a good dose of human sensibility. Of course the study and analysis of vintage audio equipment is a tremendous aid to understanding what has come to be appreciated as good sound – sort of a short cut to understanding.
Mr. Okabayashi : Defining "good sound" is very difficult. It can be different things to different people at different times. For one person it might mean listening while seated on a luxurious sofa in a comfortably climate-controlled room. It can depend on a person's mood, too. The challenge for us is to satisfy what is essentially a subjective evaluation with the most effective technology we can muster. We believe that VCM has been very successful in that regard, probably at least in part because of the enormous amount of feedback from professional sound creators that has gone into it.
Mr. Kunimoto : Yes, and much of that feedback has been very specific. For example, some people might simply say the sound is "fat" and is therefore good. But we need to go a step further and understand what constitutes "fat" sound – what kind of response at what frequencies is that? Our consultants were able to give us the necessary information.

Q: Very interesting. It sounds like it was more a process of communication than simply bringing in some vintage gear, analyzing it with tests instruments, and attempting to reproduce its characteristics.

Mr. Kunimoto : Of course we needed to do some of that as well, but that type of laboratory work alone is not going to produce satisfactory results.
We knew from our experience in creating musical instruments that, in contrast to the end listener, professionals who work with sound on a daily basis and who are successful in creating sound that is generally accepted as being top-class usually agree on what constitutes good sound. So when we started seeing agreement from our consulting engineers we knew we had something of real value.

Q: It can't have all been easy. Were there any obstacles that slowed you down?

Mr. Kunimoto : Actually, it all went quite smoothly! Doing the measurements on the vintage open-reel recorders did take a long time though. We worked around the clock for several days on that. Even the slightest change in bias, azimuth, speed, or tape on an analog tape machine completely changes the response. It only takes a few minutes to measure the deck's characteristics, but then the moment we make a change it becomes necessary to bring out the standard tape and go through another 30 minutes of azimuth and level adjustment. So that process repeats: measure for a few minutes, adjust for 30, then do it all over again. Many times.
The studio pros we worked with on that part of the project were delighted with the quality of the simulations we were able to achieve. To their trained ears we had managed to replicate the musicality of the original equipment. So much so that our model would serve as a "virtual archive" of classic recording equipment that will at some point become unserviceable. Comments like that were most heartening, and in many ways a measure of our success.
Mr. Okabayashi : But we should also point out that we were never aiming for exact simulations. Our goal was simply good sound. One of the most difficult aspects of that was bridging the gap between what was required by recording and live sound engineers. That was challenging and enlightening at the same time.
Studio engineers generally have access to a greater variety of equipment, and they have more time to mix and refine the sound. They are also working in a sonic environment that has usually been optimized for the job at hand. Live sound engineers, on the other hand, have to achieve the best possible sound in less-than-optimum acoustic environments that might change from day to day. And they have to do it with a minimum of equipment in the shortest time possible. The working conditions are quite different. So for us it was difficult to strike an appropriate balance between detailed control and quick operation.
There were some surprises too. Originally we hadn't even thought that the Open Deck effects would be used for live applications, but it turned out that live engineers found it extremely useful as a master compressor for final output.
Mr. Kunimoto : Initially we weren't even going to make the Open Deck effects available for live consoles such as the PM1D. But more and more engineers began using them for live sound, which was quite a revelation for us.

Q: How abut the pedal type effects, such as distortion?

Mr. Okabayashi : The basics for live sound are reverb, delay, compression, and EQ, but a few engineers like to experiment with wilder effects as well. That's who the stomp-box effects are for. At the moment it's a relative minority, but we believe that as more engineers realize the creative potential of those effects they will become more widely used.
Mr. Kunimoto : And, frankly, the sound of our flanger, phaser, and other pedal type effects is way ahead of the competition. Our stomp-box simulations are extremely musical and can really add something to a mix if used well.

Q: Some people might wonder why, since digital mixers already have EQ and sometimes dynamics built in, it is necessary to have additional equalizers and compressors.

Mr. Okabayashi : The functions of the head amplifiers, equalizers, and compressors that need to be included in every channel, and those that the engineer wants to insert on specific channels tend to be very different. If the equalizers and compressors on all channels had a strong, distinct character, that would influence the overall sound of the console in a big way. The built-in EQ and compression needs to be more transparent and "practical." But you still need equalizers and compressors with a distinctive sound for certain sources: some compressors ideally match certain types of voices, for example. The Add-on Effects fall into the latter category. They're the type of effects engineers will want to use on specific sources. But I would like to add, once again, that these effects are not exact simulations of specific vintage devices. They are based on in-depth analysis of vintage gear as well as current production and live sound mixing needs. They are, in fact, a blend of the most desirable characteristics of a range of classic equipment, brought together in a way that best applies to state-of-the-art sound.

Mr. Toshifumi Kunimoto

Q: Before we move on to discussing Rev-X and ISSP technology, I'd like to ask Mr. Sendo for his views on the VCM effects from the perspective of a sound engineer.*

** Mr. Sendoh is a recent addition to the Yamaha team. Prior to joining Yamaha he was a live sound engineer with a sound company that is a major player in the field of theater and concert sound.*

Mr. Sendoh : Certainly. As mentioned earlier in the discussion, while the standard dynamics included in Yamaha digital mixers are very good at level compensation and control, many engineers would generally use an external inserted compressor to achieve a more "driven" sound for picked or chopper style bass, for example. But now the Add-on Effects 260 compressor does an outstanding job in that situation, and is in many ways preferable to external hardware. One of the main drawbacks of external gear is the difficulty of achieving accurate, consistent settings. Noise is another issue. Since you can easily switch between patches and the bypassed sound with the Add-on Effects, it's much easier to decide on the optimum setup for a particular situation. As Mr. Okabayashi said earlier, the standard built-in compressors on Yamaha digital mixers are ideal for basic level compensation and adjustment, whereas the Add-on Effects compressors can replace external devices when you want some extra sonic nuance. The same goes for the equalizers. While the built-in channel equalizers are the right tool for basic response compensation, the Add-on Effects equalizers are perfect for enhancing or coloring a sound.
While delivering vintage-style "good sound," the Add-on Effects offer all the advantages of digital control and repeatability, which makes them a more practical and reliable alternative to using actual vintage outboard gear. Also, you can recall settings that were successful in a previous situation and use them as a starting point for any other situation, which can significantly improve setup speed and efficiency.

Mr. Satoshi Sendoh

Q: Any comments about using the Open Deck effects in live situations?

Mr. Sendoh : Open Deck can be a very effective tool for live sound, and I know a number of engineers who us those effects consistently. They use them to provide a touch of "saturation" at the output stage for a more driven, exciting sound while keeping the levels in line.

Q: How about the stomp-box effects?

Mr. Sendoh : Unfortunately they weren't available when I was mixing live on a daily basis, but if I had them I would have used them. One of the big advantages of those effects for live use is the ability to tap-specify the tempo of the phaser and other effects. That's something that simply was not possible with actual vintage gear.

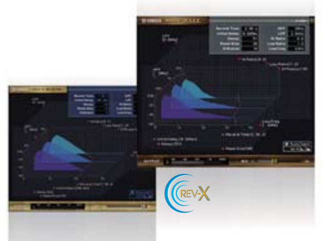
ADD-ON EFFECTS

An Interview with the Creators of Yamaha's Add-on Effects

Part 2: Rev-X Reverb and iSSP (Interactive Spatial Sound Processing)

This is Part 2 of a two-part interview in which we get a rare behind-the-scenes view of the conception and genesis of some of Yamaha's most advanced audio signal processing software: the Add-on Effects.

In Part 1 we discussed VCM technology and the extraordinary compressor, equalizer, open deck, and stomp box simulations it enables. In Part 2 we talk about Rev-X reverb and iSSP surround technology.



Reverb
The advanced Rev-X reverb algorithm, originally developed for high-end stand-alone reverb units, is now available in plug-in form.

Surround Post
Revolutionary iSSP technology developed by the Yamaha Innovative Technology Division powers a selection of amazing surround processors.
These are the surround production tools you need for ultimate control and realism.

Mr. Tomomi Miyata

Q: Reverb is one of the most basic and vital "effects" for both production and live sound, and Yamaha reverb has long been considered among the best in the field. Can you tell us a bit about the idea behind the relatively new Rev-X reverb effects?

Mr. Miyata : Rev-X actually has quite a long history. Development was started about ten years ago, and like VCM it was originally developed for use in electronic musical instruments. The original idea was bring the sound of MIDI tone generators as close as possible to the sound of a recorded CD. When we analyzed the factors that contribute to the overall sound of a CD, we realized that reverb was one of the most significant. At that point we began exhaustive analysis of studio reverb systems as well as dialog with a number of leading studio engineers. Of course many Yamaha reverb systems were already in use in recording studios at the time, but we were determined to find a way to deliver even better reverb effects.

A number of the studio engineers we consulted felt that the Yamaha reverb systems of the time sounded cool and metallic, whereas some preferred a warm, "woody" reverb. The challenge was to find a way to achieve that type of sound, and that's where the work of closely examining the characteristics of both natural and simulated reverb began.

The most important elements – and the most closely guarded secrets – in algorithm-based digital reverb systems are the all-pass and comb filters, and how those filters are used. That's where most of the design energy went in the developing the Rev-X algorithms as well. It's basically a long process of trial and error.

Q: The history of Yamaha digital reverb units is much longer than ten years.

Mr. Okabayashi : That's correct. The earliest units such as the REV-1 and REV-7 go back a long way. The now-legendary SPX-90 was released in around 1985 I think. There was actually an R1000 reverb unit based on BBD technology even before that, but in terms of purely digital reverb the REV-7 and SPX90 can be considered the origins.

Q: Is the technology from those earliest digital reverb units still employed in the Rex-X systems?

Mr. Okabayashi : The background of Rev-X is actually completely different. The basic technology in the SPX series remained fairly constant from the earliest models right up to the SPX2000, but when we were planning the SPX2000 we wanted to add something extra that had not been possible in the preceding SPX models. Yamaha SPX-series reverb units were already established as one of the industry standards at the time, but we wanted to offer expanded sound and capabilities as well. And, as with VCM technology, we were delighted to learn that the kind of innovation we were looking for already existed in-house in the form of the Rev-X algorithms.

Q: So does that mean that Yamaha reverb changed drastically with the release of the SPX2000?

Mr. Okabayashi : No, the SPX2000 offers the best of both worlds. It has all the original and universally accepted SPX type effects such as Rev Hall and Rev Plate, plus the new Rev-X effects.

Q. It seems that wherever Rev-X reverb is mentioned, the SPX2000 is mentioned as well. Apparently the SPX2000 is a very popular product.

Mr. Okabayashi : Yamaha SPX reverb units in general are a well-established standard in studios and in life applications worldwide. If you look at the numbers the SPX2000 just keeps on selling at a surprisingly constant rate, so we must have got it right. It seems as though we hit a "sweet spot" in terms of reverb density and sound that engineers find appealing and easy to use.

Q: OK, lets ask an engineer … Mr. Sendoh?

Mr. Sendoh : From a purely sonic perspective, back when I was using other types of artificial reverb I was never satisfied with the results because the reverb always sounded separate from the source. There was no cohesion or unity to the sound, the way there is with natural reverb. That changed dramatically the moment Rev-X became available in the SPX2000 and newer versions of Yamaha digital consoles. Now the blend is excellent, and very natural.

In addition to adding ambience, a common use for reverb is to add a bit of thickness or solidity to a drum sound, for example. I often used a short room reverb effect for that purpose, but prior to Rev-X it was sometimes difficult to avoid a metallic sounding, almost ring-modulation type effect. Once again, the REV-X reverbs solved that problem completely, and gave me exactly the sound I was after with a minimum amount of work. It's great to be able to achieve the desired sound without having to carry racks of additional gear just for that purpose.

Mr. Daisuke Miura

Q: What were the difficulties encountered in creating such an effective reverb?

Mr. Miyata : Evaluating reverb effects is very difficult. What sounds great one day can sound deficient the next. The listening environment, source, and even listening position all have a huge effect. Even the listener's physical condition on a particular day can make a difference. I'm sure that applies to all effects, but it does seem to be particularly significant when working with reverb, so it's necessary to combine subjective listening tests with empirical data to obtain a reliable evaluation. It requires repeated listening and measurement, which means that progress is incremental and very slow. Persistence is a key ingredient.

Mr. Miura : In the final stages it becomes more craft that design. You change the data a bit and listen, then repeat. The sound changes very subtly each time, and it is necessary to keep correcting course to get closer and closer to the sonic goal. And there are subtle differences between the development platform and the final product that need to be resolved, as well limitations imposed by the hardware. It's a very tedious process, but the end result is well worth it.

Q: Let's move on to the iSSP (Interactive Spatial Sound Processing) technology used in the Surround Post Package. How did the project initially arise?

Mr. Okabayashi : We were very conscious of the need for top-level 5.1 surround production facilities when planning the DM2000 console. At the time many post production and music production facilities were using two cascaded 02R consoles for 5.1 surround production. We wanted to provide a single console that would be able to handle the same type of production, with the addition of 96 kHz operation as well as advanced surround production features. At the time engineers were creating extremely complex automation setups to achieve what can be done quickly and easily with iSSP now. The main idea was to make those very complicated and time-consuming surround production tasks as easy and as efficient as possible. Producers were writing up the equivalent of "storyboards" for sound positioning and movement, for example, and then the engineers would have to use whatever signal routing and processing facilities were available to produce the desired result. They might first have to use multiple auxiliary sends to pan the sound, and then add the required reverb to match the sound as it moves. Tricky application of delay and EQ are also essential ingredients. This is a very labor-intensive process that we really wanted to make easier. So the initial concept was very direct: to provide unprecedented surround processing convenience and speed.

Q: When was development begun?

Mr. Sekine : We actually started field evaluations in around 2002. But at the very early stages of development, which was around 2000, we were occupied with finding ways to effectively move and position sound in the surround field. The first problem was finding a suitable development platform, and we ended up cascading two of Yamaha's most powerful processors at the time. That worked out very well, and as soon as we had a system that we could take around for evaluation we began spending time with the producers and engineers at the national broadcasting corporation as well as several other major broadcasters around the country.

Mr. Takahashi : We also took the prototype to Chicago and New York. That was in around July of 2002 I think.

Mr. Sekine : That's right. Those evaluations led us to the conclusion that we had created some valuable tools for surround post-production, and work on transplanting those functions to an actual mixing console was begun. Since we had used a Yamaha platform for early development, the transfer to Yamaha digital consoles went very smoothly. The most difficult part was creating a user interface that was easy to use. We had to create something totally new that would make the equivalent of running multiple auxiliary sends to pan a sound while simultaneously making dynamic reverb and EQ changes, for example, easy and intuitive. Making it easy to specify a constantly changing position, distance, and speed was a real challenge. We think the Auto Doppler input display is an excellent example of how well we've succeeded. That interface makes it surprisingly easy to match sound motion to movie or video images.

Mr. Takahashi : We also had to deal with the finite number of parameters available on the DM2000. We had 32 available parameters … and it's actually difficult to decide whether that's too many or too few. Deciding what surround parameters would be assigned to the available control parameters for optimum ease of use was not easy. That process alone took many months. Actually it's probably more accurate to say that those decisions were continuously being made throughout the development process. And since different engineers have different ways of working, there's no single correct answer.

Mr. Sekine : When we were working on the development platform we sometimes ended up with as many as 90 parameters!

Mr. Takahashi : Right. And controlling that many parameters from a computer, as we were doing during development, is not too difficult. But when you attempt to transfer that functionality to a mixing console you run into limitations. At that stage the Yamaha industrial design team were a tremendous help. They came up with interface ideas that eventually resulted in the incredibly intuitive control system we have now. Of course there's also the Studio Manager computer software that allows even more detailed control.

Q: The interface is definitely one of a kind. So would it be correct to say that the main purpose of iSSP is to expedite surround processing operations that were previously difficult and time consuming?

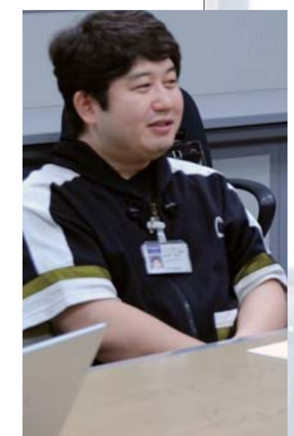
Mr. Sekine : Basically, yes. But we also put a great deal of energy into providing the best possible sound quality, particularly in terms of depth and presence. The Room ER is a good example of what we've achieved in terms of depth and presence. Room simulation is something that Yamaha has been working on in a variety of contexts for a long time, and that background has really come to life in the Room ER effects. It produces a realistic sense of the acoustic space, right down to the materials of the simulated room's surfaces.

Q: Finally, a general question. I know this is probably a difficult question to answer, but what are Yamaha's plans for the future in the field of digital sound?

Mr. Okabayashi : We're working on new ideas and refinements all the time, but we can't provide any details just yet. There's still plenty to do. For example, there are many wonderful vintage processors left to examine and consider. And development of reverb technology is still ongoing. The same goes for EQ. There might even be some fundamental changes in mixing technology.

All we can really say is: be prepared for some pleasant surprises!

Mr. Satoshi Sekine



Mr. Akio Takahashi

DME Designer – DME Control Software for PCs

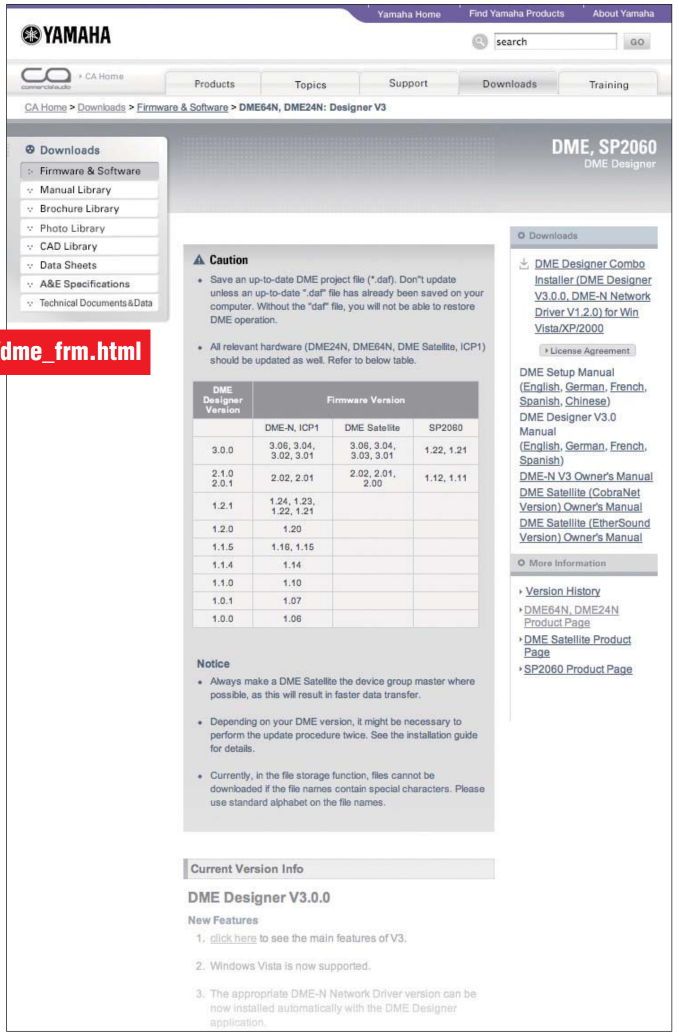
Quick Guide to Download, Installation & Use

With DME Designer installed on your PC, you get an extremely easy-to-use graphic interface for programming DME Series devices in a convenient and intuitive fashion. Whether you already own a Yamaha digital mixing engine or simply want a sneak peek, this application is free to download – no strings attached. Please follow the simple steps below to see for yourself all that the DME Series has to offer.

1 Downloading DME Designer

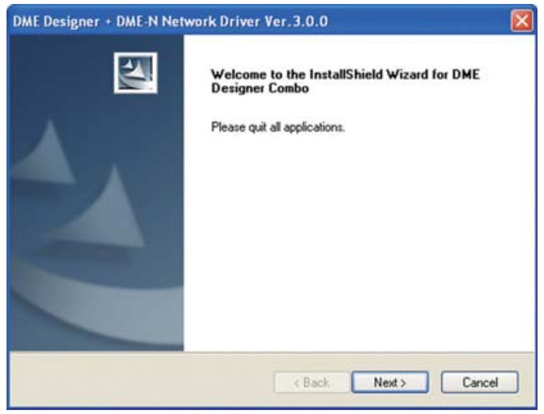
DME Designer is free to download by anyone from the Yamaha Pro Audio website. Please confirm that your PC satisfies the following requirements, and then use the following link to download the DME Designer installer.

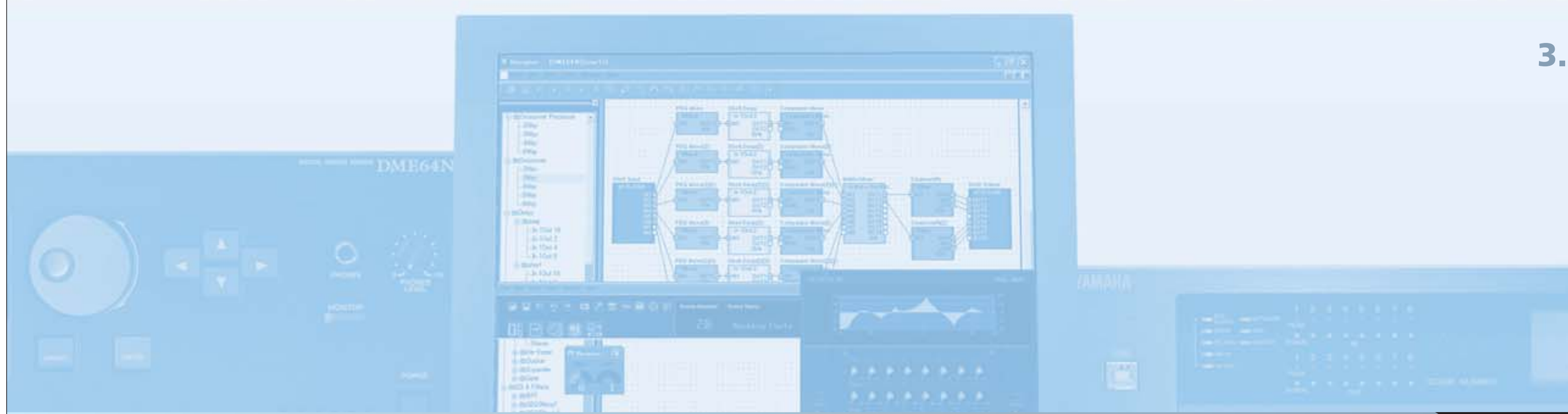
http://www.yamahaproaudio.com/downloads/firm_soft/dme64n/dme_frm.html



2 Installing DME Designer

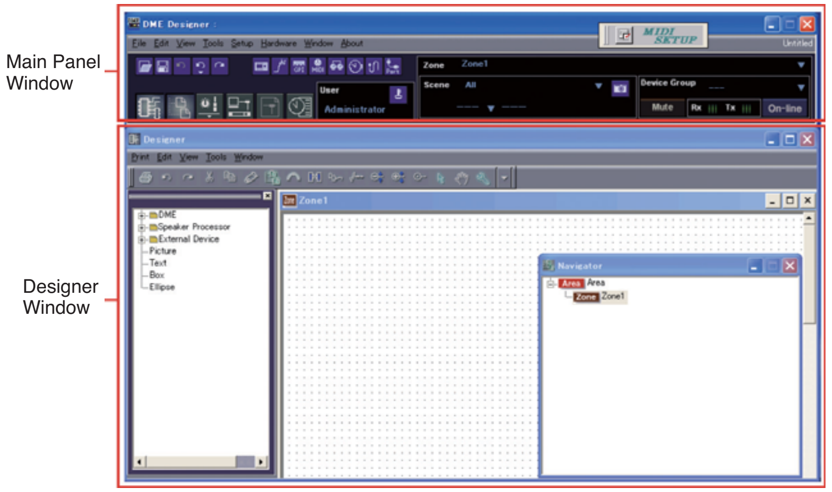
To install DME Designer on your PC, double-click "setup.exe" from the Installer folder and then follow the on-screen instructions.





3 Using DME Designer

Even if you do not yet own a DME Series device, DME Designer is the ideal way to see for yourself just how powerful and flexible the Yamaha digital mixing engines can be. The following is a quick guide to get you started.

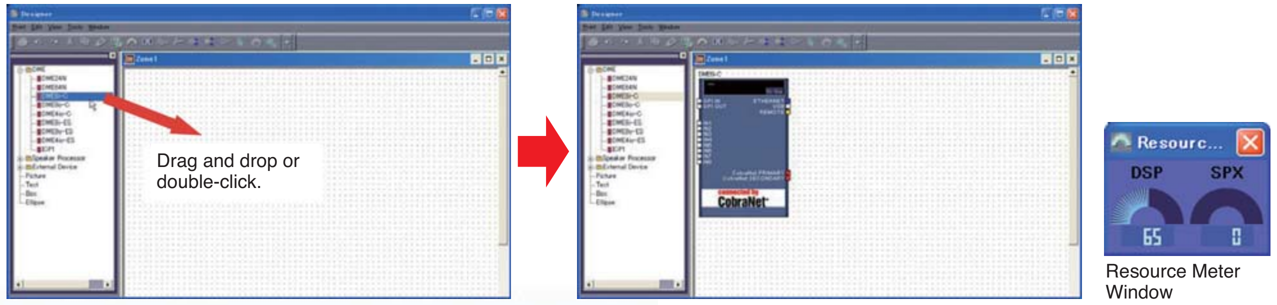


Creating Configurations

With DME Designer, the process of selecting, combining, and connecting various audio components is extremely easy and intuitive. Please note that these configurations can be created even with no DME device connected to your PC.

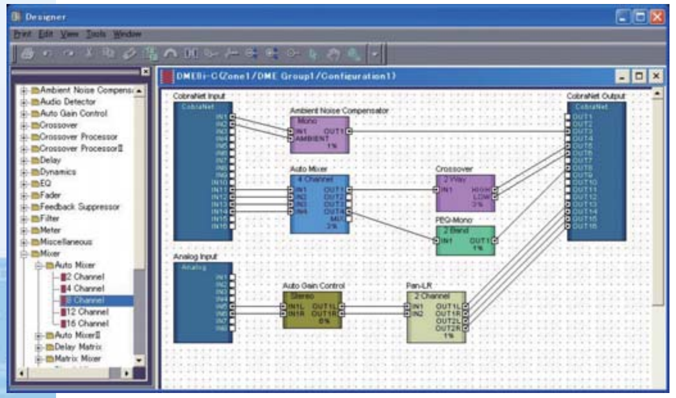
1. **Launch DME Designer by clicking [Start] → [All Programs] → [YAMAHA OPT Tools] → [DME Designer] → [DME Designer].**
   As shown here, a new project will be created and a new zone (Zone 1) will be displayed in the Designer window. (If you cannot see the Designer window, please click [Show/Hide] in the Main Panel Window to display it.)

2. **Add the desired DME objects to your project.**
   All available DME hardware objects can be accessed in the Toolkit window on the left. To add one to your project, simply drag and drop into Zone 1 or just double-click it in the Toolkit window.



***Note:** Where necessary, set the device group or sampling rate using the dialog box that appears when adding a DME object. If the sampling rate set here and the actual sampling rate set for the DME operation are not the same, excessive amounts of signal processing may be required, and in certain cases, operation may not be possible. Please, therefore, ensure that you match sampling rates.*

3. **Double-click a DME object to add and connect components.**
   Once a hardware object has been added to your project, you can begin to create your configuration by freely adding, connecting, and setting audio-processing components. To do so, double-click the object to open the Configuration window. On the left, you will find components to suit practically every need. Either drag and drop or double click to add them to your DME Series device. For more details on creating configurations, please refer to the DME Designer manual.



For details on the settings required for connecting DME Designer and DME Series devices, please refer to the DME Setup Manual, which is free to download from the following site:

http://www.yamaha.co.jp/manual/english/result.php?WORD=DME64N&div=pa

Quality Control

"Quality" is one of those little words that cover a lot of ground. It can mean different things to different people at different times, but at Yamaha it applies to a whole spectrum of concepts that form the backbone of a uniquely conscientious approach to product development and manufacture. Sonic quality, although it is often the first aspect that comes to mind, is only the beginning. Reliability and durability are just as important, and are in many ways more difficult to achieve with any degree of consistency. Then of course there's safety, both personal and environmental, to which an extensive gamut of important standards apply. Unique to electronic devices is the need to prevent electrical interference, both incoming and outgoing, which is an area that requires an extraordinary level of skill combined with advanced facilities for effective management and control. And quality management must continue even after the product is sold, in the form of support and service.

To achieve the kind of quality that satisfies all conditions all of the time requires focused, unrelenting attention to detail and control right from initial product planning and design through final manufacture and packaging to post-sales support. It is not a simple task, and requires a dedicated organization and infrastructure for effective implementation. This is where many manufacturers fall short, but is where Yamaha's commitment to delivering unequalled quality in all areas is overwhelmingly clear. And the fact that the Yamaha approach works is evident in an outstanding track record and enviable reputation.

Large EMC Test Chamber

Overall Quality Management

Yamaha's Quality Management System conforms to ISO 9001:2000 standards and is certified by DNV (Det Norske Veritas – an internationally recognized certification company based in Norway). The Yamaha system, however, has been customized to even more stringent criteria that reflect some very ambitious internal quality goals. The Quality Management System applies not only to operations in Japan, but to Yamaha's factories in China and Indonesia as well. The all-inclusive scope of the system ensures that the same policies, objectives and standards are shared by all Yamaha staff and facilities, no matter where they may be, so that the required level of product and service quality can be maintained on a worldwide scale.

The Yamaha Quality Support Center

Near the entrance to one of Yamaha's main office and factory complexes stands an imposing, almost windowless structure that is a vital arm of Yamaha's Quality Management System. The Quality Support Center is a world-class testing laboratory that houses some of the most advanced and sensitive testing facilities for electronic devices available anywhere, plus some tortuous durability tests that are almost shocking in their severity. The Quality Support Center complies with ISO 17025 standards: "general requirements for competence of testing and calibration laboratories." Not many manufacturers operate an internationally accredited facility of this scale or capability.

Factory Quality Control

Factory production can only begin after the final engineering samples have passed all tests and have been fully approved by the Quality Support Center. But that is by no means the end of quality management. Monitoring and testing continue throughout the manufacturing process to ensure that quality goals are maintained. Parts received from external suppliers must pass testing at the factory Quality Assurance Center before they can be accepted as stock or passed on to the assembly staff. Then, when assembly is complete, each and every unit undergoes a thorough final inspection right at the point of manufacture, so that if a problem is detected it can be rectified immediately and effectively. In addition to inspection and testing of every unit produced, samples are taken from every production run for even more in-depth testing. Approximately five samples will be taken per month, depending on the product, with at least one sample taken at the beginning of each production run. Sample production units are taken to a separate area of the factory where they are tested under actual-use conditions.

The Ultimate Goal

In addition to the obvious need for absolute safety, the ultimate goal of the Yamaha Quality Management System is total customer satisfaction. Total customer satisfaction can only be achieved by providing a stable supply of products of the highest quality at the lowest possible cost along with responsive and effective support. Easier said than done. Like the products themselves, quality management must continually evolve to keep pace with continuously changing markets, user needs, and technology. The Yamaha Quality Management System is right at the leading edge.

Cell Production (Made in Toyooka/Japan)

Non-destructive X-ray Tomography

Cable Durability Testing

Computer-controlled Vibration Table

Encoder Durability Testing

Drop Test

Notes:

Notes: