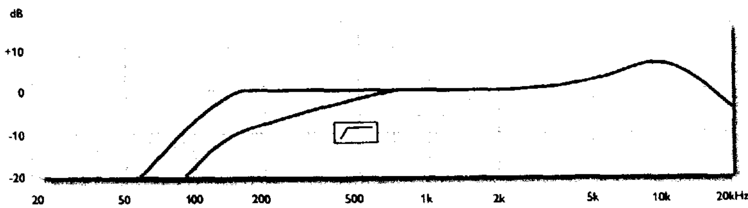
7. Technical Specifications KMS 140/150

Acoustical oper. principle	Pressure gradient transducer
Directional pattern	cardioid/hypercardioid
Frequency range	20 Hz...20 kHz
Sensitivity[1] at 1 kHz	15/10 mV/Pa ± 1 dB
with preattenuation	4.7/3.1 mV/Pa
Rated impedance	50 ohms
Rated load impedance	1000 ohms
S/N ratio CCIR 468-3	68 dB/66 dB
Equivalent SPL CCIR 468-3	26 dB/28 dB
Equivalent SPL DIN/IEC 651	16 dB-A/18 dB-A
Max. SPL for less than 0,5% THD[2]	138 dB/142 dB
with preattenuation	148 dB/152 dB
max. output voltage	10 dBu
Phantom powering (P48, IEC 1938)	48 V ± 4 V
Current consumption	2 mA
Matching connector	XLR 3 F
Weight	approx. 285 g
Diameter	48 mm
Length	175 mm

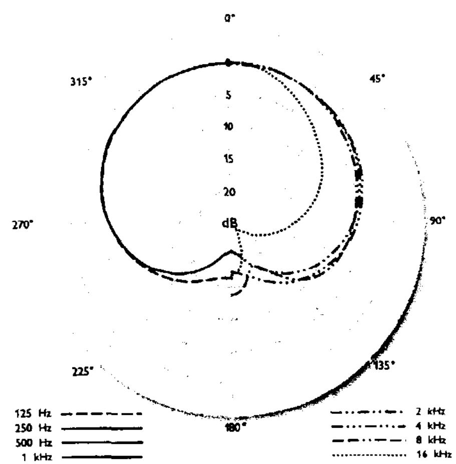
[1] at 1 kHz into 1 kOhm rated load impedance. 1 Pa ≘ 94 dB SPL.
[2] THD of microphone amplifier at an input voltage equivalent to the capsule output at the specified SPL.

8. Frequenzgänge und Polardiagramme / Frequency Range and Polar Pattern

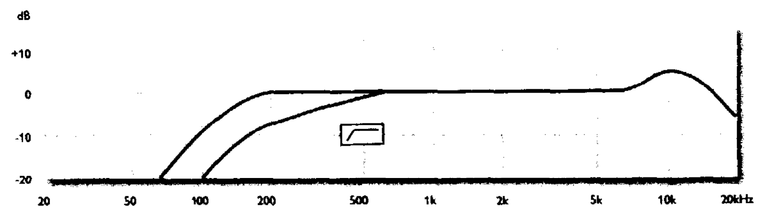
KMS 140



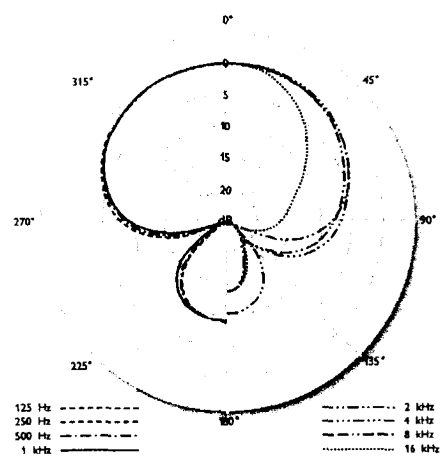
gemessen im freien Schallfeld nach IEC 60268-4
measured in free-field conditions (IEC 60268-4)



KMS 150



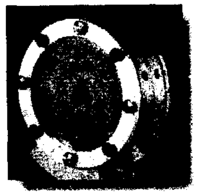
gemessen im freien Schallfeld nach IEC 60268-4
measured in free-field conditions (IEC 60268-4)



U 89 i

Large Diaphragm Microphone

The U 89 is a studio microphone for universal applications. The headgrille protects a dual-diaphragm capsule. A rotary switch below the headgrille selects from five different polar patterns. Therefore the microphone can be adapted easily to large sound sources, and those that are spread wide apart, or to sound sources to be recorded at a greater distance.



The amplifier accepts sound pressure levels up to 134 dB without distortion. This figure can be increased to 140 dB. An additional rotary switch activates a filter that changes the low frequency response either below 80 Hz or 160 Hz.

Features

- Variable large diaphragm microphone
- Pressure-gradient transducer with double membrane capsule
- Five directional characteristics: omni, wide angle cardioid, cardioid, hypercardioid, figure-8
- Thereby most versatile in all recording situations
- Two-stage roll-off filter
- Switchable 10 dB pre-attenuation
- Extended frequency range in comparison to U 87 Ai

Applications

The U 89 i is similar in appearance to the U 87. It is of smaller size, and lighter weight. It features five instead of three directional characteristics and a higher maximum sound pressure level which make this microphone easier adaptable to different applications.

Polar patterns

In addition to the usual directional polar patterns: omnidirectional, cardioid, and figure-8, we have added a hypercardioid and wide-angle cardioid characteristic.

When compared to the standard cardioid pattern, the hypercardioid characteristic suppresses sound from the side more efficiently. The wide-angle polar pattern is especially useful to record large sound sources.

Acoustic features

The microphone is addressed from the front, marked with the Neumann logo. The large diaphragm capsule has a very smooth frequency response for all polar patterns over a wide acceptance angle. The frequency response curves are flat up to 10 kHz within a pickup angle of ± 100°.

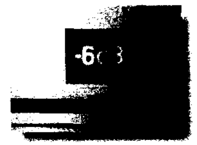
As a result the U 89 i has a very even diffuse-field response for all polar patterns. This is important in a reverberant environment when more reflections arrive at the microphone capsule. The acoustic information is not affected in its tonal quality when recorded by the microphone.

This characteristic is achieved without resorting to corrective resonance effects.

The capsule it is elastically mounted to avoid any structure borne noise that could interfere with its operation.

Filter and attenuation

The amplifier handles sound pressure levels up to 134 dB without distortion.



With a self noise level of 17 dB (A-weighted) the total dynamic range is 117 dB. Maximum sound pressure level is 140 dB when the - 6 dB rotary switch is in the ON position.

A low frequency roll-off at 80 Hz or 160 Hz can be activated with the third rotary switch below the headgrille. This filter suppresses low frequency interference, yet maintains an even frequency response for close-up sound sources, for example, when proximity effect could adversely affect the program material.

A steep high-pass filter in the LIN position prevents the output transformer of the microphone from being overloaded due to undesired subsonic frequencies.

Operational safety

All exposed surfaces of the microphone capsule, including the diaphragms, are at ground potential. This technology makes them highly immune to electrical and atmospheric interference and contamination through microscopic dust particles.

Delivery Range

Microphone U 89 i (mt)
Wooden box

Catalog No.

U 89 i	ni	06449
U 89 i mt	blk	06450

Selection of Accessories

Battery supply, BS 48 i	blk	06494
Power supply, N 48 I-2 (230 V)	blk	06500
Power supply, N 48 I-2 (117 V)	blk	06502
Power supply, N 48 I-2 (without plug-in mains unit)	blk	06504
Auditorium hanger, MNV 87	ni	06804
Auditorium hanger, MNV 87 mt	blk	06806
Elastic suspension, EA 89 A	ni	07195
Elastic suspension, EA 89 A mt	blk	07196
Stand mount swivel, SG 389 mt	blk	06620
Popscreen, PS 20	blk	07346
Windscreen, WS 89	blk	07197
Microphone cable, IC 4 mt (with stand mount swivel)	blk	06557

A complete survey and detailed descriptions of all accessories are contained in the accessories catalog.

Meaning of color codes:
blk = black
ni = nickel

Application Hints

- A microphone for universal usage
- Use as spot mic for
  - wind instruments,
  - strings,
  - piano

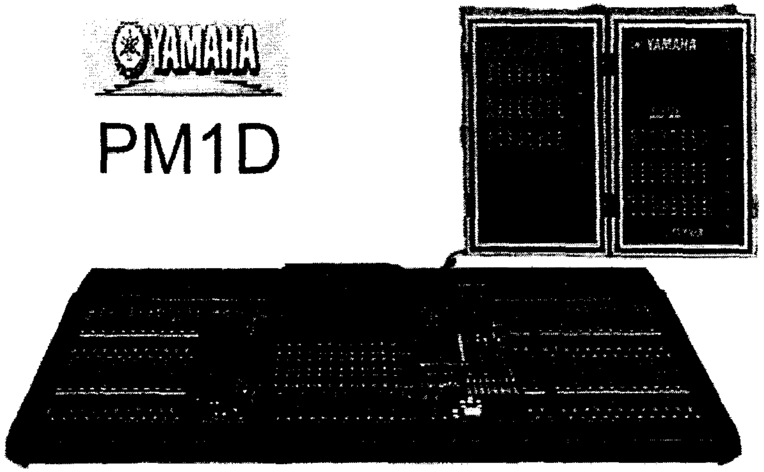
These are just some of the most common applications. We recommend additional experimentation to gain maximum use from this microphone.

Technical Data

Acoustical operating principle	Pressure gradient transducer
Directional pattern	Omnidirectional, wide angle cardioid, cardioid, hypercardioid, figure-8
Frequency range	20 Hz...20 kHz
Sensitivity at 1 kHz into 1 kohm	8 mV/Pa
Rated impedance	150 ohms
Rated load impedance	1000 ohms
Equivalent SPL CCIR 468-3	28 dB
Equivalent SPL DIN/IEC 651	17 dB-A
S/N ratio CCIR 468-3	66 dB
S/N ratio DIN/IEC 651	77 dB
Maximum SPL for THD 0.5%	134 dB
Maximum SPL for THD 0.5% with preattanuation	140 dB
Maximum output voltage	800 mV
Dynamic range of the microphone amplifier DIN/IEC 651	117 dB
Supply voltage	48 V ± 4 V
Current consumption	0.8 mA
Matching connector	XLR 3F
Weight	400 g
Diameter	46 mm
Length	185 mm

YAMAHA

PM1D



Puissance de traitement, flexibilité, mise en oeuvre, c'est une console de mixage audio-*numérique, c'est la PM1D*

Ergonomie très intuitive, programmation simple, total reset
Beaucoup d'ingénieurs du son familiers des consoles analogiques ont une certaine réticence à travailler sur des machines numériques, et ce, à juste titre, car, il faut bien le reconnaître, certaines consoles numériques ne sont pas d'un abord toujours attractif. C'est tout le contraire avec la PM1D. Le système PM1D offre un contrôle très intuitif et flexible de ses composants grâce notamment à la surface de contrôle de style analogique qui comporte des encodeurs très précis, des faders motorisés et des contrôles visuels exhaustifs. En outre, tous les paramètres de mixage sont programmables via l'interface informatique sur écran couleur offrant une visualisation très précise et une lisibilité optimale. Le logiciel de programmation peut fonctionner off-line sur un ordinateur PC standard. Les mémoires peuvent alors être stockées sur une cartouche de type PCMCIA et rechargées ensuite sur la console. Jusqu'à 990 mémoires de scène les plus complexes pourront être stockées et rappelées instantanément avec fonctions d'undo et de preview pour réduire au minimum les risques d'erreurs.

Traitement intégralement numérique
Yamaha n'est pas un nouveau venu dans le monde de l'audio numérique. Les consoles de mixage numériques Yamaha ont révolutionné l'industrie de l'enregistrement et de la production et leur qualité audio et la puissance que leur traitement procure sont de tout premier ordre. La technologie de Yamaha a bien sûr été intégrée dans la PM1D avec de nombreuses avancées et des améliorations substancielles qui placent ses performances audio à un niveau encore jamais atteint. Toutes les opérations de mixage et de traitements sont entièrement effectuées dans le domaine numérique à 44,1 ou 48 kHz de fréquence d'échantillonnage. De nouveau convertisseurs AD (en 28 bits) et DA (en 27 bits) assurent le passage entre l'analogique et le numérique avec une transparence et une dynamique extraordinaire. Des circuits DSP très puissants assurent le traitement pour chaque tranche en EQ, compression ,effets, etc. La PM1D possède un ensemble de traitements audio si vaste qu'il n'est plus besoin de câbler des racks entiers de modules de traitement dynamiques et autres générateurs d'effets externes. Bien sûr, il sera toujours possible d'insérer n'importe quel appareil de traitement externe grâce au patch interne sans limitation intégré à la console.

Mise en oeuvre très simple, transport aisé, modularité des composants
Le système complet PM1D est extrèmement compact : la console elle-même et le ou les racks (en fonction des besoins en entrées/sorties) ont un encombrement considérablement réduit, léger et facile à manipuler comparativement à une console analogique de même architecture. Cette particularité rend la PM1D particulièrement recommandée dans les salles de concert ou la place est souvent comptée ainsi que pour les régies mobiles.

Câblage minimal et rapide
Le contraste est saisissant entre le câblage nécessaire pour une régie analogique et celui de la PM1D. Les racks d'entrées/sorties étant déportables jusqu'à 200 mètres de la surface de contrôle, le "multipaire" de la PM1D se compose de deux câbles Ethernet et d'un câble numérique à 68 broches véhiculant 32 canaux. Le système pourra être prêt à fonctionner en 5 minutes.

Configuration souple des canaux d'entrées
Le coeur du système PM1D est constitué par un rack de traitement appelé DSP1D. Ce rack effectue tous les traitements audio de la console. Les entrées et les sorties sont accessibles sur des racks de convertisseurs qui sont câblés sur le rack DSP1D. Un système 48 entrées par exemple, utilise 4 modules AI8, les trois premiers comportent 16 entrées micro doublées A/B, le quatrième comporte des entrées micro et des entrées ligne pour gérer les entrées stéréo et d'éventuels retours d'inserts. Il suffira de doubler ces modules et d'ajouter une carte DSP supplémentaire dans le rack DSP1D pour constituer une PM1D à 96 canaux. La surface de contrôle CS1D pourra contrôler deux racks DSP1D à elle seule pour des configurations encore plus importantes.

Fiabilité et redondance
La redondance est un point essentiel car on ne sait jamais ce qui peut arriver en situation de "live". Deux racks DSP1D peuvent être utilisés en mode "miroir" avec une fonction de basculement automatique. D'autre part, si la surface de contrôle devient inactive, le système continue de fonctionner, il est en outre possible de "prendre la main" avec un ordinateur.

Configurations de base:

	48ch (units)	96ch (units)
CS1D	1	1
PW1D	1	1
DSP1D	1	0
DSP1D-EX	0	1
AI8-ML8	3	7
AI8-AD8	0	1
AI8-ML4AD4	1	0
AO8-DA8	2	3
Inputs	48 mono, 4 stereo, 16 inserts **total 72 inputs**	96 mono, 8 stereo, 32 inserts **total 144 inputs**
Outputs	2 stereo, 24 mix, 12 matrix, 16 insert out, 8 (talkback, monitor, etc) **total 64 outputs**	2 stereo, 24 mix, 12 matrix, 32 insert out, 24 (talkback, monitor, etc) **total 96 outputs**

Overview of the PM1D system

The PM1D system is a full-digital SR mixing system that consists of a CS1D console, PW1D power supply, DSP1D-EX {DSP1D} DSP unit(s), AI8 analog input unit(s), AO8 analog output unit(s), DIO8 digital input/output unit(s), and input/output cards. This section describes the ways in which the PM1D system differs from conventional analog mixing consoles.

Full-digital/separate type SR mixing system

The PM1D is a full-digital SR mixing system using cutting-edge digital audio processing technology. 28 bit linear equivalent AD converters and 27 bit linear equivalent DA converters are used to ensure a dynamic range of better than 120 dB, for astoundingly high quality.
The system is divided into components such as engine, console, and input/output units. The compact modules allow the system to be configured flexibly, provide an amazing number of inputs and outputs, and ensure excellent portability and operability.

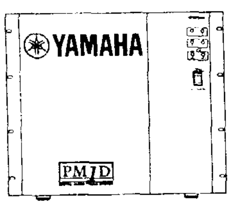
Component structure

The following types of components make up the PM1D system.

- **Engine (DSP1D-EX {DSP1D})**
  Up to ten input units and six output units can be connected to this DSP unit, which performs the majority of audio processing such as audio signal input/output, mixing, routing, and EQ/dynamics/effects.

  The PM1D system offers the following two types of engine.

Engine	Monaural input channels	Stereo input channels
DSP1D-EX	96	8
DSP1D	48	4



Hint
By installing an optional input DSP board (IDB1D) in the DSP1D, it can be upgraded to the same specifications as the DSP1D-EX.

⚠ The board must be installed by a Yamaha service engineer. Never attempt to install this board yourself.

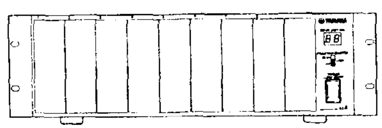
- **Analog input unit (AI8)**
  This input unit inputs analog audio signals to the engine. It has eight slots in which input cards can be installed.

  The following two types of cards can be installed in the AI8.

Card		Input jacks	Number of channels
LMY2-ML	Mic/line input card	1A, 1B, 2A, 2B	2 (select either A or B)
LMY4-AD	AD card	1–4	4

The following three models of AI8 are available, with different cards installed at the factory.

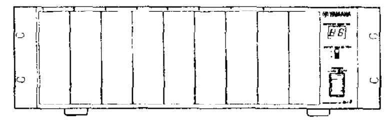
Input unit	Cards installed
AI8-ML8	LMY2-ML × 8 cards
AI8-AD8	LMY4-AD × 8 cards
AI8-ML4AD4	LMY2-ML × 4 cards, LMY4-AD × 4 cards



⚠ Cards must be installed in the AI8 by a Yamaha service engineer. Never attempt to install these cards yourself.

- **Analog output unit (AO8)**
  This output unit outputs analog audio signals from the engine. The AO8 has eight slots, with eight LMY4-DA DA cards installed at the factory.

Card		Output jacks	Number of channels
LMY4-DA	DA card	1–4	4



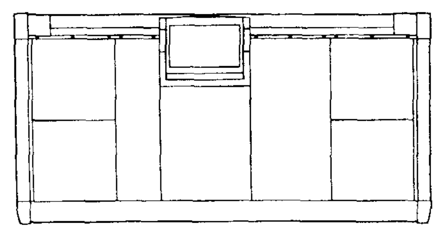
⚠ Cards must be installed in the AO8 by a Yamaha service engineer. Never attempt to install these cards yourself.

- **Digital input/output unit (DIO8)**
  This unit performs input/output of digital audio signals in ADAT, Tascam, AES/EBU formats, as well as analog audio signals, to and from the engine of the PM1D system. The DIO8 has eight slots which can accommodate digital I/O cards or analog I/O cards.

  The following eight types of card can be installed in the DIO8.

Card	Format	Input	Output
MY8-TD	TASCAM	8 IN	8 OUT
MY8-AT	ADAT	8 IN	8 OUT
MY8-AE	AES/EBU	8 IN	8 OUT
MY8-AD	ANALOG IN	8 IN	—
MY4-AD	ANALOG IN	4 IN	—
MY4-DA	ANALOG OUT	—	4 OUT

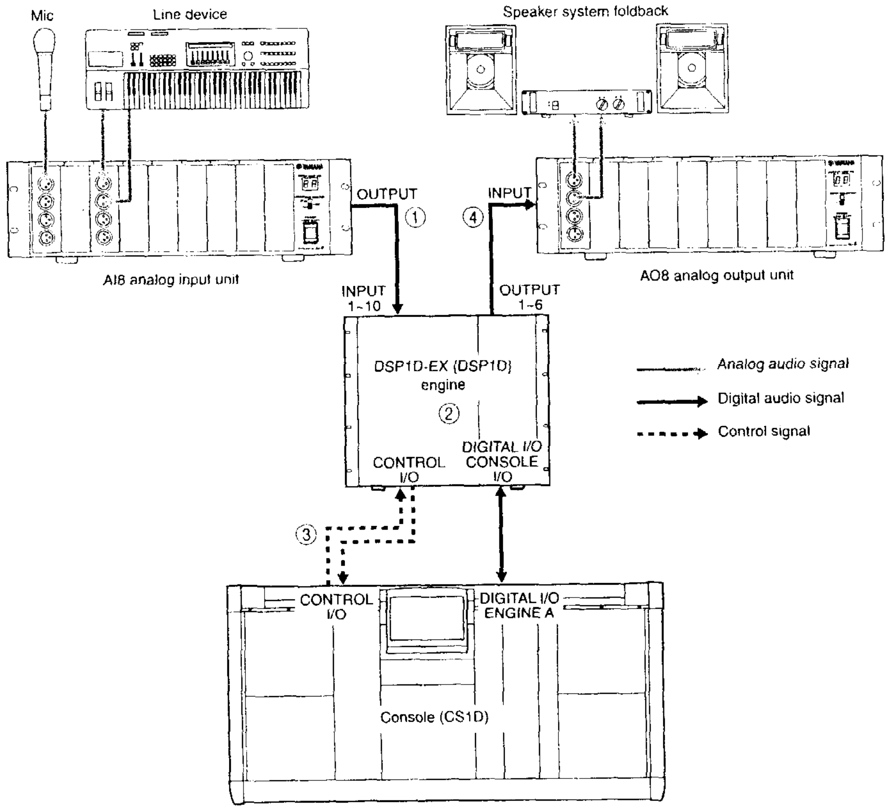
- **Console (CS1D)**
  This console controls the engine. Although it has the appearance of a conventional mixing console, the CS1D is simply a controller for controlling the engine. Please be aware that with the exception of some monitor signals, the audio signals of the PM1D system are handled by the engine.



- **Power supply (PW1D)**
  This is the power supply that provides power to the console.

Signal flow in the PM1D system

The following diagram shows the general signal flow within the PM1D system.



① The signals input to the AI8 analog input unit are AD converted, and then sent as multi-channel digital audio signals to the DSP1D-EX {DSP1D} engine.

② The signals sent to the DSP1D-EX {DSP1D} engine are processed by mixing, routing, EQ/dynamics/effects.

③ In general, the operation of the engine and of the input unit is controlled from the CS1D console.

Hint

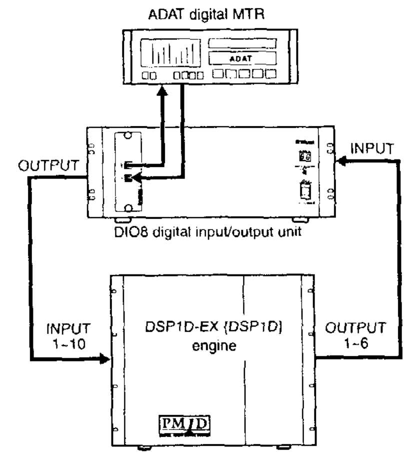
The signals that are input from the 2-TRACK IN DIGITAL jacks 1-6 and 2-TRACK IN ANALOG jacks 1/2 of the CS1D can also be sent to the engine.

④ The signals processed by the engine are DA converted by the output unit, and sent to the speaker system, foldback system, or recording system.

Hint

Signals can also be output from the STEREO OUT DIGITAL jacks or MONITOR OUT ANALOG jacks of the CS1D.

When a DIO8 digital input/output unit is used, the same unit will be used both as an input unit and output unit, so that the signal flow will be as shown on the next page.

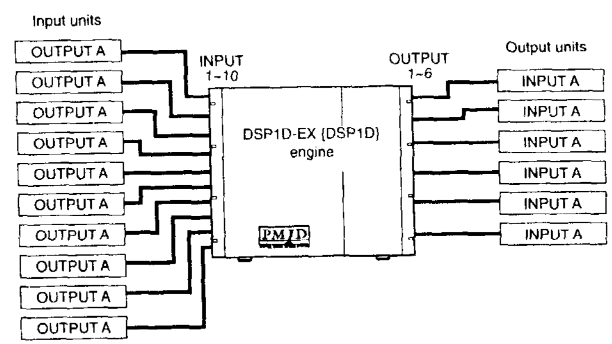


Number of inputs/outputs and channel structure

The DSP1D-EX {DSP1D} engine provides INPUT connectors 1-10 for connecting input units, and OUTPUT connectors 1-6 for connecting output units.

DSP x 1

When the PM1D system is used in Standard mode, up to ten input units (maximum of 320 input connectors) and up to six output units (maximum of 192 output connectors) are connected to one engine.



BLOCK & LEVEL DIAGRAM

MIXING CONSOLES

Annexe 6