MRX7-D Specifications

General Specifications

Memory bank	PRESET: 50
Sampling frequency rate	48kHz/44.1kHz
Signal delay	1.9 msec (AD-DA @48kHz)
Total harmonic distortion*	“0.05% (+4dBu, Gain:-6dB, 48kHz) 0.1% (+4dBu, Gain: +66dB, 48kHz)”
Frequency response	20Hz to 20kHz: max:+0.5dB, min:-1.5dB
Dynamic range	107dB (typ. Gain:-6dB)
Phantom Power	+48V
Crosstalk (@1kHz)	-100dB
Heat diffusion	55.9kcal/h max
Power requirements	AC100V-240V 50Hz/60Hz
Power consumption	65W
Dimensions Weight	480W x 88H x 366Dmm 6.5kg
Included items	Power cord,, Euroblock plugs (3-pin, tabbed) x16, Euroblock plugs (16-pin) x2, Cable Ties, Manual

* Total harmonic distortion is measured with a 18dB/Oct filter @80kHz.

Digital Input and Output Specifications

Terminal	Format	Level	IN/OUT	Connector
YDIF in	YDIF	RS-422	16 in	RJ45
YDIF out	YDIF	RS-422	16 out	RJ45
Primary / Secondary	Dante	1000base-T	64 in / 64 out	RJ45

Control I/O Specifications

Terminal		Level	Connector
GPI 16IN / 8OUT	In	0V-5V (IN 16 L(0V-2.5V) /H(2.5V-24V))	Euroblock
	Out	Open Collector	Euroblock
	+V	DC5V	Euroblock
REMOTE		RS-232C (BAUD RATE : 38.4kbps or 115.2kbps)	D-sub 9pin (Male)
DCP		-	RJ-45

Analog Input Specifications

Input Terminal	GAIN	Actual Source impedance	For use with nominal	Input Level		Connector	Balanced / Unbalanced
				Nominal	Max. before clip		
INPUT 1-8	+66dB	10kΩ	50-600Ω Mics	-62dBu (0.616mV)	-42dBu (6.16mV)	EUROBLOCK (5.08mm pitch)	Balanced
	-6dB		600Ω Lines	+10dBu (2.45V)	+30dBu (24.5V)		
ST IN 1,2	-	10kΩ	600Ω Lines	-10dBV (316mV)	+10dBV (3.16V)	RCA Pin Jack	Unbalanced

* In these specifications, 0dBu = 0.775 Vrms., 0dBV = 1.00 Vrms.
* All input AD converters are 24bit linear, 128times oversampling.
* +48V DC (phantom power) is supplied to INPUT EUROBLOCK connectors via each individual software controlled switch.

Analog Output Specifications

Output terminal	Actual Source impedance	For use with nominal	Output Level		Connector	Balanced / Unbalanced
			Nominal	Max. before clip		
OUTPUT 1-8	75Ω	10kΩ Lines	+4dBu (1.23 V)	+24dBu (12.3V)	EUROBLOCK (5.08mm pitch)	Balanced

*In these specifications, 0dBu = 0.775 Vrms.
*All output DA converters are 24bit, 128times oversampling.(Fs=48kHz)

MTX Series / Input Expander / Output Expander

Simple solutions for commercial sound system design

Matrix Processor

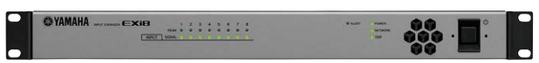
MTX3

- 26 x 8 matrix mixer and signal processor
- 8 mono microphone line inputs, 8 analog mono outputs
- Built-in SD card slot for audio file playback
- Up to 16 digital output channels via YDIF

Matrix Processor

MTX5-D

- 34 x 16 matrix mixer and signal processor
- 8 mono microphone line inputs, 8 analog mono outputs
- Built-in SD card slot for audio file playback
- Dante network support, and up to 16 digital output channels via YDIF
- One Mini-YGDAI expansion card slot



Input Expander

EXi8

- AD converter for input expansion
- Converts 8 channels of microphone/line input to YDIF for digital transmission to MRX7-D or MTX series units
- Preamps remotely controllable from MRX7-D or MTX series units

Output Expander

EXo8

- DA converter for output expansion
- Converts digital YDIF signals from MRX7-D or MTX series units to 8 channels of analog output

COMMERCIAL INSTALLATION SOLUTIONS

YAMAHA

SHARING PASSION & PERFORMANCE

*All specifications are subject to change without notice.
*All trademarks and registered trademarks are property of their respective owners.

VEGETABLE OIL INK This document is printed with Vegetable Oil Ink.

YAMAHA CORPORATION
P.O.BOX1, Hamamatsu Japan
www.yamahaproaudio.com

Printed in Japan
LPA649
P10027327

YAMAHA

SIGNAL PROCESSOR

MRX7-D

SIGNAL PROCESSOR

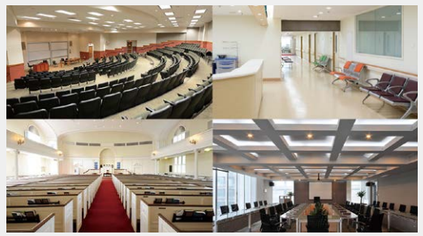
MRX7-D

Simple, flexible control for complex, high-capacity sound systems

Sound system requirements for corporate conference rooms, educational facilities, shopping malls, and a variety of other environments are becoming increasingly large and complex. The MRX7-D offers outstanding control efficiency and flexibility for a broad spectrum of audio installations.

Features for a Wide Range of Installation Needs

AEC (Acoustic Echo Canceller) for remote conferencing systems, automatic mixing for multiple microphone inputs, a speech privacy function that can mask conversations in hospital or corporate systems where confidential information must be protected, and more. The MRX7-D offers a comprehensive range of components that offer effective solutions for a variety of issues faced in today's audio installations.



Large Scale System Support

Built-in Dante digital audio networking capability makes it easy to set up systems of just about any scale while significantly reducing cabling cost and labor. Dante capable devices can be added to the system at any time, allowing flexible reconfiguration and expansion as needed.

Flexibility and Operation Ease

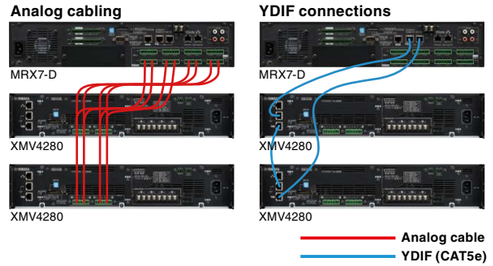
As the scale and control requirements of an audio system grow, the number of devices required and overall complexity increase correspondingly. With the MRX7-D and dedicated MTX-MRX Editor software, input and output devices can be quickly arranged via a software wizard and internal MRX7-D processing components can be combined as needed. One simple software application provides everything needed for comprehensive system design and management.

Ample I/O and Expansion Capability

- 8 mono line and 2 stereo line inputs, 8 analog mono outputs
- 64 in/out Dante capacity supports large digital audio networks
- I/O expansion via YDIF connected MTX series matrix processor, EXi8 input expander, and EXo8 output expander units

Dante

The MRX7-D, MTX series matrix processors, XMV series power amplifiers, and other Dante equipped devices can be easily connected via CAT5e or better Ethernet cables and gigabit switches to provide audio data networking in streamlined sound systems of any scale.



YDIF

YDIF is an original Yamaha digital audio protocol that allows standard CAT5e Ethernet cable to be used for bidirectional transfer up to 16 channels of audio plus word clock signals over distances of up to 30 meters. It allows cascading of multiple MTX series units as well as signal transfer to XMV series power amplifiers via connections that are fast and simple to set up.

The MRX7-D features one Mini-YGDAI card slot that can accommodate cards supporting a variety of audio formats, and expansion by up to 16 I/O channels.

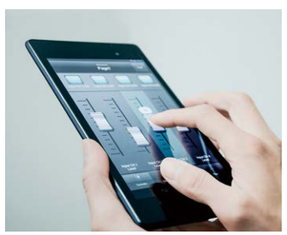
An SD memory card slot allows direct playback of MP3/WAV format audio files.

Broad Support for External Controllers

US standardized size models

DCP Series

Up to 8 DCP series control panel units can be daisy-chained over distances of up to 200 meters via CAT5e Ethernet cable. Power is also supplied via the CAT5e cable, allowing control panels to be installed at any convenient locations.



Wireless DCP

Wireless DCP allows remote control from an iPad, Android smart device, or other compatible device. This is an easy, intuitive way to control volume, select presets, and control SD player operation from a convenient location.

Available on the App Store GET IT ON Google play

*Wireless DCP can be downloaded from Apple's App Store and Google Play at no charge.(Apple, iPhone and iPod are trademarks of Apple Inc., registered in the U.S and other countries. App Store is a service mark of Apple Inc. Android and Google Play are trademarks of Google Inc.).

AMX / CRESTRON

The MRX7-D includes RS232C and Ethernet connectors that facilitate connection to a variety of external controllers such as those manufactured by AMX and CRESTRON.

ProVisionaire Touch

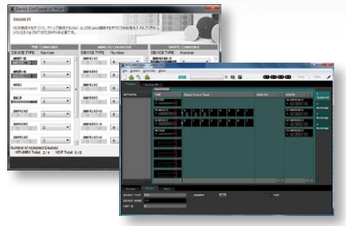
This handy iPad application allows MTX/MRX system control via a Wi-Fi network. A variety of control widgets can be arranged as required to create custom control panel layouts that are ideally configured for individual systems.

Available on the App Store

*ProVisionaire Touch can be downloaded from Apple's App Store at no charge.(Apple, iPad are trademarks of Apple Inc., registered in the U.S and other countries. App Store is a service mark of Apple Inc.).

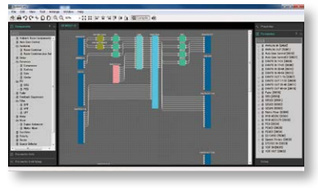
MTX-MRX Editor

The MTX-MRX Editor software facilitates sound system design with a workflow that simulates standard installation procedure. An easy-to-use wizard lets you start by arranging the devices to be used in the desired layout, then processing components can be added wherever needed. MTX series matrix processors, EXi8 input expanders, EXo8 output expanders, XMV series power amplifiers, and DCP digital control panels can be added according to the needs and scale of the application. The MTX-MRX Editor provides comprehensive, intuitive tools for designing sound systems from input to output.



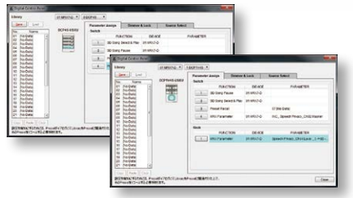
Device Layout Wizard

In addition to the MRX7-D, the wizard allows MTX series matrix processors, XMV series power amplifiers, DCP digital control panels, and EXi8/EXo8 input and output expanders to be added and arranged as needed.



Component Configuration

The MRX7-D features a versatile range of built-in processing components: faders, ON/OFF switches, matrix source selectors, GEQ, Dugan automatic mixing, speech privacy, echo canceller, and more. Simply select the required components and arrange them via a graphical interface.



External Controller Setup

Parameters that will be available to the end user can be specified via the MTX-MRX Editor. A variety of external controllers can be accommodated to ideally match the requirements of each installation.

Component
Ambient Noise Compensator *Gap type
Auto Gain Control
Combiner(Room Combiner,Room Combiner plus Automixer,Delay)
Dynamics(Compressor,Ducking,Gate,Limiter)
EQ(GEQ,PEQ)
Fader
Feedback Suppressor *Notch type
Filter(HPF,BPF,LPF)
Meter
Mixer(Dugan Automixer,Matrix Mixer)
Oscillator
Polarity
Router
Source Selector
Speaker Processor
Speech Privacy

Ambient Noise Compensator (non Gap type)
* Supported by version 2.1 and later

Acoustic Echo Canceller * Supported by version 2.1 and later

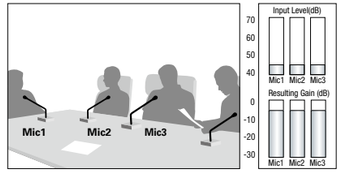
Conference Rooms, Banquet Halls, Hospitals, and More: Built-in Solutions for Diverse Needs

Dugan Automixer

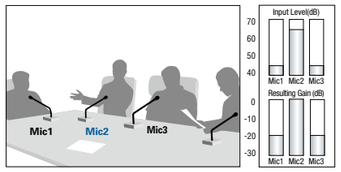
Dan Dugan Sound Design

Prevent feedback and unwanted pickup with multiple microphones

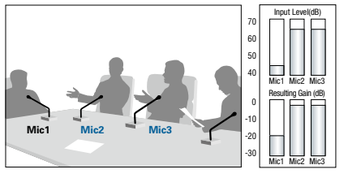
Feedback and unwanted noise pickup can be difficult to control at corporate meetings and panel discussions where multiple microphones are used, often resulting in impaired sound quality and intelligibility. The Dugan Automixer automatically increases the gain of microphones that are in use while reducing the gain of unused microphones. It also keeps track of the relationship between individual inputs and the overall input level to ensure ideally consistent mixer gain. A safe feedback margin and optimum signal-to-noise ratio are maintained at all times.



When nobody is speaking the input level at all microphones is low and the gain is equal. Rather than being muted, gain is distributed equally between all microphones.



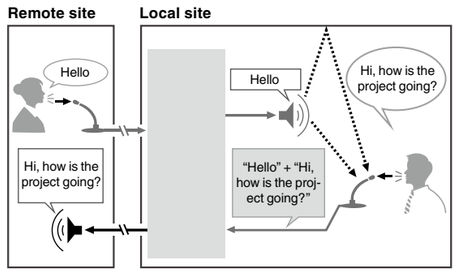
When one person is speaking the gain of that person's microphone is instantly increased to 0dB while the gain of the remaining microphones is lowered. The same occurs when any other one person is speaking.



If two people speak simultaneously the gain distribution between the active microphones is adjusted to achieve consistent overall gain, while the gain of the remaining microphone is lowered.

Acoustic Echo Canceller

Eliminate remote conferencing echo for maximum intelligibility



In remote conferencing situations speech from the remote location emanating from the local speaker can be pickup up by the local microphone and returned to the remote location in addition to the local speech. This type of echo can significantly degrade the intelligibility of the sound heard at the remote location.

The Acoustic Echo Canceller effectively suppresses this type of unwanted echo, maximizing clarity and intelligibility. The sound received at the microphone is compared with the sound emanating from the speaker and analyzed. An adaptive filter is employed to subtract the speaker output from the microphone input, preventing the speaker output from being returned as an echo by the microphone. Please note that this function is not designed to eliminate the room's own acoustic reflections or reverberation.

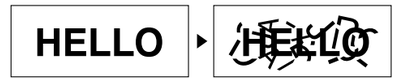
* The Acoustic Echo Canceller is available in version 2.1 and later.

Speech Privacy System

Keep confidential information in the conference room

The need to protect privacy in conference rooms and meeting areas, particularly in financial or medical institutions, is a growing concern. The Speech Privacy System uses an original sonic masking technique to prevent leakage of confidential information.

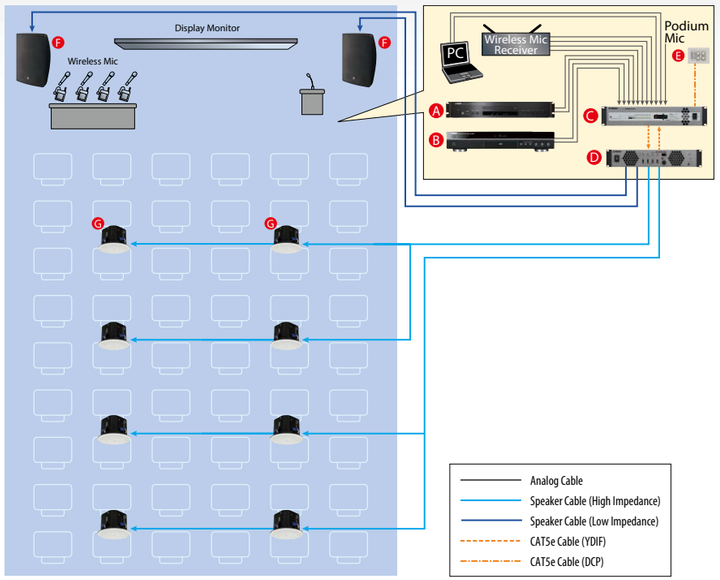
Rather than masking the conversation with high volume noise, an "information masking tone" synthesized from elements of human speech using an original process effectively camouflages spoken information. This approach allows the information to be protected by relatively low volume masking, helping to maintain the comfort of the conference environment.



Sound synthesized from human speech renders conversation unintelligible to eavesdroppers. This function does not silence the sound of the conversation.

Systems

Dugan Automixer



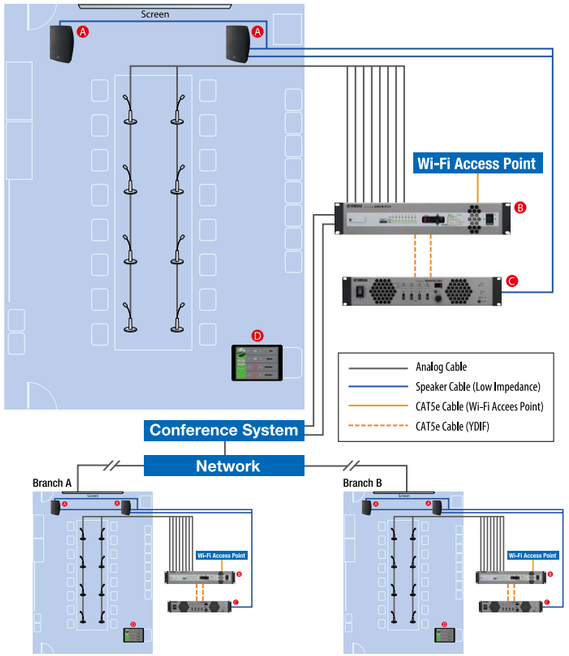
Applications

Conference center

Product List

A	CD Player	CD-S300RK
B	Blu-ray Disc Player	BD-S667
C	Signal Processor	MRX7-D
D	Power Amplifier	XMV4280
E	Control Panel	DCP4V4S
F	Surface Mount Speaker	VXS8
G	Ceiling Speaker	VXC6W

Acoustic Echo Canceller



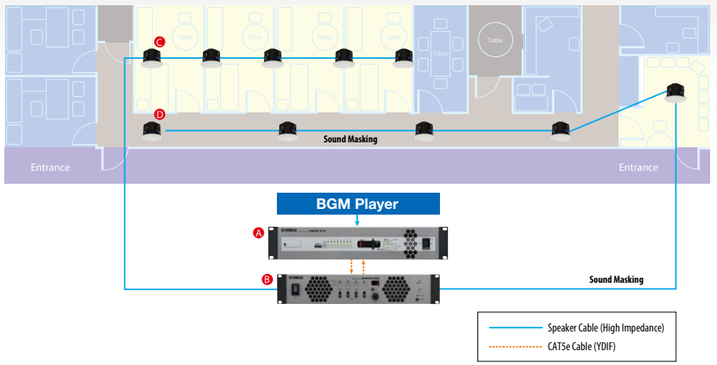
Applications

Conference rooms (remote conferencing)

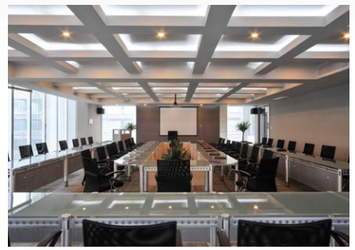
Product List

A	Surface Mount Speaker	VXS8
B	Signal Processor	MRX7-D
C	Power Amplifier	XMV4280
D	iPad	ProVisionaire Touch

Speech Privacy System



Applications



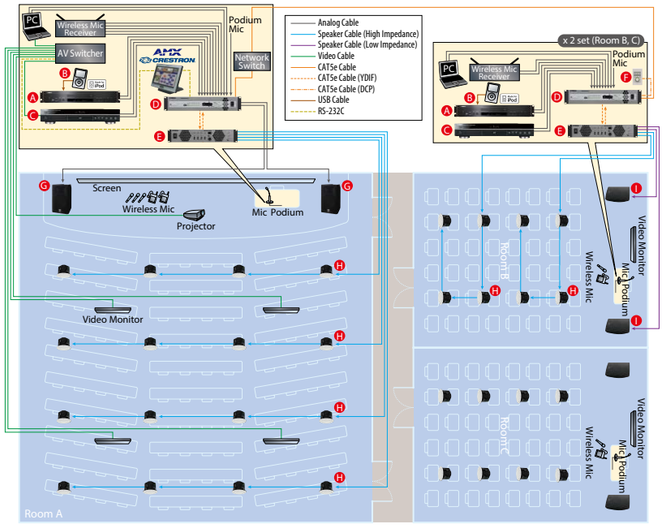
Corporate meeting rooms

Hospitals

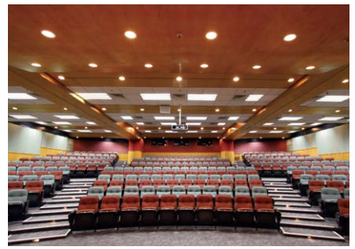
Product List

A	Signal Processor	MRX7-D
B	Power Amplifier	XMV4180
C	Ceiling Speaker	VXC6W
D	Ceiling Speaker(Sound Masking)	VXC6W

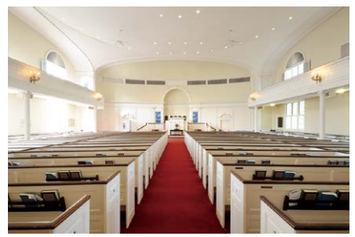
Dante Digital Audio Networking



Applications



Theaters



Houses of Worship

Product List

A	CD Player	CD-S300RK
B	iPod or other MP3/WMA Player	
C	Blu-ray Disc Player	BD-S667
D	Signal Processor	MRX7-D
E	Power Amplifier	XMV4280
F	Control Panel	DCP1V4S
G	Active Loudspeaker	DXR10
H	Ceiling Speaker	VXC6W
I	Surface Mount Speaker	VXS8