Bricasti Design

M1 Dual Mono D/A Converter USB

User Guide

V1.22 release 9/14

Conformity

EMC / EMI

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in residential installations.

Canadian Customers

This Class B digital apparatus complies with Canadian ICES-003.
Cet appareil numerous de la classe B est conforme a la norme NMB-003 du Canada.

Certificate Of Conformity

Bricasti Design, 123 Fells Ave., Medford MA, USA, hereby declares on its own responsibility the following products:

M1 –Dual Channel A/D Converter

-that is covered by this certificate and marked with the CE-label conforms to the following standards:

EN 60065	Safety requirements for mains operated electronic and related apparatus for household and general use
EN 55103-1	Product family standard for audio, video, audiovisual and entertainment lighting control apparatus for professional use. Part 1: Emission
EN 55103-2	Product family standard for audio, video, audiovisual and entertainment lighting control apparatus for professional use. Part 2: Immunity

With reference to the regulations in the following directives:
73/23/EEC, 89/336/EEC

January 2012
Brian S Zolner
President

Introduction

This is a preliminary edition of the M1 user guide covering theory of design and setup and use. In the future you can always find the latest version available at our web site www.bricasti.com.

Congratulations on the purchase of your new M1 Dual Mono Digital to Analog Converter. We at Bricasti Design have set out to design the world's best digital processors and to offer the finest products made for the professional and consumer audio markets.

Product Overview

The M1digtal to analog converter is a dual mono design; there are 2 completely isolated channels, a left and right, each with its own dedicated linear power supply, D/A converter, DDS clocking, and analog circuitry. This design insures that analog cross talk is virtually non existent, that the necessary power requirements for each channel are well met and isolated from each other and the digital processing is isolated, having its own power supply. With our twin DAC design, the dynamic range for each channel is optimized by using the stereo ADI 1955 D/A converter in a mono configuration, plus clocking is for each channel done directly at each DAC with a technique called DDS (direct digital synthesis) which takes clock induced jitter to immeasurable levels.

Build Quality

The M1 is robustly constructed of milled and CNC machined aluminum sections. There is no typical bent metal chassis and top cover found on most products. All sections of the construction, the front and rear panels, the sides and even the bottom and top plates start out as solid blocks of aluminum which are precision machined to shape, with exact tolerances for a perfect fit. These parts are then anodized and the text and markings are laser etched for a clean and enduring look.

The Sound

The intention of the M1 is to provide a state of the art, Digital to Analog converter, utilizing the best designs and materials that can be found today. The D/A converter is a very critical part of the digital audio chain, after all you have to convert it to analog to hear it, and we feel this should be a true as possible in its reconstruction of the original signal. The sound of the M1 is intended to be transparent and revealing, and fully dynamic. This in part is made possible by the lowering the jitter to extremely low levels, providing a pure digital signal chain with sample rate converters, superior digital filter design, coupled to a fast transparent analog signal path with a discreet analog output section and plenty of good clean linear power for optimum analog performance.

Many hours of listening were done to tune the M1 to an exacting sound, with all types of music, and with extensive testing done in the studio and in the home. We hope you find the M1 to be pleasing and enjoyable to hear and use in the home, or as a precision tool for high level reference monitoring for the professional.

Unpacking and Inspection

After unpacking the M1 save all packing materials in the event you ever need to ship the unit. Thoroughly inspect the M1 and packing materials for any signs of damage in shipment. Report any damage to the carrier at once.

Precautions

The Bricasti Design M1 is a rugged device with extensive electrical protection. However, reasonable precautions applicable to any piece of audio equipment should be observed.

- Always use the correct AC line voltage as set by the manufacturer. Refer to the power requirements section of the manual and adhere to any power indications on the rear or bottom of the chassis . Using the incorrect AC line voltage can cause damage to your M1, so please check this carefully before applying power.

- Do not install the M1 in an unventilated rack or directly above any heat-producing equipment like power amps, tube preamps etc. Maximum ambient operating temperature is 40 C. Exceeding the maximum ambient temperature may cause the M1 to enter thermal shutdown and stop processing sound as a safety precaution, and may cause damage to the internal processors and components.

- To prevent fire or shock hazard, do not expose the M1 to rain or moisture.

Notices

In the interest of continued product development, Bricasti Design reserves the right to make improvements to this manual and the product it describes at any time and without notice.

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Important Safety Instructions:

Notice!

- Read these instructions.
- Keep these instructions.
- Heed all warnings.
- Follow these instructions.
- Do not use this apparatus near water.
- Clean only with dry cloth.
- Do not block ventilation openings; install in accordance with manufacturer's instructions.
- Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers, pre amps) that produce heat.
- Do not defeat the safety purpose of the polarized or grounded type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade and prong are for your safety. If the provided plug does not fit in your outlet, consult an electrician for replacement of the obsolete outlet.
- Protect power cord from being walked on or pinched.
- Use only attachments/accessories specified by the manufacturer.
- Unplug this apparatus during lightning storms or when unused for long periods of time.
- Refer all servicing to qualified service personnel. Service is required when the apparatus has been damaged in any way, such as by being dropped, exposed to rain, liquid being spilled on it, or otherwise does not operate normally.

Service

- There are no user serviceable parts inside.
- All service must be performed by qualified personnel.

Warning!

- To reduce the risk of fire or electrical shock do not expose this equipment to dripping or splashing water and ensure that no objects such as vases are placed on the equipment.
- This apparatus must be earthed.
- **This equipment requires the correct AC line voltage as set by the manufacture and is not auto sensing or scaling.**
- Use a three-wire grounding-type line cord like the one supplied with this product.
- Be aware that different operating voltages require the use of different types of line cords and attachment plugs.
- Check the voltage in your area and use the correct type. See table below:

Voltage	**Line plug standard**
110-125V	UL817 and CSA C22.2 no 42
220-230V	CEE 7 page VII, SR section 107-2-D1/IEC 83 pg C4
240V	BS 1363 of 1984 Specification for 13A fused plugs and switched and unswitched outlet plugs

- This equipment should be installed near the socket outlet and disconnection of the device should be easily accessible.
- To completely disconnect from AC mains, disconnect the power supply cord from the AC receptacle.
- Do not install in a confined space.
- Do not open the unit -risk of electrical shock inside.

Caution

- You are cautioned that any change or modification not expressly approved in this manual could void your authority to operate this equipment.

Design Overview

There are 4 basic sections to the M1, the digital input section, and the left and right analog sections, and the front panel:

Digital Input Section:

This is located in the center of the unit, provides 5 transformer isolated digital inputs, are selectable from the front panel, and has its own linear power supply. This means that the digital processing section is isolated from the analog sections, providing excellent low noise performance and eliminates digital noise from entering the analog chain via the power supplies and ground plane. This section features an Analog Devices Sharc DSP that is used to run the front panel and general operations of the M1, to control and synchronize the DDS clocking on each channel, and to provide a selection of our own over sampled anti-aliasing filters.

Analog Output Sections, Left and Right:

These are identical and are laid out as mirror images of each other to fit with the over all symmetrical industrial design of the M1. Both are independently powered by their own linear power supply insuring clean double regulated low ripple power and isolation from any digital switching noise from the digital supply.

Each section has its own Analog Devices 1955 DAC, coupled with a dedicated DDS clocking circuit located millimeters away from the DAC, assuring extremely low jitter and minimal trace length for the clock signal. As both boards have their own clock, precise clock synchronization of the left and right boards is handled by the Sharc DSP on the main digital processing board.

The next stage, at the analog out of the converter for the gain and filter sections there is a fully differential analog design with fast high slew rate analog opamps. This is followed by 2 transistor designed output buffer sections, one balanced and one unbalanced, each separately buffered and isolated. The balanced output gain is precision adjustable via a back light illuminated set screw near the XLR connector at the rear panel. This gain is adjustable from +8 to +23 dbm and can be reference level set to a fraction of a db to match any setup. As a default this is set in our test lab at +16 dbm. The unbalanced is set to normal hi fi levels of 2V RMS (+8dbm) by precision resistor values on the board.

Typical circuit boards in most products are made from FR4 fiber glass. But, the M1s analog boards are made from a substrate called Arlon. This material has excellent very high frequency impedance characteristics, and was chosen for use in the M1 to yield an open and clear sound and to allow its very high slew rate audio circuit design to perform at optimal levels.

Trigger In:

On the rear panel the M1 has a stereo connector (Tip/Ring/Sleeve) for triggering the M1 into standby mode from an external device like a preamp and for optional external remote control with the M1 remote. Sleeve is connected to chassis ground, Tip/Ring is the input +/-. The M1 will go into standby when it has a positive 5V or 12V DC voltage between tip/ring.

Front Panel Overview

The front panel has a large, simple, easy to read display, an encoder for adjusting and selecting settings, 6 keys that are labeled for their use, and a power stand-by switch that will set the M1 in to low power mode and mute the outputs.

Rear Panel Overview

Looking at the rear you will find on the left and right side the analog output sections, each with its own balanced and unbalanced outputs, and level adjustment set screw. In the center input section are the 4 digital audio inputs, AES, SPDIF 1 and 2, Toslink, and USB. There is a small jack below the fuse and this is for a trigger input to remotely place the M1 in standby from a pre amp or other system controller. The main power on/off switch and AC fuse are at the rear, note that the front panel and the trigger in are used to set the M1 to stand by. Full power on off is done from the rear panel.

Setup and Operation

AC power and the M1

The AC power is connected at the rear of the unit; the filtered AC inlet also has the main power on-off switch. This filtered inlet helps provide clean AC power to the M1's power supplies and as well will prevent any digital noise from the M1s digital processing section from going back out the AC inlet to contaminate the mains. Take note that because the M1 utilizes linear power supplies care should be taken to use only the power range indicated on the unit, other wise damage can occur to the power supplies and other circuits in the M1. Please note and adhere to any voltage indications on the outer box, rear panel or chassis all of which will indicate how the M1 is set at manufacture.

Note that the main AC power switch is at the rear and the front panel switch is a low power consumption stand by switch. For complete power on of you must cut power with the rear panel switch or from an external AC power on off switch that may be used to power other devices in your setup.

Quick Connecting the M1 and power up

When you first power up the M1 it will come up in AUTO mode for the input select. This means that all you need to do is connect the digital output from the source device to one of the appropriate M1 digital inputs. The M1 will sense the digital carrier signal and automatically select it Note if multiple sources are connected then you will have to manually select inputs per below as Auto will not detect what source is playing audio.

Operating the M1

There are 6 front panel keys, input, filter, status, level, display and enter

- **Input select**
  When the M1 first powers on, it will default to the STATUS page display on the front panel. This will show what input is selected and the sample rate. Pressing the INPUT key will take you to input select mode. If you turn the knob you will scroll though all inputs. The display will flash if the input type displayed is not active. If you want to select it press enter. Inputs are:
  - #1 AES Selects the XLR connector
  - #2 SPDIF Selects the RCA connector
  - #3 EIAJ Selects the Toslink connector
  - #4 USB Selects the USB connector .
  - #5 SPDIF Selects the BNC connector
  - AUTO Selects the M1 to auto mode for automatic selection of the input
- **Status**
  The status display key has 5 levels in the menu. On first press or on power up it will display input type selected and the running sample rate. For PCM this will range from 44.1k to 384K and DSD will simply display DSD for DSD 64fs or DSD 2 for 128fs.

  Pressing the status key from any other state will take you to the input and sample rate display page.

  Pressing STATUS a second time will take you to a display of the actual output sample rate. In most cases this will be 352.8k as this is 8 x 44.1k of the CD sample rate. If the source is DSD then 2.822 or 5.644 Mhz will be displayed.

  Pressing STATUS a third time will toggle to the digital over tracking display. This shows the left and right channel digital overs that have occurred since entering this display page, leaving the error counter page by pressing any other key will erase the counters and reset them to 0.

  Pressing a fourth time will show an internal temperature monitor, there are no adjustments as this is just a monitor of internal temps.

  Pressing a fifth time will bring you to the Phase Invert control. The M1 is absolute phase meaning that it does not invert the phase. With this adjustment you can invert the phase of the signal as some recordings may benefit by this change or correction. Normal setting is NON inverted.
- **Display**
  This allows you to set the display intensity in 3 levels and set it to a sleep or off mode. Press DISPAY and use the knob to select the brightness of the display, press ENTER to set it. Selecting OFF will shut the display off after a 20 sec time out, leaving one LED dimly lit. Pressing any front panel key will wake up the display so you can make adjustments to the M1, and then after a short period of no use, it will go to off mode again.
- **Filter**
  Pressing FILTER will take you to the filters select mode. Here you can select 9 different types of linear phase digital over sampling filters, labeled Linear 0-8 and 6 minimum phase filters labeled Minimum 0-5 Press FILTER, turn the knob to change, press ENTER to select. These are loaded immediately with very little delay, so the change is very fast, allowing for quick comparisons of their effect on the sound.

- **Level**
  The M1 USB includes a digital level control. For many applications this feature will allow the M1 to be used directly to the power amp, eliminating the need for an analog line pre amp, providing perfect level adjustment control for the outputs of the M1. It is a digital adjustment so it will affect both the balanced and unbalanced outputs exactly the same and insures perfect channel balance at all gain settings. Operation is simple: press Level and it will display the level in db, normally this will be set a 0db. Turn the knob and you can cut the level in one db steps. Pressing the Level key a second time will set the output to MUTE, pressing again will un-mute. Upon power up, if the M1 was left in any level state other than 0db, it will power back at the last setting. If you do not use the level adjustment and leave the setting a 0db, the unit will power on to the Status page.
- **Enter**
  This sets or selects values in the other menus.

USB Features

On the rear panel you will find the USB 2 type interface and it is based on the latest generation of asynchronous design and supports sample rates up to 384k/24 bit. For superior noise performance the interface is electrically isolated from the host computer, eliminating any grounding or power induced noise issues that could be transmitted to the M1 from the computer. No driver is needed for Macs or Linux but for PC use a driver is necessary and the latest version for Win 8 support can be acquired from our web site and or supplied with the M1.

DSD playback and the M1

DSD playback with the M1 is quite simple, it is done with DoP and as such can be read with any input, but in most common is the USB as there are few disk players that play out DSD as DoP via the AES or SPDIF from the SACD layer. DoP, or DSD over PCM, is the DSD data embedded in a 176.4k 16 bit PCM data stream with the extra 8 bits out of the 24 bits used for identifying that it is DSD not PCM. This is true DSD and not PCM conversion. When using a computer audio setup, the media player will send out the DSD as DoP , the M1 will see that as 176.4k pcm for DSD 64fs or 352.8k for DSD 128fs, read the data header and see that it is actually DSD and unpack the data in our digital signal processor as the original DSD data, and send it out to the DAC for conversion to analog.

To use it all you need is DFF files, set the media player to play the files as DoP, and the M1 will play them. When DSD is received for playback the status and filter displays will read DSD, and when it next plays a PCM file it will revert to your last used PCM or DSD filter and display will update accordingly. Playback is seamless as any other PCM sample rate change.

The current version supports both DSD 64, one bit at 64 times 44.1k sample rate, and DSD 128 or double that rate. DSD 64 is the SACD standard and 99% of all content is released an mastered at this rate. There is some content appearing that is DSD 128 or 2x the rate but for the most part you will find DSD 64 as the standard.

We also implemented the ability to have DSD post noise filtering. An artifact of DSD processing is the buildup of ultrasonic noise and with DSD 64, this noise starts at 24khz and rises to peak level at 50k and beyond. Ideally this should be removed in the digital domain before any analog stages, and we offer a few options.

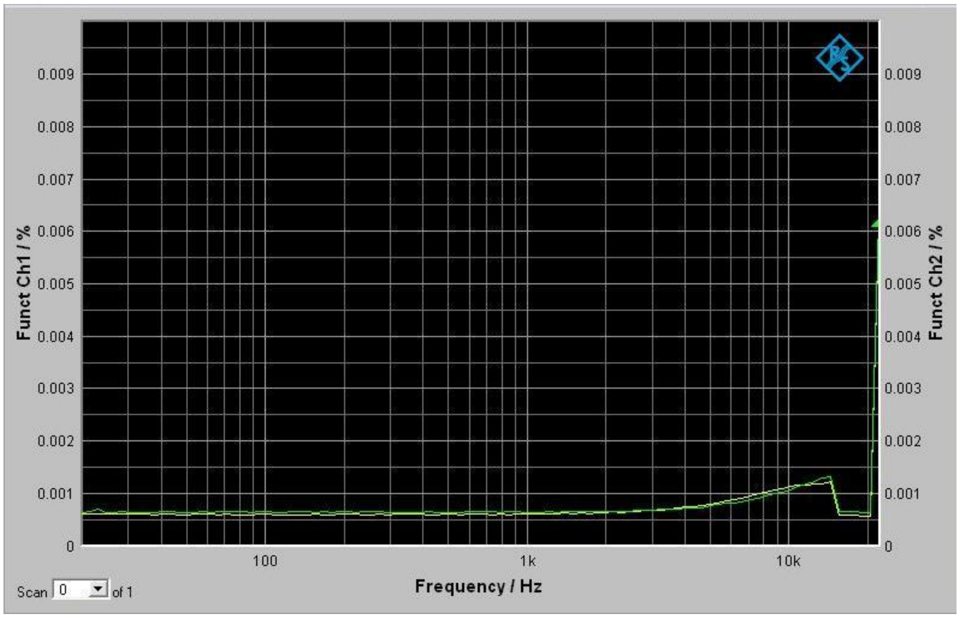
Filter 0: This filter has no post filter and allows any ultrasonic noise to pass thru. There is a belief that the noise would be filtered by other things in the chain, and is in effect benign, so we created this option to let the user decide.
Filter 1: This filter starts at 32k pass band and ends at 64k stop band, so there is less attenuation of the noise, a gentler one, so a compromise filter that might allow some noise to pass.
Filter 2: This filter starts at 28k pass band and ends for full attenuation or stop band at 48k, so good attenuation of noise takes place. Technically this is a very good filter for bandwidth and noise removal which is why it was chosen for the V1.15 first release of DSD playback.

A great source for DSD downloads is: www.channelclassics.com

Specs and Performance

Audio Performance

The typical audio performance spec of the M1 is outstanding; Total Harmonic Distortion is an impressive 0.0006% at all frequencies.



Digital Oversampling Filters

There are 9 Linear Phase filters and 6 Minimum Phase Filters that can be selected for use in the M1 and they are selected via the filter menu on the front panel. These are labeled Linear 0-8 and Minimum 0-5 in the user interface. Since the M1 converters use over sampling techniques, these filters are calculated at a very high rate for very high precision filter construction. .

Note: The the M1 uses delta sigma 8 x oversampling conversion so it is not recommended to “up-sample” the digital audio that is being sent to the M1. Defeat all up-sampling features in your media player or CD transport. Up-sampling the data before the M1 will yield poor results and always use the original source audio bit and sample rates, so for example if the source is 44.1k then send this data unprocessed to the M1and let the M1 reconstruct the data correctly.

You will find despite the close numbers of the specs, that they all have very unique and different sound characteristics, and you may find one more suitable for different kinds of music than others as well. Here is a brief description of them. Note that since both types have the identical characteristics that there is one table of characteristics for both filer types

Filter Descriptions

For filters 0-5 this chart is accurate for both Linear Phase and Minimum Phase filters.. The 6 Minimum phase filters have the same characteristics as the linear phase filters of the same number, allowing easy comparison between filter types. As an example, Linear 2 has the same basic frequency response characteristics as Minimum 2 but they are based on a different filter construction techniques and yield different results. Filter types 6-8 are only available as linear phase. For simplicity and as a general guide to their characteristics, the list below lists filter # but in the M1 menu they are called out as Minimum and Linear.

- Filter 0 - 20kHz bandwidth, Stop-band at Nyquist frequency with low ripple and high attenuation
- Filter 1 - Low delay filter with full attenuation at Nyquist Frequency
- Filter 2 –Same as # 1 with a gentler slope and the passband at 19.5kHz
- Filter 3 - Same as # 1 with a gentler slope and the passband at 19kHz
- Filter 4 - Same as # 1 with a gentler slope and the passband at 18.5kHz
- Filter 5 – Same as # 1 with a gentler slope and the passband at 18kHz
- Filter 6 - A halfband type filter with 6dB attenuation at Nyquist frequency
- Filter 7 - Similar to 0 with a slightly gentler slope filter at 19k
- Filter 8 - Steepest slope, highest bandwidth, with low ripple and high attenuation

Filter Tables

Table of filter characteristics at 48khz.

48khz	Passband	Stopband	Passband ripple	Stopband attenuation	delay
Filter 0	20kHz	24kHz	.001dB	111dB	.73ms
Filter 1	20kHz	24kHz	.005dB	102dB	.63ms
Filter 2	19.5kHz	24kHz	.004dB	103dB	.56ms
Filter 3	19kHz	24kHz	.005dB	102dB	.51ms
Filter 4	18.5kHz	24kHz	.003dB	106db	.50ms
Filter 5	18kHz	24kHz	.001dB	114dB	.50ms
Filter 6	21.8kHz	26.3kHz	.0002dB	110db	.72ms
Filter 7	20kHz	24kHz	.001dB	110dB	.7ms
Filter 8	20kHz	24kHz	.001dB	110dB	.7ms

Table of filter characteristics at 44.1khz

44.1khz	Passband	Stopband	Passband ripple	Stopband attenuation	delay
Filter 0	20kHz	22.05kHz	.001dB	110dB	1.43ms
Filter 1	20kHz	22.05kHz	.161dB	71dB	.72ms
Filter 2	19.5kHz	22.05kHz	.046dB	82dB	.72ms
Filter 3	19kHz	22.05kHz	.014dB	92dB	.72ms
Filter 4	18.5kHz	22.05kHz	.005dB	102db	.72ms
Filter 5	18kHz	22.05kHz	.001dB	112dB	.72ms
Filter 6	20kHz	24kHz	.0002dB	111db	.78ms
Filter 7	19kHz	22kHz	.001dB	110dB	1ms
Filter 8	20kHz	22kHz	.001dB	110dB	1.5ms

Technical Specifications

Digital Inputs

Connectors:	XLR: AES/EBU 24 bit Single Wire
	BNC: SPDIF
	RCA: SPDIF
	USB: USB 2
Sample Rates AES, SPDIF, AUX in:	44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192khz
Sample Rates USB:	44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192khz 352.8kHz, 384kHz, DSD 64fs 128Fs as DoP
Jitter:	8 psec @ 48k / 6psec @ 96k

Balanced Analog Outputs

Connectors:	XLR balanced (pin 2 hot)
Impedance:	40 ohm
Output Level Range:	+8 dbm to +22 dbm
D/A Conversion:	24 bit delta sigma 8x oversampling
Frequency Response @44.1k:	10 hz- 20 kHz +0dB, -.2 dB
Dynamic Range:	>120dB A-Weighted
THD+N @ 1k:	.0006% @ 0dbfs / .0004% @-30dbfs

Unbalanced Analog Outputs

Connectors:	RCA
Impedance:	40 ohm
Output level:	+8 dbm (2V RMS)
D/A Conversion:	24 bit delta sigma 8x oversampling
Frequency Response @ 44.1k:	10 hz- 20 kHz -.2 dB
Dynamic Range:	>120dB A-Weighted
THD+N @ 1k:	.0006% @ 0dbfs / .0004% @-30dbfs

General Specifications

EMC

Complies with:	EN 55103-1 and EN 55103-2 FCC part 15, Class B

RoHS

Complies with:	EU RoHS Directive 2002/95/EC

Safety

Certified to:	IEC 60065, EN 55103-2

Environment

Operating Temperature:	32 F to 105 F (0 C to 40 C)
Storage Temperature:	-22 f to 167 F (-30 C to 70 C

General

Finish:	Anodized Aluminum
Dimensions:	17” x 12“ x 2.5”
Weight:	12 lbs
Shipping Weight:	15 lbs
Shipping Dimensions:	22”x 17”x 7”
Mains Voltage:	100, 120, 220, 240 VAC, 50 Hz – 60 Hz
AC inlet fuse:	T1A 250V slow blow for all voltages/frequencies.
Trigger In:	TRS connector for 5V external trigger.
Power consumption:	28 Watts (6W standby
Warranty parts and labor:	2 years

Bricasti Design