MYTEK DIGITAL

8X96 Series ADC and DAC
User Manual

Rev. 9/2004

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Contents:

8X96 Series Converters – General description

The MYTEK 8X96 series consists of two 2U units- the 8X96 ADC and the 8X96 DAC. They provide very high performance 24 bit 96 kHz conversion comparable to the most expensive mastering converters (DR=120 dB, audiophile signal path). These units can be used as stand-alone devices for variety of studio applications such as 5.1 DVD mastering, multitrack recording and whenever high quality conversion is required.

Users choose Mytek 8X96 converters primarily for their outstanding sound quality. The sound of Mytek converters can be described as "transparent". We design our converters to be as faithful to the signal as possible, rather than the philosophy of some other manufacturers who offer "analog" or "tube" sounding converters. Mytek converters are closest to a straight wire, which is especially evident when used at full 24/96 resolution- our tests have shown that 90% of listeners cannot hear the converters' impact on the sound at all.

On Mytek website you can find and download various sound samples to evaluate the 8X96 sound quality and compare it to the sound of other high end converter units.

Log onto: http://www.mytekdigital.com/8x96compare.htm to download samples.

Thanks to the wide choice of daughter interface cards (MDM), 8X96 Series becomes an 8 channel "digital swiss knife" which allows the user to leverage converter performance to most studio setups and situations. Mytek is the first converter company that offers practical interfaces to record 24/96 format on Tascam DA78HR , for example. The units perform extremely well in a high end environment of Sonic Solutions HDSP ™ or Sadie 2496™ or workstations but can be equally beneficial for an ADAT owner who can now make use of 24 bit 96 kHz technology to dramatically improve ADAT sound. Even when used in 16 bit mode, the 8X96 converters produce 20dB (10 times) less distortion than standard ADAT converters.

The 8X96 features include:

- 44.1, 48, 88.2 and 96kHz internal sampling frequencies (ADC)
- Ext. wordclock or AES11 25-100kHz sync
- 120dB Dynamic Range
- 24 bit resolution
- High performance SuperShaper-HR™ psychoacoustic noiseshaping algorithm for 16 or 20 bit output (ADC)
- ADAT<>AES/EBU<>TDIF<>Sonic<>Protools digital format conversion
- Prism MRX™ bit splitting and S/MUX™ sample splitting for ultimate flexibility in encoding hi-resolution sound on standard 16 and 20 bit equipment.

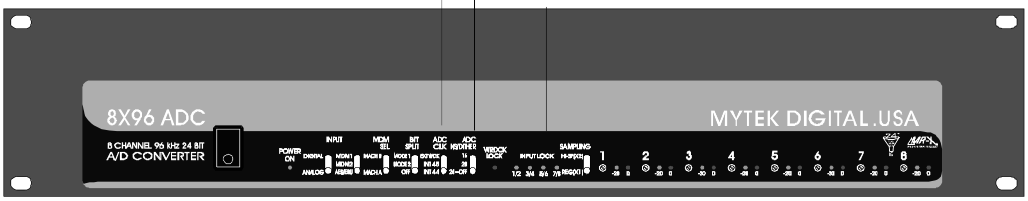
The combination of great sound and interfacing capabilities make the 8X96 Series an ideal choice as a high quality front end for any piece of digital studio equipment.

8X96 ADC – Quick Start

ADC sync clock can be either internal (preffered for sound quality) or external wordclock. The 75 Ohm termination resistor (default=off) can be switched on by internal jumpers next to the BNC jack.
For 88.2Khz select 44.1k and Hi-Speed X2 switch. Similarly 96kHz is 2x48kHz.
External wck for 88/96 kHz can be either 88/96 or 44/48 (see “dip switch”)

High performance SuperShaper HR ™ noise shaping algorithm provides extremely transparent and effective wordlenght reduction preserving practically all 24 bit resolution on 20 bit output and most of it on 16 bit out. See “Noiseshaping”.

Lock leds indicate lock of pairs of digital inputs. Note that for 8 channel DIO interfaces there is one common clock, AES pairs have to be synchronous to be properly converted to another digital format.



Input select switches
ANALOG = A/D converter

Some DIO interface cards such as ADAT and TDIF have 2 sets of 8 chan DIO - A and B. In regular speed mode, this switch selects either port A or B. In high speed mode and w/MRX
and S/MUX modes (see “dip switch”)

Mode of MRX codec - turn it off if you are not using MRX. Relevant to TDIF Only.

Sampling frequency range:
Reg= 25kHz - 50kHz
Hi= 50kHz - 100kHz
Switch automatically engages S/MUX on ADAT and TDIF

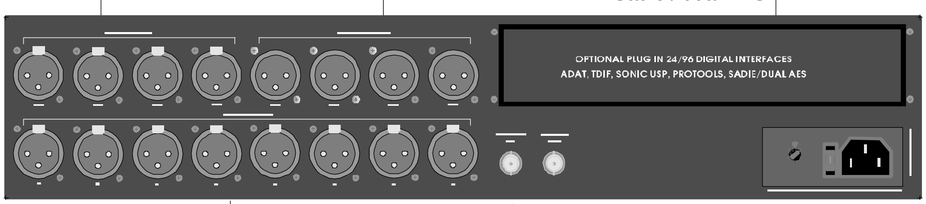
Input level adjustment. Nominal 0VU=+4dB=1.225VRMS=-18dBFS
Can be adjusted +12/-6dB

-20dB and -0dB peak leds display the peak signal level relative to 0 dBFS. Red means 1 or more samples=0dBFS

Hi-speed AES/EBU inputs lock to signals of 25-100kHz FS. Standard 110 Ohm, balanced, transformer isolated

Hi-speed AES/EBU outputs for signals of 25-100kHz FS. Standard 110 Ohm, balanced, transformer isolated implementation. As all outputs they repeat selected input. If “AES IN” is selected, the signal is a reclocked (jitter reduced) clone i.e. all bits and flags are exactly repeated. If other DIO is selected, certain flags may not be repeated.
(See “Digital format conversion”)

Up to **two** Digital Input-Output Interfaces can be installed (apart of standard AES/EBU). These interfaces are daughter cards manufactured by Mytek and can be:
-ADAT with S/MUX 96/24 recording
-TDIF with S/MUX 24/96 and Prism MRX Bitsplit
-Sonic HDSP/USP
-Protools ® compatible
-Sadie / dual AES



Analog inputs. Nominal balanced +4dB level. Consumer -10dBV level can be achieved by turning the level trimpots up and, connecting sleeve to pin 1 and pin 3 and tip to pin 2.

Wordclock is a TTL level signal = FS. Wck for 88/96 kHz can be either 88/96 or 44/48 (see “dip switch”)

This BNC jack outputs Wordclock- a TTL level signal = FS and phase locked to incoming sample clock. If AES/EBU input is selected the clock is stripped from pair #1/2. Wck for 88/96 kHz can be either 88/96 or 44/48 (see “dip switch”). A 75 Ohm terminating resistor can be switched in via jumper inside.

Setting up the 8X96 ADC is fairly straightforward - install the unit in the rack, and make all necessary connections. If you rack frame is grounded, you may consider evaluating inter-equipment ground connections and power cord grounding. If unbalanced audio is involved, connect the sleeve to pin 1 and pin 3 tied together and the ring to pin 2.
If an external house clock is used, make sure that the clocks are properly fed and terminated (75 Ohm) if necessary, especially for long cable runs.

8X96 Converters are microprocessor controlled and as such are subject to static interference. If unusual circumstances such as a lightning storm cause any malfunctions, reboot the unit. For further questions regarding setup and features contact your dealer or Mytek technical support.

8X96 ADC – Analog Input Alingment

Before using the 8X96ADC you need to align its analog inputs. The alignment generally does not affect audio quality, only input sensitivity. The alignment is performed using 10 turn trimpots on the front panel. A small tweaker or screwdriver is necessary.

You have to arbitrarily decide what will be your studio "0 VU" analog/digital reference level. It is usually between -20 and -14dB. It defines how much headroom you have left over the normal operating "0 VU" level. You may set it at the same level as other piece of equipment in your studio. For example Panasonic 3700 is fixed at -18dB (which is low for most rock and roll recordings). ADAT is fixed at -15dB which is an optimal level.

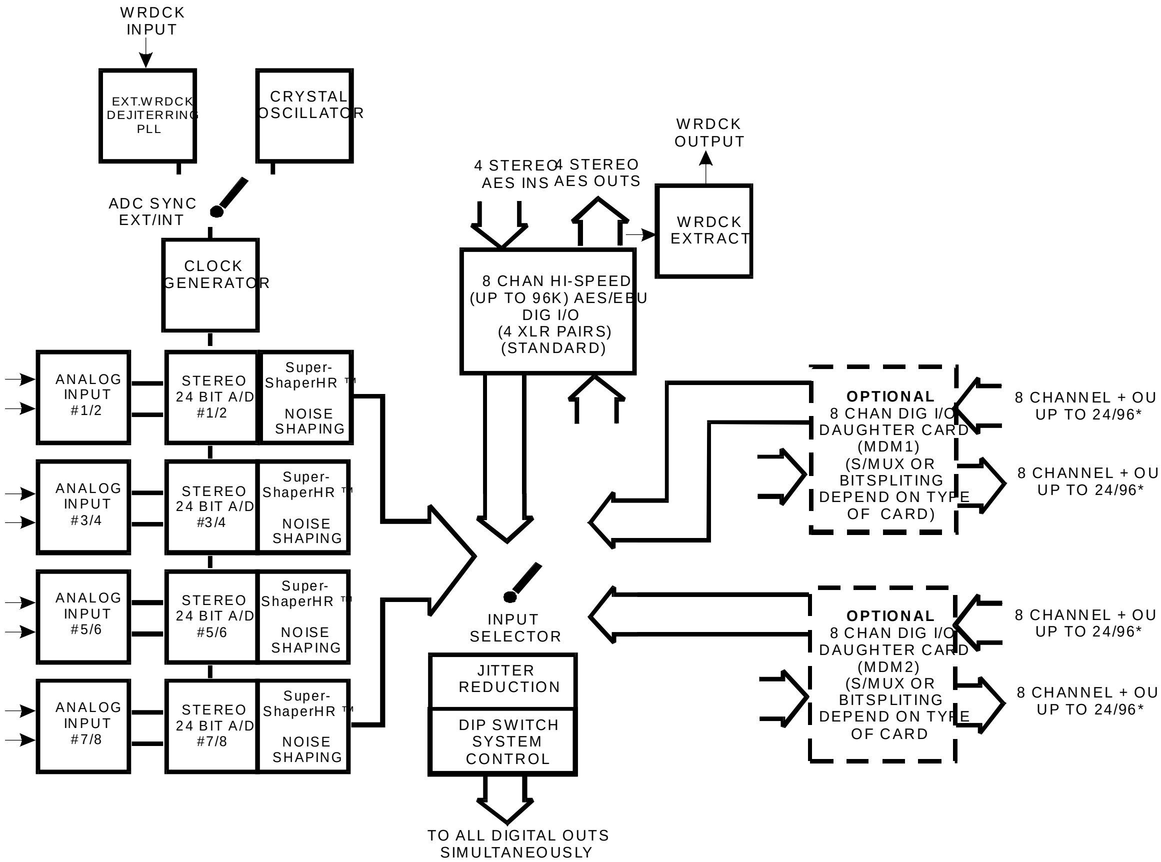
1.Set the oscillator in your console at 1kHz and "0 VU". Send the oscillator to the interface analog input."0 VU" at + 4dB corresponds to 1.225 Volts RMS measured between pin 2 and 3 of the output XLRs. If you do not have an analog oscillator, you can use a calibrated analog out of a DA converter and generate sinewave inside the DAW.

2.Select the "analog input". Connect a digital meter to a digital output. If you don't have a dedicated digital meter use the most precise meter available in you existing digital recording equipment or DAW.

3. Adjust the analog input level to get appropriate reading of the meter (for example -15dB).

4. Repeat the same for all analog channels.

8X96 ADC – Signal Flow



8X96 ADC - Features

Sampling Frequencies and External Synchronization

The Analog-Digital converters are implemented as 4 stereo pairs. They operate synchronously , i.e. all 4 converters always receive the same clock. As a result, all 8 digital outputs (or 4 stereo pairs of digital signals) are in sync and in phase.

The converters need a sampling clock of any frequency btwn. 25-100 kHz. If Analog Input is selected, the A/D converters are active and can be clocked by the internal crystal oscillator (set ADC SYNC =“Internal”) or external wordclock (set ADC SYNC = “Wrdck.”).

The internal crystal oscillator provides 4 sampling frequencies:

-44.1 kHz (Set SAMPL=”44.1” and SAMPLING=”Reg(x1)”)
-48 kHz (Set SAMPL=”48” and SAMPLING=”Reg(x1)”)
-88.2 kHz (Set SAMPL=”44.1” and SAMPLING=”Hi-Sp(x2)”)
-96 kHz (Set SAMPL=”48” and SAMPLING=”Hi-Sp(x2)”)

The SAMPL and SAMPLING set the sampling frequency of the ADC only, but they always have to be set properly as they are used to set the digital output sampling flags. When digital input is selected the incoming sample rate is maintained throughout the unit.

The 8X96 ADC has a robust, very low jitter (<10ps) crystal oscillator for sampling frequencies of 44.1, 48, 88.2 and 96kHz. For all tasks that don’t require slaving to an external clock, the use of this internal clock is recommended as it achieves the lowest possible jitter, resulting in maximum converter performance. It is recommended that external devices such as DAW or recorders be slaved to the A/D. If the converter is resolved to external wordclock, the clock jitter may increase depending on the stability of the wordclock source. Incoming clock jitter is attenuated but not eliminated. Thus the sampling clock quality is proportional to the quality of incoming wordclock. Therefore, the use of the internal crystal generator is recommended for highest quality. If the 8X96ADC is used to feed a DAW, we recommend setting the DAW clock source to “digital input” and setting the ADC to internal clock. This arrangement will provide the best performance. If external clock source must be used (e.g. for video work), a stable external wordclock generator is recommended. Proper 75 Ohm termination (set through internal jumpers) can reduce wordclock jitter, especially for long cable runs. (Note: make sure that the wordclock source is able to drive a 75 Ohm load).

8X96 ADC – Features

Digital Inputs, Outputs and synchronization

8X96ADC comes equipped with **hi-speed AES/EBU** digital output and input standard. Additionally up to two daughter interface (MDM) cards can be installed. There are several different MDM cards available for Mytek 8X96. They are the same for both ADC and DAC:

ADAT – Features 2 ADAT lightpipe inputs (2x 8chan@48k) and 2 ADAT lightpipe outputs. Incorporates S/MUX sample splitting for 4 channel of 24/96 through one lightpipe (8channel 24/96 max total)

TDIF – Features 2 TDIF inputs (2x 8chan@48k) and 2 TDIF outputs (2 TDIF connectors). Incorporates S/MUX sample splitting for 4 channel of 24/96 through one lightpipe (8channel 24/96 max total). Also incorporates Prism MR2024 compatible codec for hi-res recording on Tascam DA88 and DA38

Protools ® Compatible – Features 2 888-style connectors for direct connection to the computer card. Since Mytek 8X96 ADC and DAC come in two different physical boxes, second connector can be used to jump from ADC to DAC. Additionally there is an optional stereo piggyback DAC that can be mounted on this MDM to provide simple high quality 2 channel monitoring. Because of current status of the Protools system, the interface works up to 48kHz only.

Sonic Solutions HDSP/USP – Features 2 Sonic-style 68 pin connectors for direct connection to the computer card. Since Mytek 8X96 ADC and DAC come in two different physical boxes, second connector can be used to jump from ADC to DAC or another unit in the chain. The card features a switch to set the unit address. In the current firmware version (as of Dec 2000) 8X96 interfaces with Sonic, but the parameters have to be set physically with the unit's switches.

Sadie/Dual AES – Features 2 BOB style 25 pin connectors for direct connection to the computer card. Second connector can be used with second Sadie card for full 8 channel 24/96 system. One Sadie card provides interface for 4 channels of 24/96 or 8 channels of 24/48. The card is essentially a Dual-wire (S/MUX) AES interface on a 25 pin D connector and can be also used with other dual-wire AES equipment.

8X96 user can take full advantage of the choice of interfacing and use both ADC or DAC boxes to record 24/96 on standard 48k equipment and to decode, and copy data between any installed format. Detailed description of all cards can be found in the chapter "Digital Inputs, Outputs and Daughter cards"

If the converter is outfitted with an optional card, and a digital input is selected, a bit-by-bit digital format conversion can be performed. If the bit splitting and sample splitting function is off (see MRX, S/MUX), data bits are cloned and digital clocks are reclocked and cleaned up from jitter. The data path is 24 bit. No noise shaping can be performed on digital input data. If a bit splitting method is used , a bit-split signal arriving at the digital input can be decoded and output digitally at the proper wordlength (see the next chapter).

During normal operation, when the analog input (ADC) is selected as the signal source, the digital signal is present at both the AES outputs and the output of one the optional DIO card. If the A/D is outfitted with two such cards, the active one is selected using internal DIP Switch (see “Special DIP Switch Options”).

It is also possible to select either AES/EBU or any of the installed optional DIO (MDM card) as inputs. In this case the digital signal is looped through to the AES outputs as well as the outputs of the optional interface. Any interface specific flags except emphasis are dropped. Output sampling rate flags are set according to that of the sampling rate selector switch. There are only flags for 44.1 and 48k: 88.2 and 96k are not flagged as there is currently no AES/EBU standard for flagging high sample rates. The signal clock is regenerated and the internal clock PLL removes incoming clock jitter, so jitter is attenuated from the ADC digital output.
The AES/EBU interface will pass through (from its inputs to outputs) even 4 asynchronous signals. The format conversion, however, requires pairs to be synchronous as only the clock of pair #1/2 is used for format conversion. If less than 4 pairs are used, at least one pair should be feeding channels #1/2.
For example: if you need to record a stereo AES/EBU signal on your eight channel MDM machine on tracks #7/8 only, feed the signal to input #1/2 and the loop the AES output #1/2 to the AES input #7/8. Arm only tracks #7/8 and you are ready to go.

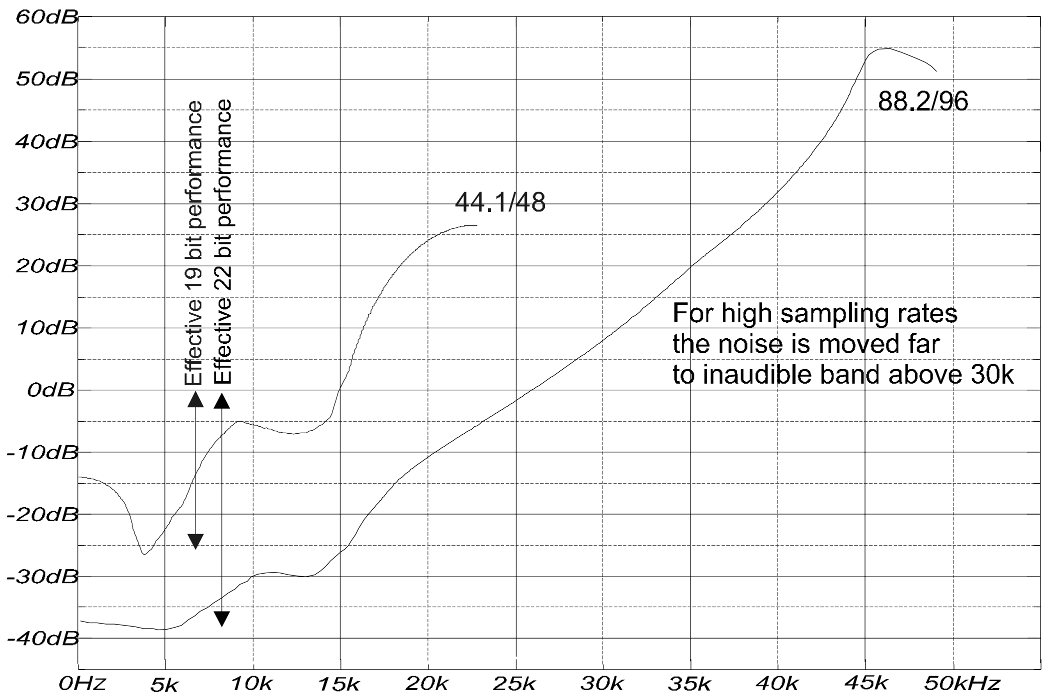
A stripped wordclock is present at the wordclock output. This signal is stripped off the clock of incoming signal. If AES/EBU input is selected, the wordclock is stripped from the pair #1/2 only. The chapter “Special DIP Switch Options” describes how to set wordclock output to be either 44/48 or 88/96, depending on application.

The 8X96 ADC does not do sample rate conversion.

8X96 ADC – Features

Mytek "SuperShaper-HR" ™ Noise Shaping & Dither

The 8X96 A/D converter employs SuperShaper-HR ™ - a sophisticated psychoacoustically optimized noise shaping and dithering algorithm. SuperShaper-HR ™ uses 9th order high resolution noise shaping filters with proprietary filter coefficients developed by Mytek Inc. The appropriate noiseshaping curve is programmed into noiseshaping filters for each selected sampling frequency. This feature allows for preserving 24 bit performance in 16 and 20 bit data output. The noiseshaping is automatically engaged whenever 16 or 20 bit wordlength is selected and is active only when analog input is selected. If digital input is selected the input data is cloned without any processing.



Noise shaper performance graphs at 16 bit output.

Note the increased benefit of noiseshaping at hi-speed where 22 bit performance can be achieved using 16 bit/96kHz storage.

When noise-shaping processing is on, the quantization noise is removed from the audio band, where the noise floor is kept as close as possible to the 24 bit level, and moved to the 20-22 kHz range (for FS=44 or 48k) or to 30-44kHz range (for high sampling rates). Noise shaping is especially beneficial at higher sampling rates. At 88.2 or 96kHz the noise is moved far beyond the audio band. As a result, the noiseshaping is much more effective. A noise shaped 20 bit, 96k signal sounds virtually as good as 24 bit, 96k. Thus using a 20 bit ADAT for 4 channel 20 bit, 96k recording (see S/MUX) can be very effective.

The noise-shaping filters are automatically engaged at appropriate levels when either 16 or 20 bit output wordlenght is selected. If 24 bit output is selected, the noise-shaping filter is off. There is an option to defeat the noise shaping and replace it with flat dither accessible via a DIP switch inside the unit. (see "Special DIP switch options")

More information about use of noise shaping in high resolution recording can be found in the following Audio Engineering Society Preprints (http://www.aes.org):

"Coding Methods for High Resolution Recording Systems"

Preprint Number: 4639 Convention: 103 1997-09
Author: J. Robert Stuart

This paper reviews the recording and reproduction chain from the viewpoints of digital audio engineering and psychoacoustics. It also attempts to define the audio requirements of a transparent digital audio channel. The theory and practice of selecting high sample rates such as 96kHz and word lengths of up to 24 bits are examined. The relative importance of sampling rate and word size at various points in the recording, mastering, transmission, and replay chain is discussed.

"Dynamic Range Enhancement Using Noise-Shaped Dither at 44.1, 48, and 96 kHz"

Preprint Number: 4236 Convention: 100 1996-05
Author: J. R. Stuart
Author: R. J. Wilson

This paper presents an overview of the work done on noise shaping, summarizing how noise shapers are designed, implemented and measured, and outlining the hearing model used to evaluate the various designs achieved.

"Dynamic Range Enhancement Using Noise-Shaped Dither Applied to Signals with and without Preemphasis"

Preprint Number: 3871 Convention: 96 1994-03
Author: Robert Stuart
Author: Rhonda J. Wilson

This paper presents new noise-shaped dithers designed to give increased subjective dynamic range when applied to signals with and without pre-emphasis.

"Optimal Noise Shaping and Dither of Digital Signals"

Preprint Number: 2822 Convention: 87 1989-10
Author: Michael Gerzon
Author: Peter G. Craven

This paper discusses optimizing noise shaping, with or without dither, using filtered error-feedback round a quantizer.

"Psychoacoustically Optimal Noise Shaping"

Preprint Number: 2965 Convention: 89 1990-09
AES Journal Vol:Issue: 40:7/8 Page: 611 Year: 1992
Author: Robert A. Wannamaker

This paper examines the design of psycho-acoustically optimal noise shaping filters for requantization in non-oversampling digital audio applications.

8X96 ADC – Features

Digital Inputs, Outputs and synchronization

Both ADC and DAC 8X96 Converters can perform digital format conversion between any installed format (MDM Daughter cards), including standard AES/EBU DIO. See “Digital Inputs, Outputs and Daughter cards)

8X96 ADC – Features

S/MUX ™ sample splitting and Prism MRX™ bit splitting

8X96 Converters bridge the advantages of 24 bit and 24/96 technology with the technical abilities of older digital equipment. S/MUX ™ and MRX ™ sample and bit splitting techniques allow for use of 16, 20 and 24 machines to record larger bit depth and higher sampling rates. These techniques are implemented on MDM daughter cards and are described in detail further in the chapter “Digital Inputs, Outputs and Daughter cards”.

8X96 ADC – Features

8X96 ADC - Special DIP Switch Options

Both ADC and DAC units are equipped with a dip switch mounted on the main motherboard next to BNC wordclock jack. This switch allows tailoring certain features of the A/D converters to the actual system setup.

Bold are factory default settings

Switch #	ON	OFF
1	**ADC dither for 16/20 out ON**	ADC dither for 16/20 out OFF
2	**Supershaper™ for 16/20 out ON**	Supershaper™ for 16/20 out OFF
3	NOT ASSIGNED	**NOT ASSIGNED**
4	**WCK I/O =44/48 for 88/96 data**	WCK I/O =88/96 for 88/96 data
5	**TDIF IN is 24bit**	truncate bits 17-24 on TDIF in
6	**Cloning Disabled**	Cloning MDM Connectors
7	AES is Dual Wire for 88/96	**AES is Hi-Speed for 88/96**
8	NOT ASSIGNED	**NOT ASSIGNED**

Description:

ADC dither for 16/20 out- when 16 or 20 bit conversion is selected appropriate dither is applied

Supershaper™ for 16/20 out – when 16 or 20 bit conversion is selected the SuperShaper noise shaping is engaged. For best transparency we recommend using it in conjunction with dither. With this , the 9th order noiseshaper is very similar to Po-wr ™ noiseshaping algorithm #3.

WCK I/O =44/48 for 88/96 data– This setting facilitates synchronizing the converter working in 88/96kHz mode with external equipment working in 44/48k mode. For example Tascam machines require 44/48k wordclock , when they record 88/96k in bit/sample split mode. This setting affects both wck in and out.

Truncate bits 17-24 on TDIF in (default OFF, default pass 24 bit) – This setting truncates lowest 8 bit on TDIF input, allowing only for 16 bit input resolution. We have implemented this setting after discovering that Tascam DA88 machine outputs “rubbish” on bits 17-24.

Cloning MDM Connectors (default disabled) – This setting is active in S/MUX or MRX mode and with ADAT and TDIF cards only. The cloning function is primarily intended for output to a backup machine and for easy A/B’ing of two sources.

Since the cards feature 2 sets of connectors called “machine A” and “machine B” and the ADC is 8x24/96, while connectors are 44/48k, there are several modes the unit can operate with.

Cloning disabled, (88/96k, MRX OFF):

All channels are sample-split (S/MUX) . Channels 1-4 are output/input in S/MUX pairs on connector A and channels 5-8 are on connector B.
Machine A/B select switch is not operational.

Cloning disabled, (88/96k, MRX MODE1–96 =4x24@88/96):

Channels 1-2 and 5-6 are bit-split (MRX) . Channels 1-2 are output/input in MRX pairs on connector A and channels 5-6 are on connector B.
Channels 3,4,7,8 are not used.
Machine A/B select switch is not operational.

Cloning disabled, (88/96k, MRX MODE2–96 =6x20@88/96):

Channels 1-3 and 5-7 are bit-split (MRX) . Channels 1-3 are output/input in MRX pairs on connector A and channels 5-7 are on connector B.
Channels 4,8 are not used.
Machine A/B select switch is not operational.

Cloning disabled, (44/48k, MRX MODE1 =8x24@44/48):

Channels 1-8 are bit-split (MRX) . Channels 1-4 are output/input in MRX pairs on connector A and channels 5-8 are on connector B.
Machine A/B select switch is not operational.

Cloning enabled, (MRX OFF):

Channels 1-4 are sample-split (S/MUX) . Channels 1-4 are output/input in S/MUX pairs on connector A and B simultaneously. Machine A/B select switch can select the source to be either connector A or B.

Cloning enabled, (88/96k, MRX MODE1–96 =2x24@88/96):

Channels 1-2 are bit-split (MRX) . Channels 1-2 are output/input in MRX pairs on connector A and B simultaneously. Channels 3-8 are not used.

Machine A/B select switch can select the source to be either connector A or B.

Cloning enabled, (88/96k, MRX MODE2–96 =3x20@88/96):

Channels 1-3 are bit-split (MRX) . Channels 1-3 are output/input in MRX pairs on connector A and B simultaneously. Channels 4-8 are not used. Machine A/B select switch can select the source to be either connector A or B.

Cloning enabled, (44/48k, MRX MODE1 =4x24@44/48):

Channels 1-4 are bit-split (MRX) . Channels 1-4 are output/input in MRX pairs on connector A and B simultaneously. Channels 5-8 are not used. Machine A/B select switch can select the source to be either connector A or B.

Cloning enabled, (44/48k, MRX MODE2 =6x20@44/48):

Channels 1-6 are bit-split (MRX) . Channels 1-6 are output/input in MRX pairs on connector A and B simultaneously. Channels 7-8 are not used. Machine A/B select switch can select the source to be either connector A or B.

AES is Dual Wire for 88/96 – In on position AES I/O operates in dual wire mode ie hi-speed 96(88)k signal is split into two 48(44)k streams (similar to S/MUX- see further description about S/MUX). In this mode each AESI/O carries single channel of audio (as opposed to usual stereo). Only 4 channels (1-4) are available through AES I/O in this mode.

8X96 ADC – Specifications

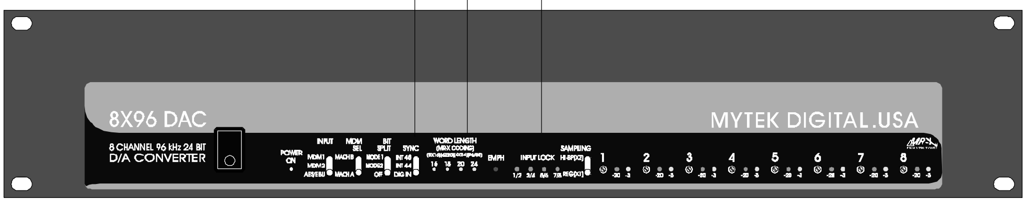
Conversion:	Linear, 128x oversampling at 44.1/48kHz 64x oversampling at 88.2/96kHz
Resolution:	24 bit
Sample rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz or wordclock 25-100kHz
Dynamic Range:	120dB A-weighted, 117dB Total
THD+Noise:	-105dB (<0.0005%)
Internal clock jitter:	<10picoseconds
Analog Inputs:	+4dBm balanced or unbalanced, 10kOhm Adjustable +6dB/-14dB
Digital inputs/ outputs:	Hi-speed (25-100kHz) AES/EBU two MDM Daughter Cards DIO optional: (choose ADAT, TDIF, Protools Compatible, HDSP/USP, Sadie)
Bitsplitting/ Samplesplitting:	Prism MRX ™ and S/MUX ™ w/ TDIF card S/MUX ™ w/ ADAT card Dual wire AES/EBU w/ Sadie card
External Sync.:	Wordclock in and out. 15 LS TTL loads max. on output
Mains:	100/115V-220/240V switchable
Dimensions:	1 rack space wide x 2U high x 5.5” deep (19”x 3.50”x 5.5”)
Weight:	8 pounds

8X96 DAC – Quick Start

Dual function leds. Normally leds display the wordlenght of the signal. If MRX is selected led display detected mode of MRX encoding and its compatibility with MRX switch setting. (see “Bitsplitting”)

DAC sync source. Normally “DIG IN” . See “DAC - internal sync” if you need a completely jitter immune D/A conversion

Lock leds indicate lock of pairs of digital inputs. Note that for 8 channel DIO interfaces there is one common clock, while AES/EBU pairs can be converted to analog independently (asynchronously). However, AES pairs have to be synchronous to be properly converted to another digital format.



Input selector switch

Some DIO interface cards such as ADAT and TDIF have 2 sets of 8 chan DIO - A and B. In regular speed mode, this switch selects either port A or B. In high speed mode and w/MRX and S/MUX modes (see “dip switch”)

Mode of MRX codec - turn it off if you are not using MRX. Relevant to TDIF Only.

Sampling frequency range:
Reg= 25kHz - 50kHz
Hi= 50kHz - 100kHz
Switch automatically engages S/MUX on ADAT and TDIF

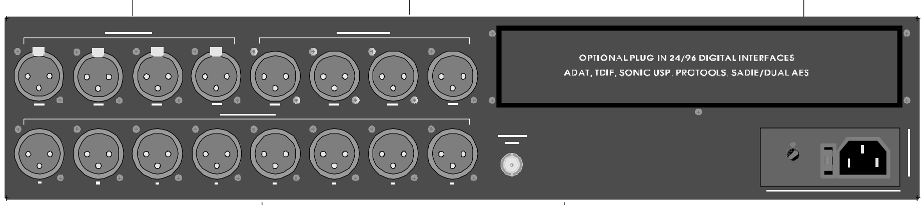
Output level adjustment. Nominal 0VU=+4dB=1.225VRMS=-18dBFS

-20dB and -3dB peak leds display the peak signal level relative to 0 dBFS.

Hi-speed AES/EBU inputs lock to signals of 25-100kHz FS. Standard 110 Ohm, balanced, transformer isolated

Hi-speed AES/EBU outputs for signals of 25-100kHz FS. Standard 110 Ohm, balanced, transformer isolated implementation. As all outputs they repeat selected input. If “AES IN” is selected, the signal is a reclocked (jitter reduced) clone i.e. all bits and flags are exactly repeated. If other DIO is selected, certain flags may not be repeated.
(See “Digital format conversion”)

Up to **two** Digital Input-Output Interfaces can be installed (apart of standard AES/EBU). These interfaces are daughter cards manufactured by Mytek and can be:
-ADAT with S/MUX 96/24 recording
-TDIF with S/MUX 24/96 and Prism MRX Bitsplit
-Sonic HDSP/USP
-Protools ® compatible
-Sadie / dual AES



Analog outputs. Nominal balanced +4dB level. Consumer -10dBV level can be achived by turning the level trimpots down and, connecting sleeve to pin 1 and ring to pin 2. Leave pin 3 unconnected. Do not ever short any pins to ground.

This BNC jack can output AES11 Sync signal or Wordclock. Switchable inside via jumpers. Wordclock is default. Wordclock is a TTL level signal = FS and phase locked to incoming sample clock. If AES/EBU input is selected the clock is stripped from pair #1/2. A 75 Ohm terminating resistor can be switched in via jumper inside.

Setting up the 8X96 DAC is fairly straightforward- install the unit in the rack, and make all necessary connections. If you rack frame is grounded, you may consider evaluating inter-equipment ground connections and power cord grounding. If unbalanced audio is involved, connect sleeve to pin 1 and ring to pin 2. **Leave pin 3 floating**. If an external

house clock is used, make sure that the clocks are properly fed and terminated (75 Ohm) if necessary, especially for long cable runs.
8X96 Converters are microprocessor controlled and as such are subject to static interference. If unusual circumstances such as lightning storm cause a malfunction, try rebooting the unit. For further questions regarding setup and features contact your dealer or Mytek technical support.

8X96 ADC – Analog Input Alingment

Before using the 8X96ADC you need to align its analog inputs. The alignment generally does not affect audio quality, only input sensitivity. The alignment is performed using 10 turn trimpots on the front panel. A small tweaker or screwdriver is necessary.

You have to arbitrarily decide what will be your studio "0 VU" analog/digital reference level. It is usually between -20 and -14dB. It defines how much headroom you have left over the normal operating "0 VU" level. You may set it at the same level as other piece of equipment in your studio. For example Panasonic 3700 is fixed at -18dB (which is low for most rock and roll recordings). ADAT is fixed at -15dB which is an optimal level.

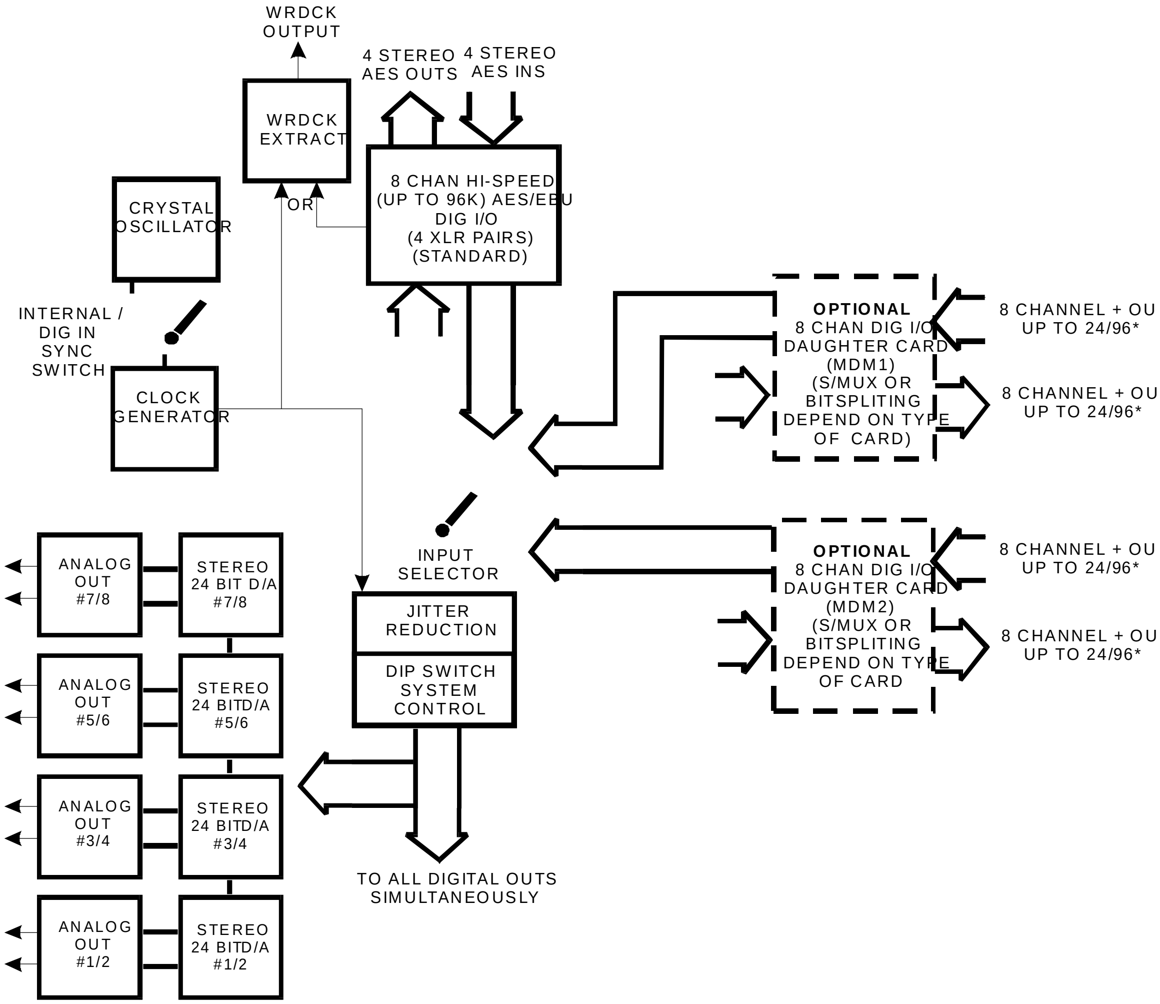
1. Play a digital audio source (DAW generator, test CD etc.) set to approx. 1kHz at the operating level you have chosen (let's say you have chosen -15dB).

2. Send the signal from the analog output to the console VU meter. If you don't have a VU meter you can use an AC voltmeter set to measure AC RMS values. "0 VU"at + 4dB corresponds to 1.225 Volts RMS measured between pin 2 and 3 of the output XLRs.

3.Using a tweaker or a small screwdriver adjust the analog outputs until VU meter reads "0 VU".

4. Repeat the same for all outputs.

8X96 DAC – Signal Flow



8X96 ADC – Features

Digital Inputs, Outputs and synchronization

8X96DAC comes equipped with **hi-speed AES/EBU** digital input and output standard. Additionally up to two daughter interface (MDM) cards can be installed. There are several different MDM cards available for Mytek 8X96. They are the same for both ADC and DAC:

ADAT – Features 2 ADAT lightpipe inputs (2x 8chan@48k) and 2 ADAT lightpipe outputs. Incorporates S/MUX sample splitting for 4 channel of 24/96 through one lightpipe (8channel 24/96 max total)

TDIF – Features 2 TDIF inputs (2x 8chan@48k) and 2 TDIF outputs (2 TDIF connectors). Incorporates S/MUX sample splitting for 4 channel of 24/96 through one lightpipe (8channel 24/96 max total). Also incorporates Prism MR2024 compatible codec for hi-res recording on Tascam DA88 and DA38

Protools ® Compatible – Features 2 888-style connectors for direct connection to the computer card. Since Mytek 8X96 ADC and DAC come in two different physical boxes, second connector can be used to jump from ADC to DAC. Additionally there is an optional stereo piggyback DAC that can be mounted on this MDM to provide simple high quality 2 channel monitoring. Because of current status of the Protools system, the interface works up to 48kHz only.

Sonic Solutions HDSP/USP – Features 2 Sonic-style 68 pin connectors for direct connection to the computer card. Since Mytek 8X96 ADC and DAC come in two different physical boxes, second connector can be used to jump from ADC to DAC or another unit in the chain. The card features a switch to set the unit address. In the current firmware version (as of Dec 2000) 8X96 interfaces with Sonic, but the parameters have to be set physically with the unit's switches.

Sadie/Dual AES – Features 2 BOB style 25 pin connectors for direct connection to the computer card. Second connector can be used with second Sadie card for full 8 channel 24/96 system. One Sadie card provides interface for 4 channels of 24/96 or 8 channels of 24/48. The card is essentially a Dual-wire (S/MUX) AES interface on a 25 pin D connector and can be also used with other dual-wire AES equipment.

8X96 user can take full advantage of the choice of interfacing and use both ADC or DAC boxes to record 24/96 on standard 48k equipment and to decode, and copy data between any installed format. Detailed description of all cards can be found in the chapter "Digital Inputs, Outputs and Daughter cards"

If the converter is outfitted with an optional card, and a digital input is selected, a bit-by-bit digital format conversion can be performed. If the bit splitting and sample splitting function is off (see MRX, S/MUX), data bits are cloned and digital clocks are reclocked and cleaned up from jitter. The data path is 24 bit. If a bit splitting method is used , a bit-split signal arriving at the digital input can be decoded and output digitally at the proper wordlength (see the next chapter).

During normal operation, the digital signal is present at both the AES outputs and the output of the optional DIO cards.

The AES/EBU interface will pass through (from its inputs to outputs) even 4 asynchronous signals. The format conversion, however, requires pairs to be synchronous as only the clock of pair #1/2 is used for format conversion. If less than 4 pairs are used, at least one pair should be feeding channels #1/2.
For example: if you need to record a stereo AES/EBU signal on your eight channel MDM machine on tracks #7/8 only, feed the signal to input #1/2 and the loop the AES output #1/2 to the AES input #7/8. Arm only tracks #7/8 and you are ready to go.

The converters need a sampling clock of any frequency between 25-100 kHz. The circuitry inside is divided into two ranges: 25-50kHz and 50-100kHz. The “Sampling” switch has to be always set to x1 (Regular) for the lower range and x2 (Hi-speed) for the higher range.

The conversion is 24 bit. 16, 18 and 20 bit signals are automatically converted with the full wordlength, resulting in performance superior to 16, 18 or 20 bit DAC’s.

No sample rate conversion or noise shaping is implemented in the DAC.

8X96 DAC – Features

Jitter Immune Operation, Internal Sync, Wordclock Out

In the normal mode DACs are always slaved to incoming digital input. Incoming clock is run through two PLL circuits for jitter reduction. The incoming data is present on all outputs with certain limitations regarding the AES/EBU asynchronous mode. (See "Asynchronous operations as 4 independent stereo DACs" below).

In normal mode wordclock stripped off the clock of incoming signal is present at the wordclock output.. If AES/EBU input is selected, the wordclock is stripped from the pair #1/2 only. The chapter "Special DIP Switch Options" describes how to set wordclock output to be either 44/48 or 88/96, depending on application.. The wordclock output is intended for special use to be determined by the user. The wordclock output is capable of driving 75 Ohm terminated destination . This is recommended whenever long (over 10 ft.) cable is used.

There is a special "internal sync mode" which eliminates any relation of sound quality to incoming jitter. In this mode the converters are clocked from internal crystal oscillator with accuracy of 10ps of jitter. Wordclock generated by DAC is then driving the signal source, rather than normal other way around. No matter how jittery the source is the converters will always sound the same. This mode is recommended for experienced users and for fixed installations as it requires special attention to overall digital system clocking. Currently, this feature is implemented for signals originating from AES/EBU inputs only.

8X96 DAC – Features

Asynchronous operations as 4 independent stereo DACs

Digital to analog converters are implemented as 4 stereo pairs. They can operate synchronously or asynchronously. During synchronous operation all 4 converters always receive the same clock, meaning that all four data streams are in phase. This is always the case when the Daughter Card input is selected. **When the AES/EBU input is selected, the 4 streams can be either synchronous or asynchronous**, meaning that up four independent AES/EBU sources can be monitored as if 4 separate stereo DACs were used. Format conversion (see further) can only be performed if AES/EBU sources are synchronous.

8X96 DAC – Features

Digital Format Conversion

Both ADC and DAC 8X96 Converters can perform digital format conversion between any installed format (MDM Daughter cards), including standard AES/EBU DIO. See “Digital Inputs, Outputs and Daughter cards)

8X96 DAC – Features

Bit Splitting (Prism MRX™) and Sample Splitting (S/MUX™)

The 8X96DAC bit splitting and sample splitting operates exactly as in the 8X96ADC (See: "Digital Inputs, Outputs and Daughter Cards- S/MUX, MRX").

8X96 DAC - Special DIP Switch Options

Both ADC and DAC units are equipped with a dip switch mounted on the main motherboard next to BNC wordclock jack. This switch allows tailoring certain features of the D/A converters to the actual system setup.

Bold are factory default settings

Switch #	ON	OFF
1	NOT ASSIGNED	**NOT ASSIGNED**
2	SYNCHRO AES INS	**AES INS INDEPENDENT**
3	NOT ASSIGNED	**NOT ASSIGNED**
4	**WCK OUT =44/48 for 88/96 data**	WCK OUT =88/96 for 88/96 data
5	**TDIF IN is 24bit**	truncate bits 17-24 on TDIF in
6	**Cloning Disabled**	Cloning MDM Connectors
7	AES is Dual Wire for 88/96	**AES is Hi Speed for 88/96**
8	NOT ASSIGNED	**NOT ASSIGNED**

Description:

Synchro AES Ins– In “on” position AES receivers for pairs 3-4, 5-6, 7-8 are slaved to AES IN 1-2. This setting must be used when simultaneous format conversion of 4 synchronous AES pairs is required. It also must be used when DAC is resolved to its internal clock and an external AES source is slaved to DAC clock.

WCK I/O =44/48 for 88/96 data– This setting facilitates synchronizing the converter working in 88/96kHz mode with external equipment working in 44/48k mode. For example Tascam machines require 44/48k wordclock , when they record 88/96k in bit/sample split mode. This setting affects both wck in and out.

Truncate bits 17-24 on TDIF in (default OFF, default pass 24 bit) – This setting truncates lowest 8 bit on TDIF input, allowing only for 16 bit input resolution. We have implemented this setting after discovering that Tascam DA88 machine outputs "rubbish" on bits 17-24.

Cloning MDM Connectors (default disabled) – This setting is active in S/MUX or MRX mode and with ADAT and TDIF cards only. The cloning function is primarily intended for output to a backup machine and for easy A/B'ing of two sources.

Since the cards feature 2 sets of connectors called "machine A" and "machine B" and the ADC is 8x24/96, while connectors are 44/48k, there are several modes the unit can operate with.

Cloning disabled, (88/96k, MRX OFF):

All channels are sample-split (S/MUX) . Channels 1-4 are output/input in S/MUX pairs on connector A and channels 5-8 are on connector B. Machine A/B select switch is not operational.

Cloning disabled, (88/96k, MRX MODE1–96 =4x24@88/96):

Channels 1-2 and 5-6 are bit-split (MRX) . Channels 1-2 are output/input in MRX pairs on connector A and channels 5-6 are on connector B. Channels 3,4,7,8 are not used.
Machine A/B select switch is not operational.

Cloning disabled, (88/96k, MRX MODE2–96 =6x20@88/96):

Channels 1-3 and 5-7 are bit-split (MRX) . Channels 1-3 are output/input in MRX pairs on connector A and channels 5-7 are on connector B. Channels 4,8 are not used.
Machine A/B select switch is not operational.

Cloning disabled, (44/48k, MRX MODE1 =8x24@44/48):

Channels 1-8 are bit-split (MRX) . Channels 1-4 are output/input in MRX pairs on connector A and channels 5-8 are on connector B.
Machine A/B select switch is not operational.

Cloning enabled, (MRX OFF):

Channels 1-4 are sample-split (S/MUX) . Channels 1-4 are output/input in S/MUX pairs on connector A and B simultaneously. Machine A/B select switch can select the source to be either connector A or B.

Cloning enabled, (88/96k, MRX MODE1–96 =2x24@88/96):

Channels 1-2 are bit-split (MRX) . Channels 1-2 are output/input in MRX pairs on connector A and B simultaneously. Channels 3-8 are not used. Machine A/B select switch can select the source to be either connector A or B.

Cloning enabled, (88/96k, MRX MODE2–96 =3x20@88/96):

Channels 1-3 are bit-split (MRX) . Channels 1-3 are output/input in MRX pairs on connector A and B simultaneously. Channels 4-8 are not used. Machine A/B select switch can select the source to be either connector A or B.

Cloning enabled, (44/48k, MRX MODE1 =4x24@44/48):

Channels 1-4 are bit-split (MRX) . Channels 1-4 are output/input in MRX pairs on connector A and B simultaneously. Channels 5-8 are not used. Machine A/B select switch can select the source to be either connector A or B.

Cloning enabled, (44/48k, MRX MODE2 =6x20@44/48):

Channels 1-6 are bit-split (MRX) . Channels 1-6 are output/input in MRX pairs on connector A and B simultaneously. Channels 7-8 are not used. Machine A/B select switch can select the source to be either connector A or B.

AES is Dual Wire for 88/96 – In on position AES I/O operates in dual wire mode ie hi-speed 96(88)k signal is split into two 48(44)k streams (similar to S/MUX- see further description about S/MUX). In this mode each AESI/O carries single channel of audio (as opposed to usual stereo). Only 4 channels (1-4) are available through AES I/O in this mode.

8X96 DAC – Specifications

Conversion:	Linear, 128x oversampling at 44.1/48kHz 64x oversampling at 88.2/96kHz
Resolution:	24 bit
Sample rates:	25-100kHz (dig. in sync), 44.1k, 48k, 88.2k, 96k (int. sync)
Internal Clock Jitter:	10 ps
Dynamic Range:	120dB A-weighted, 117dB Total
THD+Noise:	-100dB (<0.001%)
Analog Outputs:	+4dBm balanced or unbalanced, 10kOhm Adjustable +6dB/-14dB
Digital inputs/ outputs:	Hi-speed (25-100kHz) AES/EBU two MDM Daughter Cards DIO optional: (choose ADAT, TDIF, Protools Compatible, HDSP/USP, Sadie)
Bitsplitting/ Samplesplitting:	Prism MRX ™ and S/MUX ™ w/ TDIF card S/MUX ™ w/ ADAT card Dual wire AES/EBU w/ Sadie card
External Sync.:	Wordclock output 15 LS TTL loads max. on output
Mains:	100/115V-220/240V switchable
Dimensions:	1 rack space wide x 2U high x 5.5” deep (19”x 3.50”x 5.5”)
Weight:	8 pounds

Digital Inputs, Outputs and Daughter cards

8X96DAC comes equipped with **hi-speed AES/EBU** digital input and output standard. Additionally up to two daughter interface (MDM) cards can be installed. There are several different MDM cards available for Mytek 8X96. They are the same for both ADC and DAC:

ADAT – Features 2 ADAT lightpipe inputs (2x 8chan@48k) and 2 ADAT lightpipe outputs. Incorporates S/MUX sample splitting for 4 channel of 24/96 through one lightpipe (8channel 24/96 max total)

TDIF – Features 2 TDIF inputs (2x 8chan@48k) and 2 TDIF outputs (2 TDIF connectors). Incorporates S/MUX sample splitting for 4 channel of 24/96 through one lightpipe (8channel 24/96 max total). Also incorporates Prism MR2024 compatible codec for hi-res recording on Tascam DA88 and DA38

Protools ® Compatible – Features 2 888-style connectors for direct connection to the computer card. Since Mytek 8X96 ADC and DAC come in two different physical boxes, second connector can be used to jump from ADC to DAC. Additionally there is an optional stereo piggyback DAC that can be mounted on this MDM to provide simple high quality 2 channel monitoring. Because of current status of the Protools system, the interface works up to 48kHz only.

Sonic Solutions HDSP/USP – Features 2 Sonic-style 68 pin connectors for direct connection to the computer card. Since Mytek 8X96 ADC and DAC come in two different physical boxes, second connector can be used to jump from ADC to DAC or another unit in the chain. The card features a switch to set the unit address. In the current firmware version (as of Dec 2000) 8X96 interfaces with Sonic, but the parameters have to be set physically with the unit's switches.

Sadie/Dual AES – Features 2 BOB style 25 pin connectors for direct connection to the computer card. Second connector can be used with second Sadie card for full 8 channel 24/96 system. One Sadie card provides interface for 4 channels of 24/96 or 8 channels of 24/48. The card is essentially a Dual-wire (S/MUX) AES interface on a 25 pin D connector and can be also used with other dual-wire AES equipment.

8X96 user can take full advantage of the choice of interfacing and use both ADC or DAC boxes to record 24/96 on standard 48k equipment and to decode, and copy data between any installed format.

If the converter is outfitted with an optional card, and a digital input is selected, a bit-by-bit digital format conversion can be performed. If the bit splitting and sample splitting function is off (see MRX, S/MUX), data bits are cloned and digital clocks are reclocked and cleaned up from jitter. The data path is 24 bit. If a bit splitting method is used , a bit-split signal arriving at the digital input can be decoded and output digitally at the proper wordlength (see the next chapter).

During normal operation, the digital signal is present at both the AES outputs and the output of the optional DIO cards.

The AES/EBU interface will pass through (from its inputs to outputs) even 4 asynchronous signals. The format conversion, however, requires pairs to be synchronous as only the clock of pair #1/2 is used for format conversion. If less than 4 pairs are used, at least one pair should be feeding channels #1/2.
For example: if you need to record a stereo AES/EBU signal on your eight channel MDM machine on tracks #7/8 only, feed the signal to input #1/2 and the loop the AES output #1/2 to the AES input #7/8. Arm only tracks #7/8 and you are ready to go.

The converters need a sampling clock of any frequency between 25-100 kHz. The circuitry inside is divided into two ranges: 25-50kHz and 50-100kHz. The “Sampling” switch has to be always set to x1 (Regular) for the lower range and x2 (Hi-speed) for the higher range.

No sample rate conversion or noise shaping is performed during D-D format conversion.

Hi-Speed AES/EBU interface (Standard)

Hi-speed AES/EBU interface is standard for both ADC and DAC converters. The interface is capable of transferring of signals up to 24 bit and sampling rates of 25-100kHz. It is also possible to select either AES/EBU or optional MDM DIO as inputs. The digital signal is looped through to the AES outputs as well as the outputs of the optional interface. Any interface specific flags except emphasis are dropped. In ADC output sampling rate flags are set according to that of the sampling rate selector switch. There are only flags for 44.1 and 48k: 88.2 and 96k are not flagged as there is currently no AES/EBU standard for flagging high sample rates. The signal clock is regenerated and the internal clock PLL removes incoming clock jitter, in other words jitter is attenuated.
The AES/EBU interface will pass through (from its inputs to outputs) even 4 asynchronous signals. The format conversion, however, requires pairs to be synchronous as only the clock of pair #1/2 is used for format conversion. If less than 4 pairs are used, at least one pair should be feeding channels #1/2.

When digital input is selected, stripped wordclock is present at the wordclock output. This signal is stripped off the clock of incoming signal. If AES/EBU input is selected, the wordclock is stripped from the pair #1/2 only.

No sample rate conversion or noiseshaping is or can be performed during D-D transfers.

An upgraded version of AES/EBU interface is planned. It will feature following additional functions which are not available now:

- “dual wire” , 4 channel interface option for 96kHz operation
- automatic sample rate flag detection
- ability to use internal clock generator in DAC (now available in MDM interfaces only)

Inquire with Mytek about schedule and upgrades.

TDIF Interface (with MRX ™ and S/MUX ™)

TDIF MDM interface is equipped with two TDIF ports, A an B which allow to operate with either one or two Tascam MDM machines or interface to other Tascam equipment for up to 8 channels of I/O and signals up to 24/96. When Tascam MDM machines are used with the 8X96ADC, the ADC wordclock output has to be fed to the machine wordclock input and the machine has to be set to accept external wordclock and TDIF digital input. In case of 24 bit machines (DA78HR and DA98HR) please make sure the input wordlenght is set to 24 bit and dither is off.

No wordclock is required to interface the 8X96 DAC.

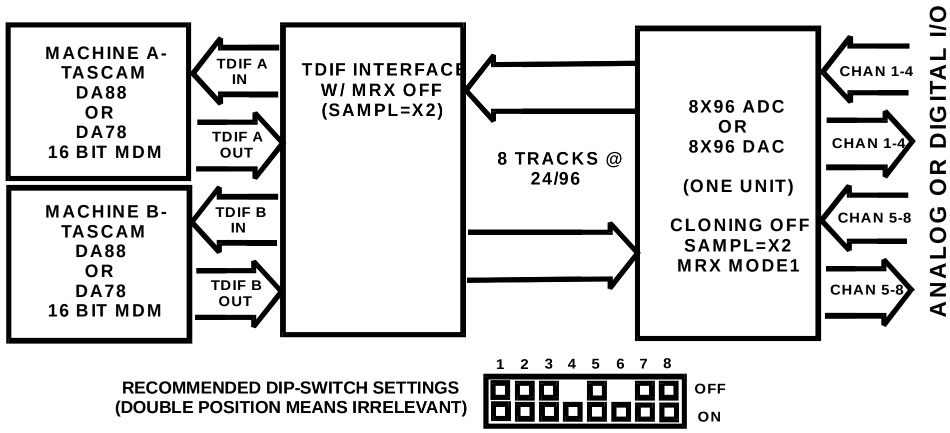
The diagrams below illustrate most popular system applications. We recommend one tailored to your equipment and needs. Because of sample accurate operation of Tascam MDMs several 8X96 units can be stacked to increase number of channels. In this case the first ADC is usually the clock master while the rest are clock slaves.
Detailed description of dip-switch functions can be found in chapter “dip-switch settings”). Not all possible setup configurations are mentioned (such as e.g. 16 bit/96 k)

48 or 96 numbers are used as an abbreviation. They mean 44.1 kHz and 48 kHz OR 88.2 kHz and 96 kHz sampling frequency.
MRX™ encoding used in some the examples is licensed from Prism and is fully compatible, i.e. can be decoded by Prism MR 2024T bit splitting interface.

Users of newer 24 bit Tascam MDMs, Tascam MX2424 harddisk recorder and certain TDIF equipped cards (SEKD) can take advantage of 24/96 capability of Mytek TDIF interface which is achieved by employing sample splitting technique known as Sonorus S/MUX (tm), Prism Sound Split96 (tm) where one high speed 96k channel is split between 2 48k channels (tracks). This technique is compatible with dual wire AES and recording technique used in Tascam DA98HR. I other words 96k tapes recorded on DA78HR using Mytek 8X96 can be played back on DA98HR without the need for any decoder.

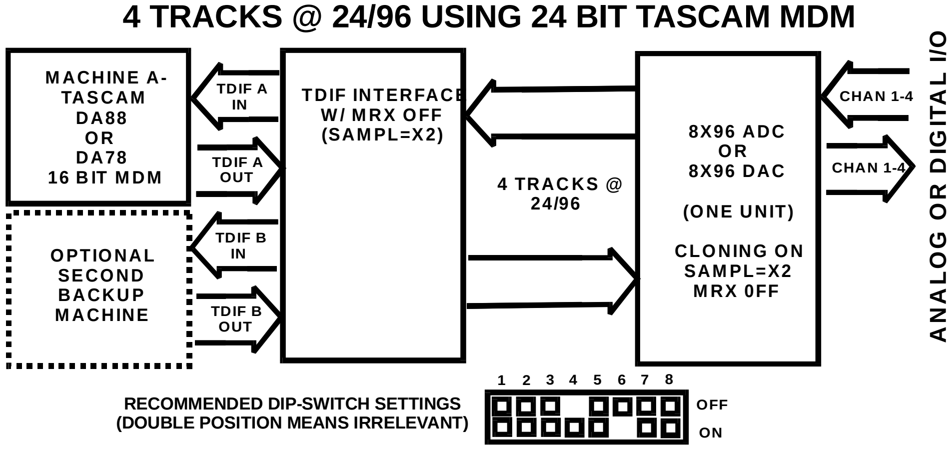
8 TRACKS @ 24/96 USING 24 BIT TASCAM MDM

Our favorite, most elegant use of Tascam 24 bit MDM and 8X96



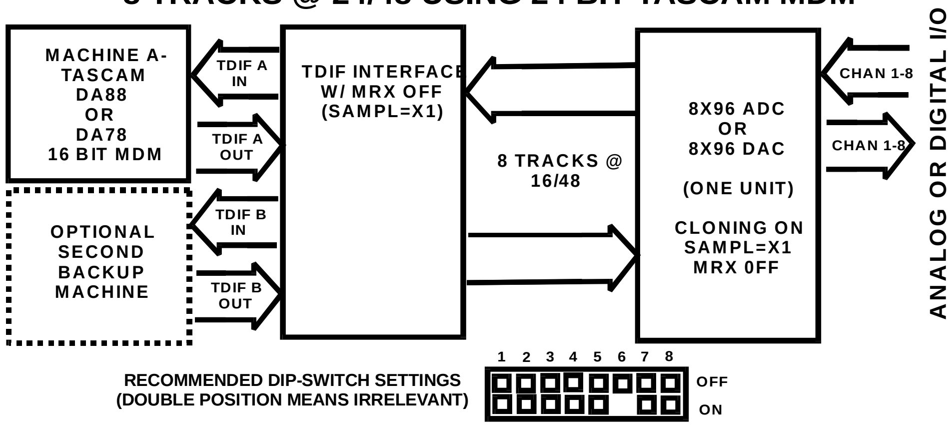
WARNING- FOR DATA INTEGRITY USE ONLY FUJI TAPE W/ TASCAM 24 BIT MDMs

4 TRACKS @ 24/96 USING 24 BIT TASCAM MDM



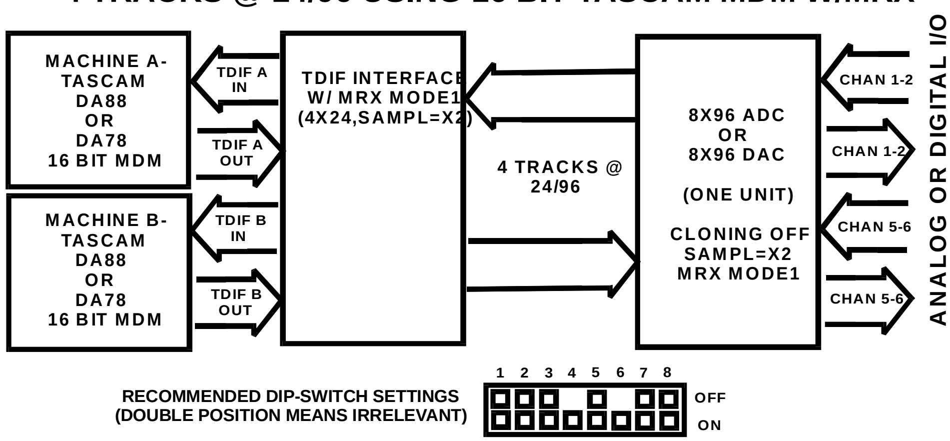
WARNING- FOR DATA INTEGRITY USE ONLY FUJI TAPE W/ TASCAM 24 BIT MDMs

8 TRACKS @ 24/48 USING 24 BIT TASCAM MDM

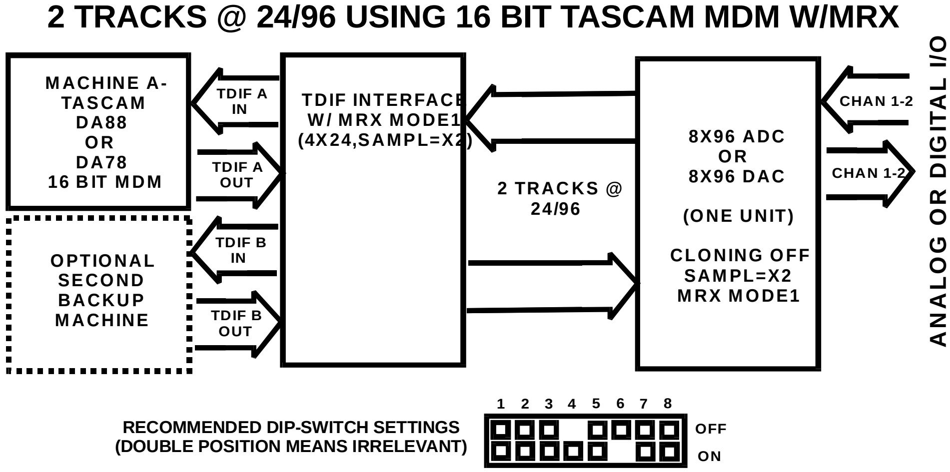


WARNING- FOR DATA INTEGRITY USE ONLY FUJI TAPE W/ TASCAM 24 BIT MDMs

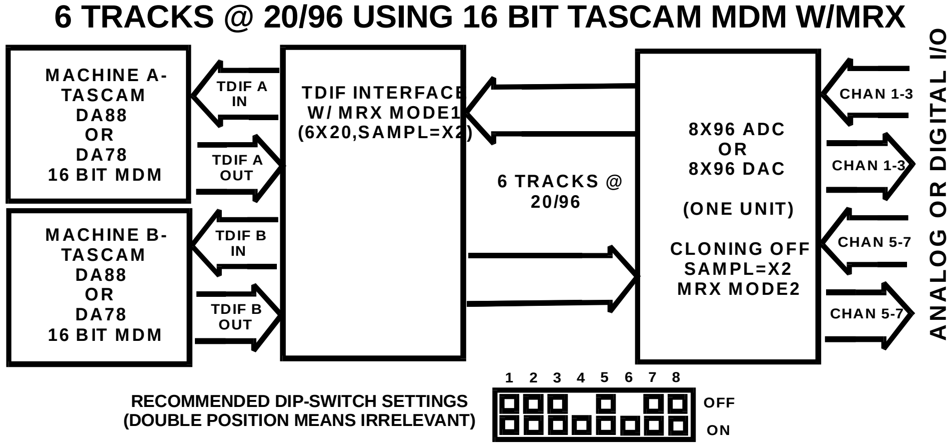
4 TRACKS @ 24/96 USING 16 BIT TASCAM MDM W/MRX



2 TRACKS @ 24/96 USING 16 BIT TASCAM MDM W/MRX

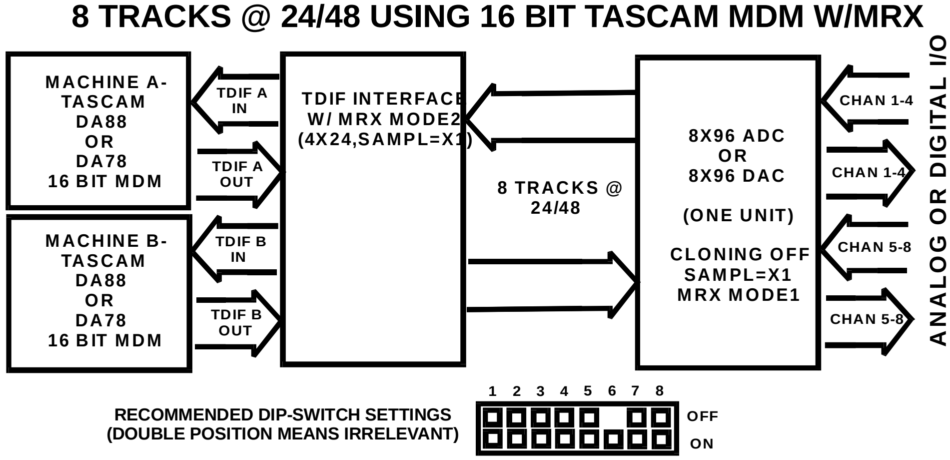


6 TRACKS @ 20/96 USING 16 BIT TASCAM MDM W/MRX



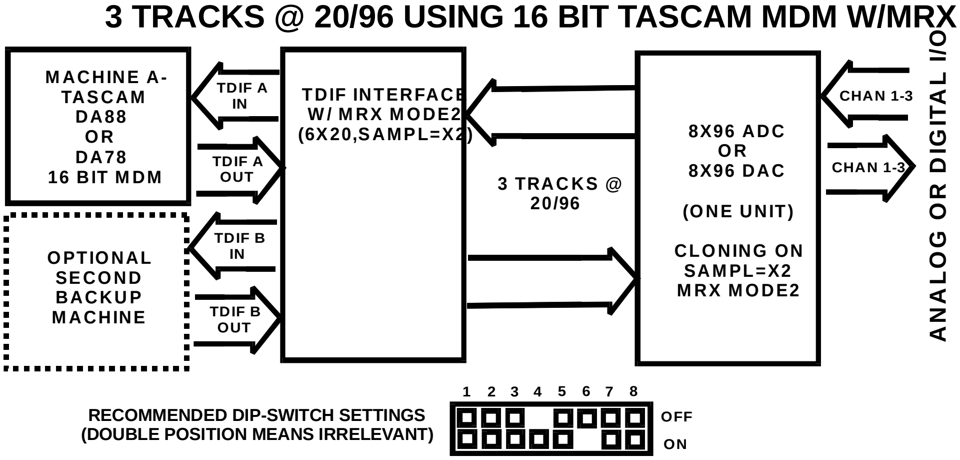
RECOMMENDED DIP-SWITCH SETTINGS
(DOUBLE POSITION MEANS IRRELEVANT)

8 TRACKS @ 24/48 USING 16 BIT TASCAM MDM W/MRX



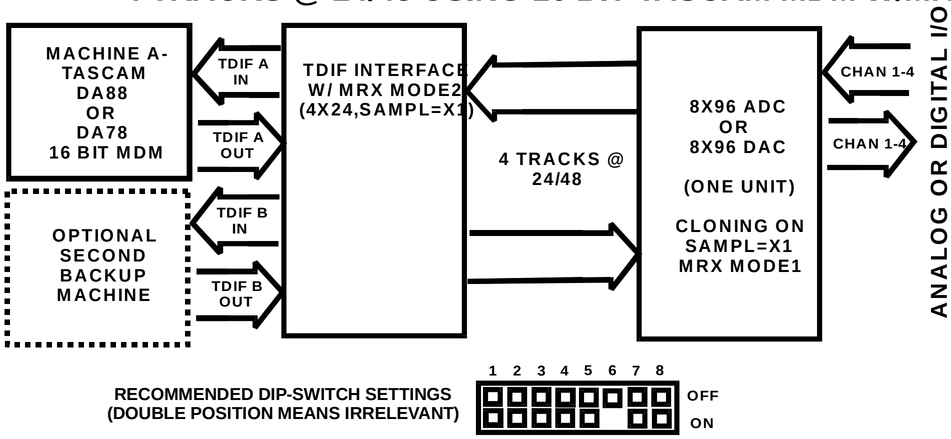
RECOMMENDED DIP-SWITCH SETTINGS
(DOUBLE POSITION MEANS IRRELEVANT)

3 TRACKS @ 20/96 USING 16 BIT TASCAM MDM W/MRX

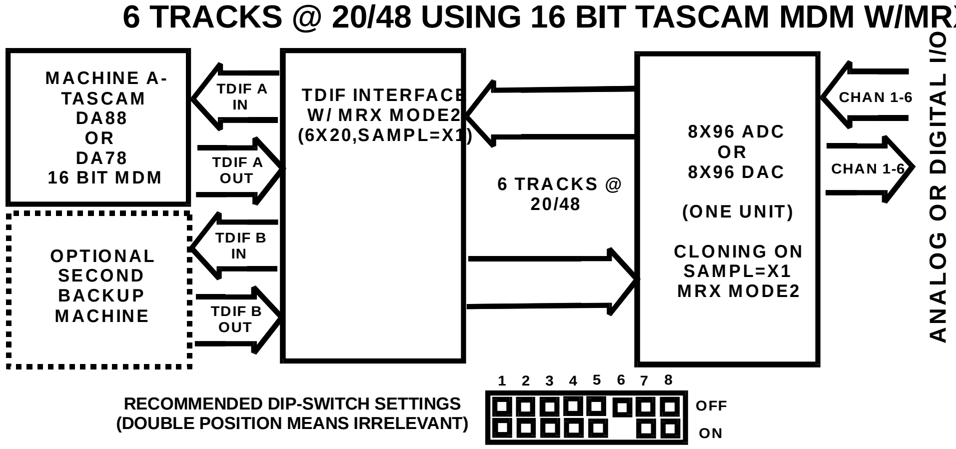


RECOMMENDED DIP-SWITCH SETTINGS
(DOUBLE POSITION MEANS IRRELEVANT)

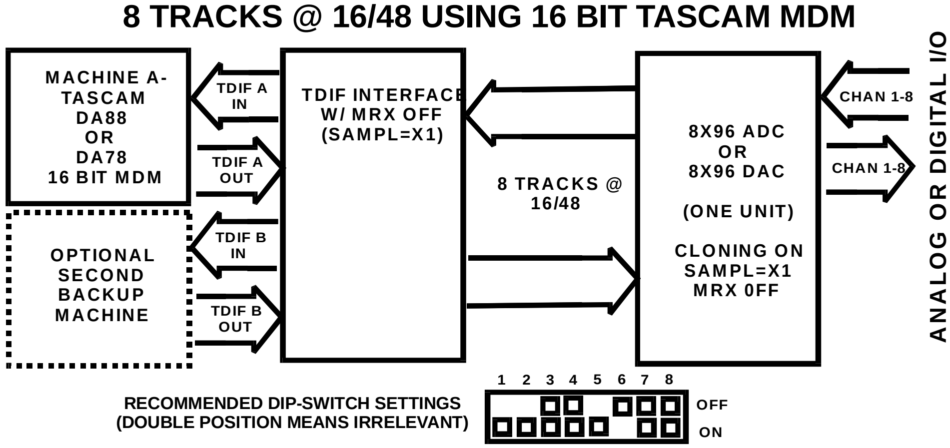
4 TRACKS @ 24/48 USING 16 BIT TASCAM MDM W/MRX



6 TRACKS @ 20/48 USING 16 BIT TASCAM MDM W/MRX



8 TRACKS @ 16/48 USING 16 BIT TASCAM MDM



ADAT Interface (with S/MUX ™)

ADAT MDM interface is equipped with two ADAT ports, A an B which allow to operate with either one or two ADAT MDM machines or interface to other ADAT compatible equipment for up to 8 channels of I/O and signals up to 24/96. ADAT interface uses clock embedded in the optical signal and no wordclock is required to operate the interface although it can be used when appropriate.

The diagrams below illustrate most popular system applications. We recommend one tailored to your equipment and needs. Because of sample accurate operation of ADAT MDMs several 8X96 units can be stacked to increase number of channels. In this case the first ADC is usually the clock master while the rest are clock slaves.
Detailed description of dip-switch functions can be found in chapter "dip-switch settings"). Not all possible setup configurations are mentioned (such as e.g. 16 bit/96 k)

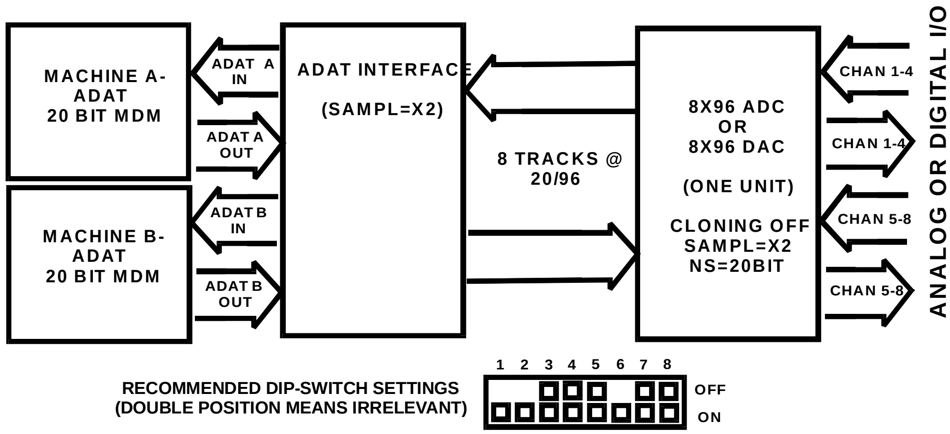
48 or 96 numbers are used as an abbreviation. They mean 44.1 kHz and 48 kHz OR 88.2 kHz and 96 kHz sampling frequency.

Users of 20 bit ADAT MDMs, ADAT interface equipped harddisk recorders and certain ADAT interface equipped computer cards (Hammerfall, Sonorus Studi/o) can take advantage of 96kHz capability of Mytek ADAT interface which is achieved by employing sample splitting technique known as Sonorus S/MUX (tm). S/MUX ™ sample splitting has been design to allow 96k operation using standard 48k ADAT interface transmission, where one high speed 96k channel is split between 2 48k channels (tracks). This technique is compatible with dual wire AES, Prism Sound Split96 (tm) and recording technique used in Tascam DA98HR. In other words 96k tapes recorded on DA78HR using Mytek 8X96 can be played back on DA98HR without the need of any decoder. Read more in chapter "S/MUX"

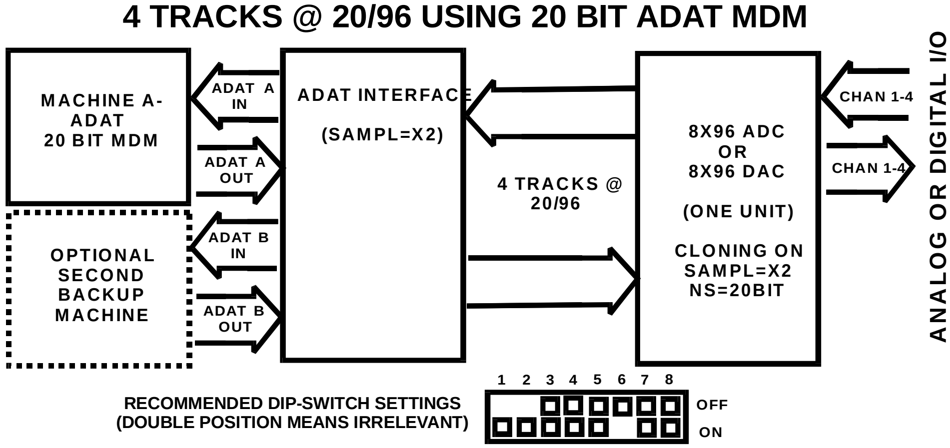
For these concerned with 20 vs. 24 bit sound quality, we'd like to assure you, that signal recorded at 20 bit using sophisticated SupershaperHR ™ is virtually indistinguishable with 24 bit. As a matter of fact we challenge anybody who can distinguish these two settings, especially at 96kHz. We can safely say that 20 bit 96k masters produced with Mytek 8X96 will outperform most of other 24/96 masters produced with different equipment.

8 TRACKS @ 20/96 USING 20 BIT ADAT MDM

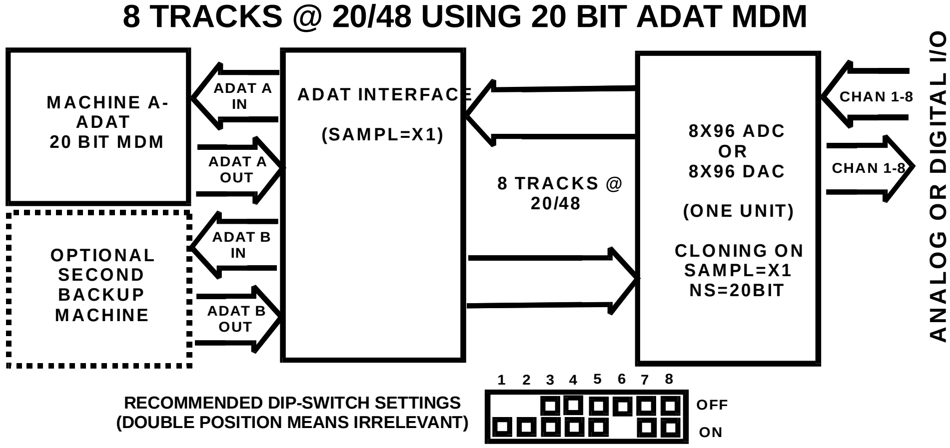
Our favorite, most elegant use of ADAT 20 bit MDM and 8X96



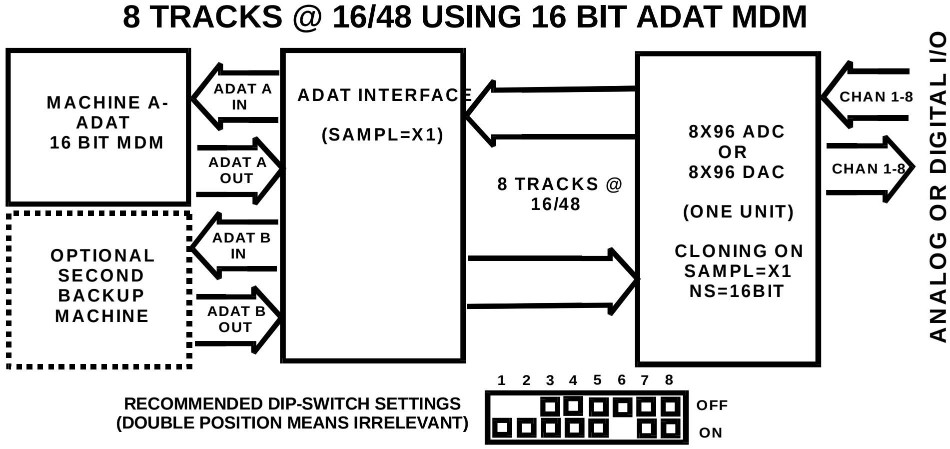
4 TRACKS @ 20/96 USING 20 BIT ADAT MDM



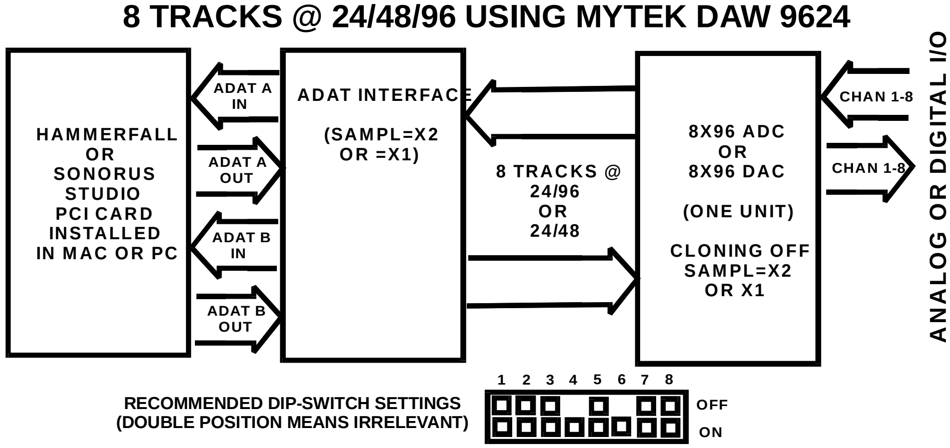
8 TRACKS @ 20/48 USING 20 BIT ADAT MDM



8 TRACKS @ 16/48 USING 16 BIT ADAT MDM



8 TRACKS @ 24/48/96 USING MYTEK DAW 9624



Sadie/ dual AES Interface

Sadie/dual AES interface is (as of Nov 2000) not yet available. Once released the card will feature 2 sets of 8 channel AES/EBU I/O on two D25 connectors. The interface will allow interfacing with up to two Sadie 2496 cards for 8 channel of up to 24/96 signal. The card AES/EBU interface will operate in “dual wire” mode additionally allowing for format conversion to hi-speed AES/EBU and other installed interfaces such as TDIF, ADAT, Protools Compatible or Sonic.

Further functional description is a work in progress. Check for new version of manual in a few weeks

Sonic HDSP/USP Interface

Sonic HDSP/USP interface is (as of Nov 2000) not yet available. Once released the card will feature set of 8 channel of up to 24/96 I/O on a HDSP/USP compatible miniD68 connector for direct connection to Sonic PCI card. There will be an additional port B on a second miniD68 connector for daisy-chaining additional HDSP/USP compatible equipment. The card will allow for format conversion to hi-speed AES/EBU effectively replacing dedicated Sonic HD I/O box and to other installed interfaces such as TDIF, ADAT, Protools ® Compatible or Dual AES.

Further functional description is a work in progress. Check for new version of manual in a few weeks

S/MUX ™ encoder and decoder

The S/MUX ™ sample splitting method was originally developed by Sonorus Inc, in order to provide means of transmitting 24/96 data over 48k ADAT lightpipe. S/MUX stands for "sample multiplexing". S/MUX ™ sample splitting has been design to allow 96k operation using standard 48k ADAT interface transmission. where one high speed 96k channel is split between 2 48k channels (tracks). This technique is compatible with dual wire AES, Prism Sound Split96 (tm) and the recording technique used in Tascam DA98HR. Mytek also incorporates S/MUX splitting on the TDIF card in order to record 24/96 using new Tascam HR MDM machines.

S/MUX is currently (Nov 2000) implemented in Mytek ADAT and TDIF interfaces. S/MUX function is automatically engaged whenever 2X sampling switch is on. For detailed options on using S/MUX with various 48k equipment see: "Dip switch options". The 8X96 ADC equipped with an ADAT or TDIF Card can both encode and decode S/MUX signals. Similarly, the 8X96 DAC can also encode and decode S/MUX.

S/MUX was developed method of de/multiplexing hi-sampling rates data in order to feed them through a standard 44/48k ADAT Lightpipe interface and onto a 48k machine or inside a computer DAW through the Sonorus Studi/o PCI interface card. In short, a 24 bit 96k signal is demultiplexed: odd samples are sent at 48kHz via the odd channel (e.g. #1) and even samples are sent via the even channel (e.g. #2). As a result, a single ADAT Lightpipe can carry 4 channels of 24 bit, 88/96kHz signal. The Mytek ADAT daughter card and Sonorus Studi/o PCI Interface both have two sets of Lightpipes, and thus are capable of transmitting 8 channels of 24 bit 96kHz audio. This method is the foundation of the Mytek/Sonorus collaboration on a hardware front end for the multichannel DAW9624™ which is described later in the manual.

More information on S/MUX ™ can be obtained from Sonorus Inc. at:
http://www.sonorus.com/audio/smux.pdf

An excerpt from this paper follows:

S/MUX is our shorthand for Sample Multiplexing. Multiplexing is the process of combining multiple streams into one stream (in this case, the streams are audio samples). Demultiplexing is the opposite process of splitting a single stream into multiple streams. The basic idea of S/MUX is to join together multiple audio channels in order to represent a single higher bandwidth channel. Unlike current bit-splitting technology, with S/MUX each sample is contained within one of the 'bonded' channels (*bchans*), not spread out among them. The audio channels joined together thus determine the resolution of the S/MUX channels (*schans*). If they are 24 bit, the S/MUX channels are 24 bit. If they're 20 bit, the S/MUX channels are 20 bit, and so on. All the bonded channels (*bchans*) are, of course, the same resolution.

This simple idea is made practical through a few conventions:

1. The lowest numbered *bchan* has the earliest-time sample.
2. The next-lowest *bchan* has the next-earliest sample, and so on.
3. The S/MUX channel (*schan*) sample rate is an integer multiple (*N*) of the *bchan* rate (e.g. 2x, 3x, 4x,etc.)
4. The oversampling multiplier *N* is called the *order* of the *schans*, notated S/MUX N (e.g. S/MUX 2 for a 2x, 48KHz to 96KHz system).
5. The first group of *bchans {1..N}* represent the first *schan*.
6. The second group of *bchans {N+1..2N}* represent the second *schan*, and so on.
7. Though any integer N can be used for the order, powers of two are recommended (2, 4, 8, 16, etc.)
8. No non-audio signaling will be present in the audio streams. Though this means the user (or system) needs to make not of the S/MUX order, it also means maximum possible fidelity (using every bit for audio data) as well as simplicity of implementation.
9. It is possible to use *schans* of different orders simultaneously though care should be taken to avoid confusion.
10. S/MUX compatible systems must not filter the audio in any way. Since each channel only sees a fraction (1/N) of the total *schan* data, any filtering would be invalid (e.g. lowpass, bandpass, highpass filters).
11. Though ADAT lightpipe is the first implementation, the concept is applicable to any digital audio format. In fact, the AES is working on a standard for just such a system using AES/EBU streams.
12. S/MUX is compatible with dithering and gain changes, as long as they are done exactly the same way for each bchan in a schan group. That is, you can dither or change the gain of the schan by applying the processing uniformly to the bchans.

Prism MRX ™ encoder and decoder

Mytek has acquired a license for use of the MRX ™ bitsplitting method, developed by Prism of U.K. and used in their popular MR2024T interface.
MRX bitsplitting is intended to pack higher resolution signals onto 8 channels of Tascam 16 bit DA88 and DA38 machines. The following formats are possible:

MODE1 - 4 channels of 24 bit @44/48k are packed onto 8 tracks of 16 bit.

MODE2 - 6 channels of 20 bit @44/48k are packed onto 8 tracks of 16 bit.

MODE1-96 - 2 channels of 24 bit @88/96k are packed onto 8 tracks of 16 bit.

MODE2-96 - 3 channels of 20 bit @88/96k are packed onto 8 tracks of 16 bit.

MODE*-96 is engaged automatically whenever 2X sampling switch is on. For additional details see “Dip switch options”.

The MR2024T also encodes a special control signal that can be detected in order to determine whether mode 1 or 2 was used.

TDIF card can both encode and decode MRX signals.

It’s recommended that one use the MRX™ with 16 bit machines only, unless you are making a 24 bit stereo master. If you own 20 bit ADAT, a 24 bit signal noise shaped to 20 bit will practically produce the same results, with no future problems decoding the tapes. Similarly a 20 bit ADAT can be used as 4 channel 20 bit/96k machine with absolutely stellar results (see S/MUX).

If you are overdubbing and punching-in, do not use MODE2 of MRX as it cannot produce smooth crossfades. In MODE1 punch-in is possible , but double-check the quality of the crossfades when recording audiophile quality acoustic music.

Using the 8X96 ADC and /or DAC

As a set of converters, the 8X96 ADC and DAC have a number of applications. The goal is always to get a substantial improvement of sound quality - the sound of your digital recorder is always determined by the converters - even a 16 bit ADAT can be turn into 24 bit, 96kHz stereo digital master recorder. The 8X96 sound quality is far superior to any stock piece of digital audio equipment. The 8X96 has 20db less distortion than the sound of an average studio DAT machine. As a result, the signal is virtually free of digital artifacts and remains a faithful mirror of the input – Mytek converters are designed for **transparency**, not coloration.

The 8X96 converters are designed for the most demanding mastering and purist recording applications. The sound quality is superior, even when 44/48k or 16 bit recording is used.

There is a substantial quality improvement when either ADC or DAC is used. If a the total improvement was described as 100%, roughly 60% could be attributed to ADC and 40% to DAC.

The ADC and DAC can be used for stereo, surround sound and DVD applications and multitrack applications with channel configurations in multiples of 8.

Typical applications include:

Front End for Tascam and ADAT MDM

Stereo 24 bit 96kHz recording and mastering (16 bit Tascam)

Multitrack 96kHz recording (24 bit Tascam and 20 bit ADAT)

Multitrack 24 or 20 bit recording (16,20,24 bit MDMs)

Remote recording and long runs of fiberoptics

Front end for Protools ®

Front end for Sonic Solutions HDSP/USP ®

Front end for Sadie ®

Front end for MOTU ®

Front end for Merging Technologies Pyramix ®

Front end for AudioCube ®

Front end for other DAWs

Front end for Tascam MX-2424 ®

Front end for Mackie HDR2496 ®

Front end for Sony DMX—100R ® and other digital consoles

Front end for DAW

8X96 Converters are one of the most flexible choices for a high performance hardware front end. Mytek focuses exclusively on the performance of the converters and therefore can offer a solution superior to any offered by a DAW manufacturer. A choice of Daughter Interface cards allows for direct interfacing with several popular DAW's: Native Nuendo systems and Sonic Solutions HDSP ™. 8X96 Converters interface well with a number of other digital IO interfaces such as RME Hammerfall, Lynx AES16, MOTU and many cards from Digital Audio Labs, Frontier Design, Event Electronics, etc.

8X96 Converters support nearly all physical and data formats as long as the particular software/DIO-card supports them.

8X96 provides a DAW with following features:

1. Superior multichannel 16,20, or 24 bit digital audio conversion with sampling rates up to 100kHz. Channels I/O can be stacked in increments of 8 to an unlimited number.
2. Digital format conversion between a large number of digital formats.
3. Direct connection to selected DIO cards without a need for native hardware.
4. Stable (jitter of <10ps) clock system.
5. External wordclock synchronization.
6. Interfacing and format conversion with all available bit splitting algorithms for high resolution editing and mastering- MRX™, and S/MUX ™ are supported.

Mytek, in cooperation with Sonorus, Inc. has also developed its own high performance, cost effective multichannel DAW hardware solution called DAW9624™ described in the next chapter.

DAW9624 ™ package

with Lynx AES16 or RME Hammerfall

Mytek converters, can be found in the front end of many professional DAWs such as Sonic Solutions, Sadie and Protools 24. The superior performance of their 24 bit sound is fully appreciated by the users. Not everyone though, could enjoy benefits of high end 24 bit sound without spending a fortune on the DAW hardware.

For the last couple of years we were experimenting with several editing packages, closely watching spectacular progress in the development of "used to be lower end" audio software market. Continuous increase of computer speed and the recent move to floating 32 bit sample resolution and 96 kHz sample rate, have dramatically closed the gap between these and the $10,000+ professional editing packages. In our tests several low cost 24 bit software packages such as Nuendo, Logic Audio, Sequoia, VegasPro, Spark, Peak, Samplitude, Wavelab or Cubase coupled with good processing packages, produced results rivaling to what could be achieved with the high end audio editing software. They may lack certain "comfort and ergonomics" features but, if used properly, don't lug behind with sound quality , the feature that is dearest to Mytek and never compromised in our designs. The key is at least 32 bit signal processing.
There is a multitude of growing choices, and abundance of cost effective, high performance software plug-ins, often not found in the high end, "closed platform" editing software.

We have quickly realized, that buyers who decide to built their studio around that software, are faced with choices of several cheap hardware solutions of mediocre sound quality. Costing less than a $1000 they may claim 24 bit performance but deliver cheap DAT machine quality sound and no features, an understandable result of "made in XX", "cut costs" hardware design.

Mytek has a reputation of building first grade audiophile quality audio converters. Our new products that include 8X96 Series 8 channel 24 bit, 96 kHz ADC and DAC, push performance further, to 120dB dynamic range, flat to 40 kHz, truly transparent audiophile sound.

In Spring of 1998 Mytek, Inc. has teamed its R&D efforts with Sonorus, Inc. of New York, makers of STUDI/O (tm) PCI audio interface, and developed S/MUX ADAT interface, bit split method of transmitting 8 channel of 24/96 over two ADAT lightpipes, similar to “dual wire” 96k AES/EBU standard. Subsequently, a German Company RME has announced the Hammerfall 9652 PCI card, a 16 to 24 channel ADAT card with absolutely unrivaled speed and latency performance. Hammerfall has S/MUX implemented in the hardware and it’s the fastest card yet under our tests. Sonorus and RME have built reputation of technical excellence and
excellent customer support. Hammerfall 9652 and Sonorus STUDI/O interfaces is now supported by all major operating systems (MacOS, Windows 9X, NT and 2000 and

Linux) and all general audio software packages available today. S/MUX interface drivers required for 24 bit 96kHz work are also available for almost every software package.

To support S/MUX Mytek has developed appropriate ADAT lightpipe 24 bit, 96 kHz interface daughter card for our 8X96 ADC and DAC. The result is DAW 9624(tm) - very high performance 8+ channel, 24 bit, 96 kHz DAW hardware package, an open hardware platform capable of running most of audio software available today.

RME Hammerfall is equipped with 2 24 bit ADAT lightpipes that can be configured to carry 16 channels of 24 bit, 48 (or 44.1) kHz or 8 channels of 24 bit, 96 (or 88.2) kHz audio. The 8X96 converters provide analog conversion, attenuation of ADAT interface clock jitter, conversion to/from AES/EBU and optional formats such as TDIF and wordclok in/out for external synchronization. DAW
9624(tm) modularity allows for setup consisting of any number of I/O channels supported by particular software in increments of 8. For example, a setup of 8 inputs and 32 outputs @ 24 bit, 48 kHz requires 2 PCI cards, 1 8X96 ADC and 4 8X96 DACs. This setup, can also be used for 8/16 channels of 24 bit, 96 kHz with a flip of a switch. DAW 9624(tm) is sold as a system (PCI card (1pc), 8X96 ADC (1pc) and 8X96 DAC (1pc)) or a la carte, per piece. Current 24 bit 96 kHz capable Mac or PC software choices include Nuendo, Logic Audio, Spark, Peak, Sonic Foundry Vegas, Sequoia and Samplitude, Cubase VST/24, SAWPro, Wavelab, Cakewalk 8 and a wide choice of plug-ins such as Waves NPP etc. The DAW 9624 will run all 48 kHz and 44.1kHz software as well. Compatible software is described here. Similarly we leave the choice of PCI card to the user. The choice between Hamerfall and Lynx AES16 should be made based on best combination of software and appropriate drivers.

DAW9624 setup was recently used in some spectacular achievements. Read on our website about live transmission of 12 channel of uncompressed 24/96 audio plus video over Internet2 between Montreal and Los Angeles.

DAW 9624 (tm) is MAC, PC and Linux compatible.

Further description of the DAW 9624 ™ is available at:

http://www.mytekdigital.com

Warranty

This 8X96 Series digital audio converter is warranted by Mytek to the original purchaser against defects in workmanship and materials used in manufacture for a period of one year from the date of purchase. Faults due to customer misuse, unauthorized modifications or accidents are not covered by this warranty.
No other warranty is expressed or implied.
Any faulty unit should be sent, shipping prepaid, to the manufacturer. The serial number of the unit should accompany any request for service.