

# AXUM WEBSITE DATA

## DESCRIPTION

the AXUM VERSION 3.0 is a modular completely preprogrammed platform that can grow with you and your organization any time, any moment.

I/O cards can be shared between systems and a Broadcast station that has decided to use the AXUM as their main and support mixers can be sure to have invested in a future proof product.

The Axum can be used for small local 4 fader desktop use as well as large scale (128 faders max) production work and everything in between.

A large matrix of 1280x1280 with extensive signal processing on a maximum of 128 stereo channels is available that can be assigned to Sixteen stereo summing mixing busses and sixteen monitor buses as well as directly between the maximum of 4 DSP's

Imagine a console where you can decide yourself what a switch/encoder or fader will do for you. Not possible?

With the AXUM this is possible and... you can do that yourself by selecting a function out of hundreds of functions simply selectable from a database in your AXUM browser.

Two control surfaces (with their own CRM sections) in different studios sharing the same I/O rack is also no problem for the AXUM platform.

Hardware purchasing choices are straightforward:

Rack ONE accepts a maximum of 21 I/O single slot cards.

Rack TWO accepts a maximum of 37 I/O single slot cards.

A Control Surface Frame starts with 4 faders as a minimum and can be extended to a maximum of 128 motorized faders plus Control/CRM section.

Each individual Control Surface Frame can be mounted drop through in a work surface left or right of your script space.

Interfacing with studio equipment is with high quality low cost readily available STP cable, also called shielded STP Ethernet cable, see breakout boxes below to choose from.

An internally generated webpage allows you to remotely control all settings from anywhere in the world over Ethernet and /or Internet.

The MambaNet protocol controls the communication between all system parts linked to each other with world standard Ethernet.

It gives such an enormous freedom in system solutions so we hardly can imagine that your needs can not be met with the AXUM.

A software Meter application is included and runs directly from the Control Surface, but also from any PC in the network. Also hardware meters can be connected in the network anywhere, thanks to the MambaNet protocol.

## FEATURES

- Matrix up to 1280x1280.
- 128 motorized faders max
- 4 interactive studios connected to one I/O rack possible
- Free assignable/programmable switches
- Remote control software available
- Extensive used security system
- 16 stereo mixing busses.
- (Program | Sub(Rec) | Cue | Comm | 12x stereo Aux).
- 4 stereo monitor busses (max. 16 monitor busses)
- Max. 4 DSP cards per I/O rack.

## SPECIFICATIONS

MIC inputs	<ul style="list-style-type: none"><li>: Electronically balanced</li><li>: Input impedance 2k Ohm</li><li>: Input sensitivity -70dBu up to +20dBu (PAD) (PGA2500)</li><li>: Dynamic Range 118dB (AD converter PCM4202)</li><li>: Total Harmonic Distortion plus Noise -108dB (30dB gain)</li><li>: CMRR MIC inputs: 85dB @ 1kHz, maximum gain</li><li>: Frequency response 20Hz - 20kHz <math>\pm</math> 0.1dB (sample rate 48kHz)</li><li>: Crosstalk 1kHz &lt; -118dB</li><li>: Phantom is switchable +48 Volts</li><li>: Transformer balancing is optional on the break-out panel</li></ul>
Line inputs	<ul style="list-style-type: none"><li>: Electronically balanced</li><li>: Input impedance 10k Ohm</li><li>: Input sensitivity +6dBu, maximum input +26dBu (+/- 20dB gain range).</li><li>: Dynamic Range 118dB (AD converter PCM4202)</li><li>: Total Harmonic Distortion plus Noise -105dB</li><li>: CMRR Line inputs: 30dB @ 1 kHz</li><li>: Frequency response 20Hz - 20kHz <math>\pm</math> 0.1dB (sample rate 48kHz)</li><li>: Crosstalk 1kHz &lt; -123dB</li><li>: Transformer balancing is optional on the brake-out-box</li></ul>
Line Outputs	<ul style="list-style-type: none"><li>: Electronically balanced</li><li>: Output impedance 56R Ohm.</li><li>: Nominal output level +6dBu, maximum output +26dBu</li><li>: Dynamic Range 118dB (AD converter PCM4104)</li><li>: Total Harmonic Distortion plus Noise -100dB</li><li>: Frequency response 20Hz - 20kHz <math>\pm</math> 0.1dB (sample rate 48kHz)</li><li>: Crosstalk 1kHz &lt; -118dB</li><li>: Transformer balancing is optional on the brake-out-box</li></ul>
Phones Output	<ul style="list-style-type: none"><li>: Stereo unbalanced</li><li>: Output impedance 5R Ohm.</li><li>: Nominal output level +6dBu, maximum output +26dBu</li><li>: Max. Output power, 1W into 32R Ohm, 80mW into 600R Ohm</li><li>: Dynamic Range 114dB (AD converter CS4385)</li><li>: Frequency response 20Hz - 20kHz <math>\pm</math> 0.1dB (sample rate 48kHz)</li></ul>
Digital Inputs	<ul style="list-style-type: none"><li>: AES/EBU (AES3) or S/P-DIF Transformer balanced</li><li>: Input Impedance: 110R Ohm / 75R Ohm (jumper setting)</li><li>: Differential input sensitivity 200mV</li><li>: Dynamic Range (sample rate converter) 144dB</li><li>: Total Harmonic Distortion plus Noise (sample rate converter) -140dB</li><li>: 16/20/24 bit, 32 kHz to 96 kHz (optional built in sample rate converter)</li></ul>
Digital outputs	<ul style="list-style-type: none"><li>: AES/EBU (AES3) or S/P-DIF Transformer balanced</li><li>: Output Impedance: 110R Ohm/75R Ohm</li><li>: Output level: AES3 5 Vpp, S/P-DIF 1Vpp</li><li>: 16/20/24 bit, 32kHz, 44.1kHz ore 48kHz</li><li>: Total Harmonic Distortion plus Noise -140dB</li></ul>
GPIOs (8mA) out	<ul style="list-style-type: none"><li>: All GPO's are by opto isolated relays able to handle a max of 50V at 200mA or 5V TTL 560E 560R</li><li>: All GPI's have a 5V TTL 100kOhm circuitry. GPIO-MIC has a 5V/560Ohm LED driver circuit</li></ul>

EQ is 6 band, any band can perform one of the following functions/specifications:

Off	: no function
High Pass Filter	: +/- 18 dB (10Hz up to 20 kHz shelving/bell/notch), Q: 0.1 to 10 variable.
Low Shelf	: +/- 18 dB (10Hz up to 20 kHz shelving/bell/notch), Q: 0.1 to 10 variable.
Peaking	: +/- 18 dB (10Hz up to 20 kHz shelving/bell/notch), Q: 0.1 to 10 variable
High Shelving	: +/- 18 dB (10Hz up to 20 kHz shelving/bell/notch), Q: 0.1 to 10 variable
Low Pass Filter	: +/- 18 dB (10Hz up to 20 kHz shelving/bell/notch), Q: 0.1 to 10 variable
Band Pass Filter	: +/- 18 dB (10Hz up to 20 kHz shelving/bell/notch), Q: 0.1 to 10 variable
Notch Filter	: +/- 18 dB (10Hz up to 20 kHz shelving/bell/notch), Q: 0.1 to 10 variable
<b>DYNAMICS</b>	: Interactive one knob control of threshold, compression ratio, expander ratio as well as attack and release times.
<b>Processing</b>	: 32 bit floating point
<b>Channels</b>	: 32 stereo channels per DSP card.
<b>Busses</b>	: 16 stereo mixing busses, free assignable Prog/CUE/Aux etc. etc.
<b>Monitor busses</b>	: 4 stereo monitor busses per DSP card.

**Additional information:**

By configuration its possible to create more separate mixers, for example two consoles of 6 stereo channels to 8 stereo mixing busses, 2 stereo monitor busses.

A maximum of 4 DSP cards may be inserted giving you a mixing console of 128 stereo channels, 16 stereo mixing busses and 16 stereo monitor busses.

**Module processing:**

- Gain
- Low cut
- Insert
- 6 bands full parametric EQ
- D&R designed one knob dynamics
- 16 buss sends pre or post fader

**Overall Level**

- : 0dBu=0.775Vrms
- : 0dB internal = -20 dBFs.

**Clock**

- : Sample rate: 32kHz, 44.1kHz, 48kHz, +/- 20ppm (internally synchronized)
- : External sync: 32kHz, 44.1kHz, 48kHz +/- 50ppm
- : Jitter max 150pSec

**Power supply**

- : Neutrik™ PowerCon™ (delivered in the package).
- : 100-240 Volt, 50/60Hz (1.7A Max)

More detailed Specifications of the Burr Brown A to D and D to A Chips we use can be found on the following link <http://www.ti.com/lit/ds/symlink/src4392.pdf>

The Mic card uses the PGA2500 digital mic input chip followed by the 4392 chip as sample rate converter in 24 bits

The Line in card used the PCM 4202 A to D 24 bit converter chip

The Line Out card uses the PCM 4101 D to A 24 bit converter chip

We also confirm that the specs of our console can easily meet or succeed the below specifications for inputs and outputs.

Analog audio input/output, THD+Noise 0.0005% @-20dBFS

Digital audio input/output, THD+Noise 0.002% @-20dBFS

Mic/line input, THD+Noise 0.036% @-20dBFS

**VIDEO**

<http://www.d-r.nl/wiki/dokuwiki/doku.php?id=video>

<http://www.youtube.com/watch?v=X1e7gv1rtuU>

<http://www.youtube.com/watch?v=WVsGBqBkvWY&feature=related>

<https://youtu.be/XzEOvpUdCmk>

<https://youtu.be/QRfEYkgx0xc>

<https://youtu.be/RGrdBpGZfh4>

#### DOWNLOADS

<http://www.d-r.nl/assets/1-axum-manual-3.0.pdf>

<http://www.d-r.nl/assets/axum-remote-manual-1.0.pdf>

<http://www.d-r.nl/assets/axum---introduction.pdf>

<http://www.d-r.nl/assets/axum-brochure-the-facts.pdf>

<http://www.d-r.nl/assets/axum-getting-started.pdf>

[http://www.d-r.nl/assets/axum-drop-through-sections-\(dimensions\).pdf](http://www.d-r.nl/assets/axum-drop-through-sections-(dimensions).pdf)