

# GXA8 Operational Manual

.....

Revision 1.0



# Contents

CONTENTS	I
IMPORTANT NOTES	1
Unpacking Power Location Mounting Positions Conformity Technical Support	1 1 1 1
LAYOUT OF THIS MANUAL	2
INTRODUCTION TO THE GXA8	3
Description OPTIONS FRONT PANEL DESCRIPTION	3
Panel Buttons	
AUDIO	5
AUDIO FORMATS	555566
CLOCKS	7
Lock Mode Precision Wide Clock Source Master External Word Clock Slot One Slot Two Sample Rate	7 7 7 7 7 7 7 7
METERS AND INPUT LEVEL	9
Peak Hold Fine PPM Over Sensitivity Input Gain Level	9 9
BACK PANEL	0
SPECIFICATION1	1
A/D Performance	

## Important notes

## Unpacking

After unpacking the GXA8, save all the packing materials in case you ever need to ship the unit. Thoroughly inspect the GXA8 for signs of damage. Report any shipment damage to the carrier at once. The following accessories are included with the GXA8:-

- 1. Power Cable.
- 2. User Guide.
- 3. Four mounting feet and screws.
- 4. Four rack mounting bolts.

#### Power

- This unit is equipped with voltage-selectable linear power supply. Please ensure that the correct voltage for you region is selected (110V or 220V) before plugging into the supply.
- Make sure that the unit is turned off before connecting the power plug to the socket.
- Please be sure to connect the power cord to the AC inlet on this unit before connecting the power plug to the socket.
- When disconnecting the power plug from the socket, do not pull the cord but hold the plug to avoid damaging the cord.
- Avoid damaging the power cord.
- If the unit is not to be used for a long period of time, unplug the cord from the socket.
- It is normal for this unit to become warm whilst being operated.

## Location

- Avoid using this device in extreme heat, humidity or where it may be affected by dust or vibration.
- Do not place or drop anything heavy on the unit or its power cable.
- Ensure that the GXA8 has adequate ventilation space on all sides. We recommend at least 1U rack space top and bottom.

#### **Mounting Positions**

Do not mount the GXA8 upside down.

#### Conformity

The GXA8 bears the 'CE' mark and is therefore fully compliant. Tests performed and passed are as follows,

EN50082-1: 1992

EN55022: 1994 Class A

## **Technical Support**

Technical Support is available on line through the Genex web site at

http://www.genex.co.uk/ or by emailing: support@genex.co.uk.

## Layout of this Manual

Sections in this manual appear in the following order.

- 1. A brief introduction to the feature set and a list of options.
- 2. A detailed front panel description with a functional description of every front panel button and display. This section should be used as a quick guide and reminder to basic front panel operations.
- 3. A description of the audio formats supported
- 4. An explanation of the clock structure
- 5. Rear panel schematics and connector details

## Introduction to the GXA8

## Description

The GXA8 is a professional, 8-channel, analogue to digital audio converter, supporting all standard sample rates from 44.1kHz up to 192kHz at resolutions of 16, 20 or 24 bits. It also supports the Sony DSD format (1-bit, 2.8224MHz sample rate  $\Delta\Sigma$ ) when married with the DSD upgrade kit (available separately).

All settings are available from the intuitive front panel with no complex menu structures to navigate. The front panel also includes 8 high resolution peak power meters (PPMs), one for each channel.

As standard, the GXA8 is fitted with four AES/EBU digital outputs providing eight channels of digital audio output up to 96kHz and four channels of output for 192kHz (an additional four channels are available via the AES expansion card). In addition, expansion cards with ADAT, TDIF, SDIF and workstation interfaces on are available.

## Options

- 8-channel DSD upgrade kit. Includes the  $\Delta\Sigma$  modulator i.c.s and the output and processor card
- 8-channel AES expansion input and output card
- ADAT input / output card
- TDIF input / output card
- 8-channel SDIF2 output card
- Workstation input / output card

# **Front Panel Description**

## **Panel Buttons**

#### 1. Mute

Mutes the digital audio output from the unit

#### 2. Over Sensitivity

This button determines how many consecutive maximum value sample will trigger the "OVER" LED to light

#### 3. Peak Delay

Sets the peak hold of the PPMs to either on, delay (2 seconds) or off

#### 4. Fine PPM

Changes the PPM mode to a scale of  $\pm 1$ dB centred around -12, -14, -16, -18 and -20dBFS to aid input level calibration

#### 5. Clear Peak Hold

Clears any peak or over LEDs on the PPMs that are lit

#### 6. Configuration

Selects between the following input sources: Analogue, slot one (digital) and slot two (digital). In addition, for analogue input the output format can be set to either PCM or DSD (DSD required the DSD upgrade kit)

#### 7. Lock Mode

Selects between the "wide" and "precision" lock modes when the unit is locked to an external clock source.

#### 8. Clock Source

Selects the clock source for the unit – master (internal precision-cut quartz crystal), external word clock, slot one or slot two

#### 9. AES Output

Chooses between single, dual or quadwire. Obviously, not all modes are available for all sample rates

#### 10. Bit Split

Selects whether to encode or decode (or nothing) a PAQRAT bit-splitting format (PCM mode) or a 4-wire "DSD over AES" format (DSD mode, encoding only).

#### 11. Bit Depth

Selects between 24, 20 or 16 bit fully dithered output resolution.

#### 12. Sample Rate

Allows selection of 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz.

This button does not function in DSD mode (forced to 44.1kHz) or in slave clock mode (locks automatically to input rate)

#### 13. PPM DISPLAY

The PPM displays consists of eight columns of 14 segments each, representing channels 1 to 8 respectively. The meters are digital and are calibrated in dBFS, with the top segment representing 0dBFS. Peak values can be set to decay immediately, be held for 2 seconds before decaying, or can be held until the CLEAR PEAK HOLD button is pressed.

Note the overload segments remain on until cleared by the pressing the CLEAR PEAK HOLD button.

## Audio

## **Audio Formats**

This section describes how to configure the audio output section of the GXA8. The following parameters can be adjusted;

#### **Standard Output Format**

AES.

#### Input/Output Formats Available Through Expansion Cards

AES (expansion card to provide 8 channel dual wire 192kHz output)

SDIF-2 output

SDIF-2 input

TDIF

ADAT

DSD (Direct Stream Digital) - output only

#### What is Direct Stream Digital?

DSD was invented and developed by the Sony Corporation. It is designed to remove some of the potential artifacts introduced by the decimation filter in a conventional analogue to PCM digital conversion process. The DSD modulator generates a 2.8224MHz bitstream per channel, which is a slightly larger bandwidth than for 24 bit 96kHz PCM audio.

#### **AES Output Configuration**

You can select single wire, dual wire or quad wire. The setting refers to the position of consecutive samples within an audio carrier.

In single wire mode each consecutive audio sample for a given track is located within that track. For the AES format this means that one AES cable can carry two tracks of information.

Dual wire mode means that each consecutive audio sample is contained in adjacent tracks. For the AES format this means that one AES cable can carry only one track of information.

Quad wire mode means that successive audio samples rotate through four adjacent tracks. For the AES format this means that two AES cables carry one track of information.

In master clock mode:-

at 44.1kHz or 48kHz sample-rate, the AES output is single wire and the button is disabled.

at 88.2kHz or 96kHz sample-rate, the AES output can be either single or dual wire.

at 176.4kHz or 192kHz sample-rate, the AES output can be either dual or quad wire. In slave clock mode:

with an input clock of 44.1kHz or 48kHz, the AES output can be single, dual or quad with an input clock of 88.2kHz or 96kHz, the AES output can be single or dual

Output configuration available		SAMPLE RATE						
		44.1kHz	48kHz	88.2kHz	96kHz	176.4kHz	192kHz	
NUMBER OF TRACKS	1-2 Tracks	Single	Single	Single Dual	Single Dual	Dual Quad	Dual Quad	
	3-4 Tracks	Single	Single	Single Dual	Single Dual	Dual Quad*	Dual Quad*	
	5-6 Tracks	Single	Single	Single Dual*	Single Dual*	Dual*	Dual*	
	7-8 Tracks	Single	Single	Single Dual*	Single Dual*	Dual*	Dual*	

\* - only available with the AES, TDIF, ADAT or SDIF expansion card in slot 2

#### **Bits Out and Dither**

This setting truncates the output audio to the selected number of bits. Dither is added automatically.

Available choices of bit depth are 16, 20 and 24.

#### **Bit Split – Encode or Decode**

Two different bit-split formats are used depending on whether the GXA8 is in PCM or DSD mode. DECODE will only function on digital inputs via AES, TDIF, ADAT and SDIF (input) cards.

PCM mode. The Paqrat<sup>®</sup> format is supported. This allows 24-bit recordings to be made on legacy 16-bit recorders. Two channels on the recorder are required per 24-bit audio channel. The left channel contains the first 16-bits and may be monitored. The right channel contains the top eight bits plus a rather unpleasant tone (750Hz at 48kHz; 689Hz at 44.1kHz) at -19dBFS. The tone provides signal present confirmation for the level meters on the recorder but should not be monitored through any speakers.

DSD mode. Normally in DSD mode the AES outputs provide a decimated version of the DSD signal at 44.1kHz. This is to allow back-up PCM recording for example and may also be dithered to 20 or 16-bits. However, selecting ENCODE bit-maps two channels (channels one and two) of DSD across eight 16-bit PCM channels (i.e., 4 AES cables). This allows stereo DSD recordings to be made on legacy 8-channel 16-bit PCM machines. Tape based recorders may still not be suitable however as the error correction techniques that have to be employed for digital tape will further corrupt the DSD stream over and above any errors on the tape in the first place! DECODE is not available for this format.

#### Audio Mute

The mute button mutes the audio at the outputs only (AES and any slot cards). The PPMs will still display the input level. This allows gain levels to be set with test tones without having to listen!

The exception is DSD mode. A "zero" bitstream in DSD is interpreted as a full scale DC voltage which is undesirable. Therefore the audio is muted at the converters and as a result PPM readings will be muted also. However, the PPMs may read a very low level when muted. This is normal for the converters chips being used for DSD.

## Clocks

It is vitally important that the clock used at the point of conversion between the analogue and digital domains is stable and accurate. Ideally the GXA8 should be the clock master wherever possible, as the unit uses a very stable fixed crystal oscillator to generate it's own clocks. However, operating with the GXA8 in master mode is not always possible, so the unit provides several methods for locking to external sources where necessary.

## Lock Mode

#### Precision

The precision locking mode should always be selected in the first instance. The precision mode uses a very heavily damped phase locked loop that is designed to severely attenuate any clock jitter present on the input source. The downside is, that if the input clock varies or drifts too much in frequency, then lock will not be possible.

#### Wide

Wide mode should be used when the precision mode is not capable of locking to the external source. A more relaxed phase locked loop is selected in this mode, with less attenuation of jitter.

#### **Clock Source**

The clock source can be either Master, external word-clock, slot one or slot two.

#### Master

This selects the precision quartz crystal oscillator. This is the preferred lock source. The audio converter should always be the clock master wherever possible.

#### External Word Clock

This selects the word clock input BNC at the back of the unit. The frequency should be x1 sample rate for single wire,  $\frac{1}{2}$  sample rate for dual-wire and  $\frac{1}{4}$  sample rate for quad-wire.

#### Slot One

This selects the card in slot 1 as the clock source.

#### Slot Two

This selects the card in slot 2 as the clock source.

Note that the DSD and SDIF cards cannot provide a clock source.

#### **Sample Rate**

The GXA8 supports the following PCM sample rates:-

44.1kHz; 48kHz; 88.2kHz; 96kHz; 176kHz; 192kHz

In addition, when locked to an external clock, the GXA8 supports all modified rates based around 44.1kHz and 48kHz ( $\pm$ 0.1%,  $\pm$ 4%). The GXA8 will also lock to 32kHz. Input **clock** rates from 32kHz to 50kHz, plus 88.2kHz and 96kHz may be applied.

When the GXA8 is in MASTER clock mode, the sample-rate is selected pressing the sample rate button until the desired rate is indicated.

When locked to an external clock source, the incoming clock rate will be calculated and displayed. The **wide** or **PRECISION** LED will flash until the unit has obtained lock. If the incoming rate is a "modified" rate, then the up or down arrow will also light to indicate a pull-up or pull-down rate respectively (see note below). The sample rate button is disabled in this case. However, single, dual or quad-wire mode can now be selected. The indicated rate will change based on the output wire configuration. For example:-

- A clock rate of 48kHz is connected to the clock input and detected:
  - In single-wire mode, a sample-rate of 48kHz will be indicated In dual-wire mode, a sample-rate of 96kHz will be indicated In quad-wire mode, a sample-rate of 192kHz will be indicated

In this way, the GXA8 always displays the actual **sample** rate rather than the external **clock** rate. Note that when locked to a clock rate of 88.2kHz or 96kHz only single or dual-wire mode may be selected.

**NOTE:** When either of the pull-up / pull-down LEDs are lit, the GXA8 is using a fast reacting, wide locking RC Phase Locked Loop instead of its variable crystal oscillators. As a result, the GXA8 may not notice if you subsequently change the incoming sample rate, particularly if you change from a  $\pm 0.1\%$  modified to the unmodified rate. In this case, to ensure the GXA8 locks to the new unmodified rate using its variable crystal oscillator (which has far greater jitter performance and thereby greater audio conversion quality), the unit should be forced to relock (just cycle the unit through to master clock mode and back again).

## Meters and Input Level

Eight high resolution peak power meters (PPMs), one for each channel, are provided. Four buttons are provided to configure the meters.

## **Peak Hold**

The meters default to peak hold **on** on power-up. The peak value of any signal will be held until the **CLEAR PEAK HOLD** button is pressed.

With peak hold **delay**, each peak value is held for approximately 2 seconds or until the **CLEAR PEAK HOLD** button is pressed.

With peak hold set to off (both indicators off) no peak values will be held.

The **over** LEDs will always light when an over condition is detected. Setting peak hold to off or delay has no effect. They can only be cleared down by pressing **CLEAR PEAK HOLD**.

## Fine PPM

This function changes the resolution and display style of the PPMs. The range of the PPMs changes to  $\pm 1$ dB centred around -12dBFS, -14dBFS, -16dBFS, -18dBFS or -20dBFS.

## **Over Sensitivity**

This allows the number of adjacent "over" samples (i.e., samples that are must be detected before the **over** LEDs are lit) to be set from one to six.

## Input Gain Level

Below each PPM is a trim position to set up the input level for each channel. The input gain range is from -10dBu to +6dBu (for a -18dBFS input level). Therefore for full-scale (0dBFS) the GXA8 can accept inputs from +8dBu to +24dBu.

## **Back Panel**

The back panel contains:-

- Eight balanced analogue inputs on XLR connectors
- Four balanced AES-EBU outputs in XLR connectors
- One word-clock input on  $75\Omega$  BNC. This should be externally terminated.
- One work-clock output on  $75\Omega$  BNC.
- Two slot interfaces. The top slot is Slot One. These accept a range of Genex options cards
- One AC mains input on an IEC connector. Mains supply must be 100-120V or 200-240V, 50/60Hz. The correct voltage range must be selected via the voltage selection switch above the IEC mains input. Failure to do so may result in severe damage to the unit.

# Specification

## **A/D** Performance

Resolution	
Signal / Noise ratio (un-weighted)	better than 113.5dB @ 22kHz bandwidth
Signal / Noise ratio (A-weighted)	better than 117dB @ 22kHz bandwidth
THD + Noise	better than -103dB @ -1dBFS, 1kHz
Frequency Response (±0.5dB)	5Hz to 22kHz @ 48kHz Fs
	5Hz to 44kHz at 96kHz Fs
	5Hz to 88kHz at 192kHz Fs
Miscellaneous	

## Miscellaneous

Power requirements	Power rec	guirements	.100-120V, 2	200-240V.	50/60Hz (	(switched)
--------------------	-----------	------------	--------------	-----------	-----------	------------

