DaySequerra

dts Neural Technologies

Loudness Meters & Controllers
Surround Sound Processors

DaySequerra

SONIFEX

Manufacturers of Audio & Video Products For Radio & TV Broadcasters

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Introduction

Loudness Meters & Controllers

Pages 3 - 7

The DaySequerra range of loudness products use DTS Neural Loudness Measure and DTS Neural Loudness Control to provide state of the art loudness solutions for TV broadcasters. The DTS Neural Loudness Measure (NLM) uses a superior psycho-acoustic model than those used in other products, which more closely matches the tonal characteristics of the human ear and so , in turn, agrees more with your viewer's perception of loudness.

"Your listeners will appreciate the more genuine sounding control that you'll be providing by using the DaySequerra loudness products."

Surround Sound Processors

Pages 8 - 12

The DaySequerra UpMix and Mono2Stereo products transform any mono or stereo signal into a near-discrete 5.1, or 7.1 surround sound experience. 7.1 or 5.1 encoded material can be similarly DownMixed to a watermarked stereo pair for transport purposes. The MultiMerge2 combines UpMix, DownMix & Loudness Control in one product.

Also used extensively at sporting events, these products allow you to produce ultra realistic surround sound audio beds using only mono mics placed at the sports ground - see the application notes later for more information. Loudness control can also be added to them to provide real time loudness compliance.

All of these companies have used DaySequerra products:

We believe that these products are the best in the world at what they do. To try them for yourself, contact Sonifex Ltd for a demo.

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Loudness Measurement & Control For Television

Loudness has everything to do with how a human perceives it and little to do with what a reading says on a VU or PPM meter. There may be a correlation at certain frequencies but they are fundamentally different properties.

Academically, loudness is very well understood with papers and research starting in the 1920s. Historically, audio level meters measure the amplitude of the audio signal - either the RMS (root mean square) voltage of an electrical signal or the sound pressure of an acoustical signal. Neither of these measurements, although widely accepted, provides accurate indications of how viewers will perceive the loudness of the audio programming.

Loudness is a perceptual property of an audio signal when it is reproduced acoustically which is a complex non-linear function of amplitude, frequency and bandwidth. It's only recently that a concerted unified effort has been made to really understand the affects of loudness on TV viewers/listeners and to set some compliance standards, particularly ITU R-BS.1770/1, to which both manufacturers and TV stations should adhere.

There are only four, or so, recognised loudness models in existence of which ITU R-BS.1770/1 and the DTS Neural Loudness Measure (NLM) are two. The main reason why there are so few is that they are generally patented and licenced and this is expensive to do! The DTS NLM algorithm is modelled on human hearing and is probably the best model in existence, capturing the nuances of human hearing and accounting in detail for the differences between what is broadcast and what we actually hear - after all, our hearing is designed to hear other humans and not necessarily the LFE channel or 20kHz.

Why are DaySequerra Loudness Products so Much Better Than The Competitors'?

The DTS Neural Loudness Measure (NLM) uses a superior psycho-acoustic model than those used in other products, which more closely matches the tonal characteristics of the human ear (see page 4 for more information on DTS). It's also fast enough to detect subtle shifts and has a 1.5 frame look ahead to soften loudness irregularities. It's certainly much better at representing the human ear than ITU BS.1770/1. ITU BS.1770/1 is a good approximation to a loudness model but it has a number of limitations: It has problems with dialogue without background music (e.g. the evening news), problems with the LFE content and its integration time is regarded as being too slow, which is corrected by blanking the measurement where these problems occur - not really a robust solution.

So, whilst it's acceptable to use an ITU BS.1770/1 product to **measure** the loudness, to **control** the loudness you should use a better algorithm. And this is where the DaySequerra products are so much better. Because they use the DTS NLC algorithms for controlling loudness, they are being guided by a more accurate mechanism and so produce a better result. Because DTS NLC is better, the loudness control is more exact and more natural sounding. Your listeners will appreciate the more genuine sounding control that you'll be providing by using the DaySequerra loudness products.

The DTS® NLC and NLM processes are exclusively licenced to DaySequerra dedicated hardware products so these superb algorithms can only be found in these hardware boxes.

> “ Don't mess with the mix! ”
>
> **David Day**, President of DaySequerra.

Don't Mess With The Mix – Use Single Band Loudness Control

Ideally, in a loudness controller you want to be able to regulate the programme volume without the listener losing any of the experience of the original. In radio, multi-band audio processors are used to compress and constrain the sound, 'squashing' it into a narrow bandwidth and substantially altering the original sound. TV processors are different to this and should leave the sound as untouched as possible - don't mess with the mix!

It's why the DaySequerra loudness products use a single band loudness control, which is very fast acting without transitory intermodulation distortion, so that they don't alter the tone of the broadcast. Additionally, an adaptive dead band keeps the content's dynamic range largely intact. A multiband processor shouldn't be used because it is, in effect, equalising the sound and altering the content. Now that processor speeds are fast enough, a single band control is more accurate and is able to maintain the dynamic range requirements and regulatory structure, maintaining transparency and reducing the dynamic range only as necessary to maintain the target loudness.

Why is it so Important to Choose The Correct Product?

Ultimately it's about a respect for your listeners and their viewing/listening pleasure. It's easy to switch listeners off your station (and onto someone else's) with widely varying loudness levels, especially when broadcasting advertising.

Additionally, many countries either have implemented, or are about to implement, legislation to make it illegal to broadcast without carefully monitoring and controlling the loudness of programme output. The DaySequerra products offer THE BEST loudness control available, providing both listener comfort and regulatory compliance in the one box. ■

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To Provide The Best Loudness Control, You Need Three Things:

1. A very accurate perceptual loudness model. DTS NLC is fast enough to detect subtle shifts with a 1.5 frame look-ahead.

2. An automatic level control. The DaySequerra products use single band processing to maintain true transparency without artifacts or distortion. An adaptive dead-band keeps the dynamic range largely intact.

3. A compliance monitor to check that the product is controlling the loudness as it should - DaySequerra products include logging for ingest and playout via Ethernet.

Choose DaySequerra For Your Loudness Metering & Control

DTS Neural Loudness Measure & DTS Neural Loudness Control

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The proprietary DTS Neural Loudness Measure (NLM) and Neural Loudness Control (NLC) algorithms were developed by DTS after extensive research into human hearing and perceived loudness.

The DTS NLM algorithm uses a perceptual model of human hearing to more accurately detect spectral and density differences, inter-channel relationships and temporal overlaps in any audio content, resulting in a more accurate perceived loudness measurement over time.

DTS NLC technology is an advanced loudness levelling technology that accurately measures and regulates the perceived loudness differences in audio. Annoying artifacts of traditional audio processing are eliminated and loudness is controlled without squashing the quality of the audio. The resulting audio has a naturally open, dynamic quality without the disruptive side-effects of traditional energy-based volume management solutions used by other manufacturers. The result is audio with a natural, open quality and a consistent audio experience for all viewers/listeners.

DTS NLC mimics human perception of audio and provides accurate control of loudness. After loudness measurement, DTS NLC technology applies the appropriate gain or attenuation to the audio to achieve a user-defined loudness level.

The DTS NLC and NLM processes are exclusively licenced to DaySequerra dedicated hardware products so these superb algorithms can only be found in these hardware boxes.

It stands to reason that if you're using the DTS NLC to control your loudness processing, then you're going to be using the best that there is in terms of accuracy, quality and transparency and your listeners will love you for it.

DTS Neural Loudness Control Key Features Include:

- Guarantees a consistent viewer experience despite variations in consumers' equipment.
- Adheres to the ITU BS.1770/1 loudness recommendation, and more importantly agrees with your viewer's perception of loudness.
- Uses the same loudness control for stereo and 5.1 surround content.
- Manages loudness without a "squashed" or compressed sound.
- Corrects loudness while preserving the original equal loudness curve.
- Using predicting technology, makes large gain corrections imperceptible.

> "The DTS Neural Loudness Measure is by far and away the most accurate loudness model on the planet."

David Day, President of DaySequerra.

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Loudness, Metering & Measurement

iLM4ST Stereo Loudness Meter

The DaySequerra iLM4ST Intelligent Stereo Loudness Meter was designed to make it easier for radio and television broadcasters and cable networks to deliver an audio experience that won't have viewers scanning stations or reaching for the mute button.

The iLM4ST simultaneously measures the perceived loudness of four stereo channels of audio using industry standard ITU-R BS.1770/1 and the DTS Neural Loudness Measure. Key features include:

- Four AES3 stereo inputs; unbalanced on 75Ω BNC or optionally balanced I/O on 25 way D-type TASCAM breakout cable.
- Easy-to-read numerical Measured and Target readouts on vacuum fluorescent display; bar-graph LED audio level meters.
- Front panel headphone monitor.
- Rear panel includes GPIO port as well as an Ethernet interface for long-term logging, email alerts and field software updates capability. ■

Technical Specification For iLM4ST

Inputs:	4 x AES/EBU PCM inputs for four stereo channels
Outputs:	4 x AES/EBU PCM outputs for four stereo channels
Digital Inputs & Outputs:	75Ω, unbalanced BNC Balanced digital AES via DB-25 TASCAM format with cable (option) Pass-though AES outputs
Loudness Algorithms:	ITU-R BS.1770/1 Industry Standard Loudness Measurement DTS Neural Loudness Measure
Headphone Monitor:	>150mW max into 32Ω load, 3.5mm front panel TRS connector
Sample Rate:	32kHz to 96kHz
Latency:	<22 msec
Dynamic Range:	104dB, DR any input to any output; 135dB, any input to any output [1]

Ethernet:	10/100-BASE-T for remote logging and field software updates
Dimensions and Weight:	1U, 19" [482mm] W x 8" [203mm] L x 1.75" [44mm] H; 7 lb [3.2kg]
Environmental:	Convection cooled; Operating: 0 to 60 degrees C
Regulatory:	Europe: LV Directive 73/23/EEC and EMC Directive 89/336/EEC; CE Mark [EN 55022 Class A, EN55024]; RoHS and WEEE compliant North America: Designed to comply with FCC Class A, Part 15
Power Supply:	Auto-sensing 100-240V, 50-60Hz EMI suppressed male IEC320 connector
Notes:	[1] Audio measurement made using a 0dBfs 1kHz sine wave sampled at 48kHz, 20-20kHz A-weighted

iLM8 5.1 Surround Loudness Meter

Differences in audio levels between TV programmes, or between programmes and commercials, are a constant annoyance to viewers. DaySequerra's iLM8 permits broadcasters to maintain their desired loudness level across all audio programming and minimize viewer complaints.

Key Features of the iLM8

- Standalone operation – no external encoder is required.
- Eight input channels of PCM AES/EBU and optional HD-SDI inputs with channel mapping for all 16 embedded audio channels.
- Any eight channels of de-multiplexed SDI audio can be routed to the AES outputs.
- Industry-standard ITU-R BS.1770/1 and DTS Neural Loudness Measure (NLM) algorithms; simultaneous measurement for 5.1 surround and auxiliary stereo inputs.
- Easy-to-read numerical Measured and Target readouts on vacuum fluorescent display; bargraph LED audio level meters.
- Front panel headphone monitor.
- Rear panel includes an alarm port as well as an Ethernet interface for long-term logging, email alerts and field software updates capability.

DaySequerra's iLM8 Intelligent Loudness Monitor measures perceived loudness of eight channels of program audio using the industry-standard ITU-R BS.1770/1 and DTS Neural Loudness Measure (NLM) algorithms. DTS' proprietary NLM algorithm was developed by DTS after extensive research into human hearing and perceived loudness.

The iLM8 displays the results in an easy-to-read numerical format with a moving average over time, eliminating the variations with engineers interpreting traditional VU or PPM indicators. Four AES/EBU inputs and one optional HD-SDI input are provided for simultaneous measurement for 5.1 surround and auxiliary stereo inputs. With the HD/SDI option, eight channels of de-multiplexed SDI audio can be routed to the AES/EBU outputs; channel mapping is provided for all 16 SDI audio channels.

To improve system reliability and up-time, the DaySequerra iLM8 uses a robust DSP-based processing platform, rather than the PC-based approach used by most other manufacturers, to completely avoid broadcast disruptions caused by operating system lockups. The iLM8's built-in high-performance headphone amplifier allows monitoring of any two user-selected outputs even in noisy background environments. An Ethernet interface provides long-term logging and field software updates capability.

Applications for the DaySequerra iLM8 include network and program ingest centers, satellite facilities; cable head-end facilities, turnaround

Loudness, Metering & Measurement

uplinks, broadcast quality control, and post-production facilities.

Whether pre-screening content at an ingest point or keeping tabs on the output of a broadcast air-chain, the DaySequerra iLM8 is your key to reducing viewers complaints and improving audience satisfaction. ■

Technical Specification For iLM8	
Inputs	8 x AES/EBU PCM channels (5.1 plus AUX 2.0) via four BNC connectors Optional HD-SDI (SMPTE 292M/299M) on one BNC connector
Outputs	8 x AES/EBU PCM channels (5.1 plus AUX 2.0) via four BNC connectors Optional HD-SDI passive loop-through
Loudness Algorithms :	ITU-R BS.1770/1 Industry Standard Loudness Measurement DTS Neural Loudness Measure
Headphone Monitor:	> 150mW max into 32Ω load, 3.5mm front panel TRS connector
AES Sample Rate:	48kHz, 24-bit
Latency	< 4 msec
Dynamic Range:	140dB DR, any input to any output [1]
Ethernet:	10/100-BASE-T for remote logging and field software updates
Alarm Outputs:	Opto-isolated DB-9 female connector
Dimensions and Weight:	1U, 19" [482mm] W x 8" [203mm] L x 1.75" [44mm] H; 7 lb [3.2kg]
Environmental:	Convection cooled; Operating: 0 to 60 degrees C
Regulatory:	Europe: LV Directive 73/23/EEC and EMC Directive 89/336/EEC CE Mark [EN 55022 Class A, EN55024]; RoHS and WEEE compliant North America: FCC Class A, Part 15
Options:	HD/SDI I/O
Notes:	[1] Audio measurement made using a 0dBfs 1kHz sine wave sampled at 48kHz, 20-20kHz A-weighted

Loudness Control

NLC4ST Stereo Loudness Control

The DaySequerra NLC4ST provides the ability to independently process four stereo channels of audio simultaneously and apply loudness correction, all while occupying only one rack unit of space. NLC4ST is well suited for use in large facilities where space, power and heating constraints require a higher-density solution.

NLC4ST Stereo Loudness Control measures and controls perceived loudness of the program audio using the industry-standard ITU-R BS.1770/1 as well as DTS Neural Loudness Measure (NLM) and DTS Neural Loudness Control (NLC) algorithms.

NLC4ST displays the perceived loudness measurement results in an easy-to-read numerical format with a moving average over time, eliminating the historical practice of interpreting traditional VU or PPM indicators.

Four AES/EBU inputs provide for simultaneous measurement of all stereo inputs. To improve system reliability and up-time, the DaySequerra NLC4ST uses a robust DSP-based processing platform rather than the PC-based approach used by most other manufacturers to completely avoid broadcast disruptions caused by operating system lockups.

An Ethernet interface provides long-term logging and field software updates capability.

Key Features of the NLC4ST

- Guarantees a consistent viewer experience despite variations in consumer's equipment.
- Adheres to the ITU BS.1770 loudness recommendation and, more importantly, agrees with your viewers' perception of loudness.
- EBU R128, ITU R-BS.1770/1 and ATSC A/85 compliant with upgradeable firmware to allow for future upgrades.
- Lan (network) port for compliance logging.
- Manages loudness without a "squashed" or compressed sound.
- Corrects loudness while preserving the original equal loudness curve.
- Using look-ahead technology, makes large gain corrections imperceptible. ■

Applications for the NLC4ST and NLC5.1ST

- Network and program ingest centers
- Satellite facilities
- Cable head-end facilities
- Turnaround uplinks
- Quality control
- Post-production facilities

Technical Specification For NLC4ST	
Inputs:	4 x AES-3 Inputs for four channels of stereo
Outputs:	4 x AES-3 Outputs for four channels of stereo
Digital Inputs & Outputs:	AES/EBU, 75Ω, unbalanced BNC
Sample Rate:	32kHz-96kHz
Latency:	64ms @48kHz sampling rate
Alarms:	DB-9 female connector, 0-5VDC TTL
Ethernet:	10/100-BASE-T for field software updates, logging and remote control
Dimensions and Weight:	1U, 19" [482mm] W x 8" [203mm] L x 1.75" [44mm] H; 7 lb [3.2kg]
Environmental:	Convection cooled; Operating: 0 to 60 degrees C
Regulatory:	Europe: LV Directive 73/23/EEC and EMC Directive 89/336/EEC; CE Mark [EN 55022 Class A, EN55024]; RoHS and WEEE compliant North America: Designed to comply with FCC Class A, Part 15
Power Supply:	Dual redundant Auto-sensing 100-240V, 50-60Hz EMI suppressed male IEC320 connectors
Notes:	[1] Audio measurement made using a 0dBfs 1kHz sine wave sampled at 48kHz, 20-20kHz A-weighted

Loudness Control

NLC5.1ST 5.1 Surround Loudness Control

The number one complaint of television viewers is inconsistent audio loudness – "why are the commercials so loud." These differences in audio levels between TV programmes, or between programmes and commercials, are a constant annoyance to viewers. DaySequerra's second-generation loudness control, the NLC5.1ST, gives broadcasters a 1U solution to maintain their desired loudness level across all audio programming and minimize viewer complaints.

The DaySequerra NLC5.1ST measures and maintains desired audio loudness levels simultaneously for 5.1 surround and stereo programs in a single rack space unit. The NLC5.1ST measures and controls perceived loudness of the program audio using the industry-standard ITU-R BS.1770/1 as well as DTS Neural Loudness Measure (NLM) and DTS Neural Loudness Control (NLC) algorithms. The DaySequerra NLC5.1ST then uses the proprietary DTS Neural Loudness Control, an advanced loudness leveling algorithm, to apply the appropriate gain or attenuation to maintain the broadcaster-defined loudness level. The resulting audio has a naturally open, dynamic quality without the annoying side-effects of traditional energy-based volume management solutions.

The DaySequerra NLC5.1ST displays the perceived loudness measurement results in an easy-to-read numerical format with a moving average over time, eliminating the historic practice of interpreting traditional VU or PPM indicators.

Four AES/EBU inputs are provided for simultaneous measurement for 5.1 surround and auxiliary stereo inputs.

A set of user-definable alarms can alert an operator of input loss, signal clipping and high or low signal levels referenced to the desired loudness level. An Ethernet interface provides long-term logging, email alerts and field software updates capability.

When DaySequerra's NLC5.1ST is used ahead of an AC3 (Dolby Digital) transmission, the target loudness level will also match the Dialnorm metadata information, thus providing consistent, enjoyable audio for all viewers.

Key Features of the NLC5.1ST

- Simultaneous measurement and loudness control for 5.1 surround and auxiliary stereo inputs to maintain desired loudness level across all audio programming.
- Industry-standard ITU-R BS.1770/1, DTS Neural Loudness Measure (NLM) measurement and DTS Neural Loudness Control (NLC) algorithms.
- EBU R128, ITU R-BS.1770/1 and ATSC A/85 complaint with upgradeable firmware to allow for future upgrades.
- Lan (network) port for compliance logging.
- DTS NLC with proprietary look-ahead technology provides seamless, imperceptible gain transitions.
- Easy-to-read numerical LKFS and Target readouts on vacuum fluorescent display; bargraph LED audio level meters.
- Decodes and displays program metadata.
- Ethernet interface for long-term logging, email alerts and field software updates.
- Options include LTRT output and redundant power supply.

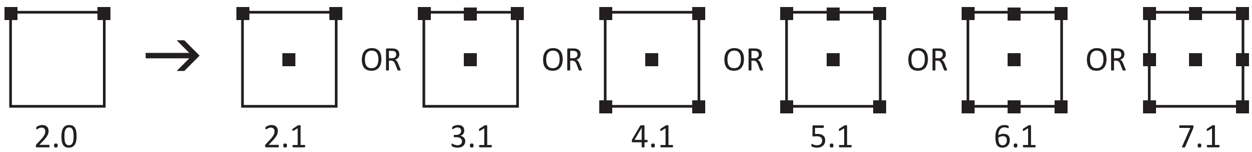
Whether pre-screening content at an ingest point or controlling the loudness of a broadcast air-chain, the DaySequerra NLC5.1ST is your key to reduce viewers complaints and improve audience satisfaction. ■

Technical Specification For NLC5.1ST	
Inputs:	3 x AES/EBU PCM inputs for 5.1 surround sound 1 x AES/EBU PCM input for auxiliary stereo
Outputs:	3 x AES/EBU PCM outputs for 5.1 surround sound 1 x AES/EBU PCM output for auxiliary stereo
Digital Inputs & Outputs:	AES/EBU, 75Ω, unbalanced BNC
Sample Rate:	32kHZ, 96kHz
Latency:	< 16 msec
Dynamic Range:	140dB DNR/ 135dB THD+N, any input to any output [1]
GPIO:	Opto-isolated DB-9 female connector; 0-5VDC TTL
Ethernet:	10/100BASE-T for software updates, logging and remote control
Dimensions and Weight:	1U, 19" [482mm] W x 8" [203mm] L x 1.75" [44mm] H; 7 lb [3.2kg]
Environmental:	Convection cooled; Operating: 0 to 60 degrees C
Regulatory:	Europe: LV Directive 73/23/EEC and EMC Directive 89/336/EEC; CE Mark [EN 55022 Class A, EN55024]; RoHS and WEEE compliant North America: Designed to comply with FCC Class A, Part 15
Power Supply:	Dual redundant Auto-sensing 100-240V, 50-60Hz EMI suppressed male IEC320 connectors
Notes:	[1] Audio measurement made using a 0dBfs 1kHz sine wave sampled at 48kHz, 20-20kHz A-weighted

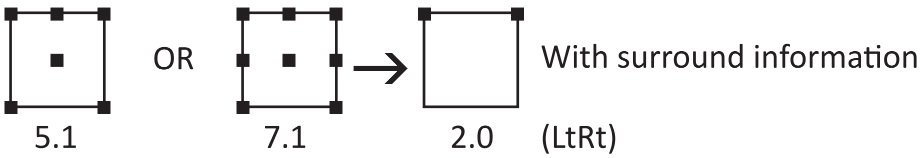
Surround Sound Processors

The DaySequerra surround sound processors are a range of UpMix, DownMix, MultiMerge and Mono2Stereo products which use DTS Neural Surround™ processing to provide a near-discrete 5.1, or 7.1 surround sound experience.

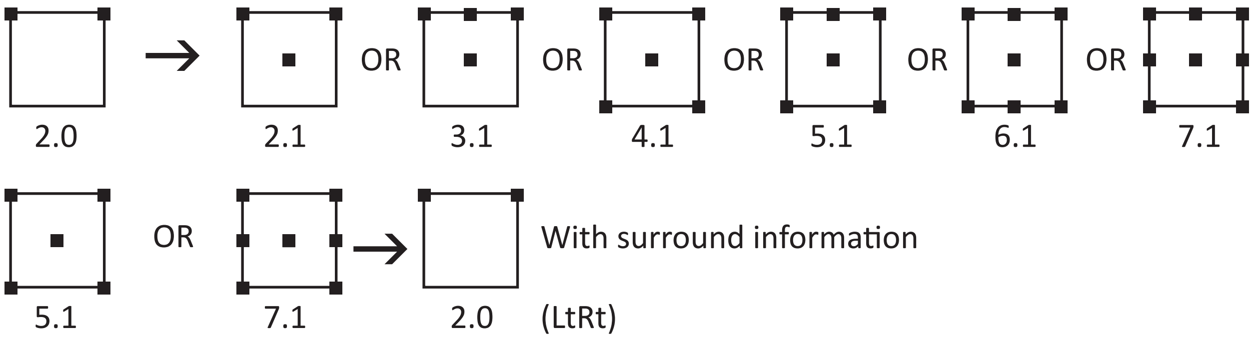
UpMix - Converts stereo to any channel configuration (2.1, 3.1, 4.1, 5.1, 6.1 or 7.1)



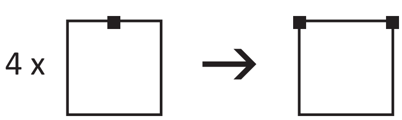
DownMix - Converts 5.1. or 7.1 to stereo with surround steering information (LtRt)



MultiMerge2 - Does the UpMix and DownMix processes together with Loudness Control



Mono2Stereo - Converts a mono signal to a stereo signal x 4



“ DTS Neural Surround UpMix technology was ranked #1 in double blind listening tests, outperforming all other methods in near discreteness, envelopment, and original spatial location. ”

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UpMix - Surround 5.1 & 7.1 Decoder

UpMix Surround Audio Decoder

The DTS UpMix transforms any stereo signal into a surround sound experience. Audio that was encoded using the DTS DownMix algorithm can be restored with the correct information in each of the 5.1 surround channels.

Stereo content that has not been encoded can be translated into width and depth information, creating a pleasing and enveloping surround image - often indistinguishable when compared to the original surround sound mix content.

Matrix-encoded LtRt stereo signals can be upmixed by DTS Neural Surround and provide a high quality surround sound experience to a surround sound image that is more stable than that of traditional matrix decoders.

DTS Neural Surround UpMix spectrally separates individual audio elements and places each in its intended location within the surround environment. This results in unparalleled image placement and stability. The audio is placed exactly where it would be heard in a professional Live End Dead End (LEDE) mixing studio listening environment. The UpMix does this while preserving the artistic integrity of the content.

Key Features of the UpMix:

- Stereo to any channel configuration (2.1, 3.1, 4.1, 5.1, 6.1 or 7.1)
- Great compatibility with all stereo signals (L/R, LtRt, LoRo, ProLogic, ProLogic II or SRS Circle Surround)
- Adaptive filters that eliminate “dialog leakage” traditionally found in matrix audio decoders
- End-to-end surround delivery system when combined with the DTS Neural Surround™ DownMix
- Using UpMix, you can decode Dolby LtRt & SRS LtRt as well as DTS Neural LtRt to 5.1 surroud sound - they are all compatible.
- UpMix is the only 7.1 hardware encoder/ UpMix available (at time of writing). ■

Technical Specification For UpMix	
Inputs:	3 x AES-3 inputs (2.0 and 5.1) 1 x AES-3 or AES-11 SYNC
Outputs:	4 x AES-3 outputs (5.1 and 7.1) 1 x AES-3 or AES-11 SYNC pass-through
Digital Inputs & Outputs:	AES/EBU, 75Ω, unbalanced BNC
Sample Rate:	32kHz to 96kHz
Latency:	<22 msec
Dynamic Range:	104dB, DR any input to any output; 135dB, any input to any output [1]
GPIO:	DB-9 female connector 0-5V TTL
Ethernet:	10/100-BASE-T for remote logging and field software updates
Dimensions Weight:	1U, 19” [482mm] W x 8” [203mm] L x and 1.75” [44mm] H; 7 lb [3.2kg]
Environmental:	Convection cooled; Operating: 0 to 60 degrees C
Regulatory:	Europe: LV Directive 73/23/EEC and EMC Directive 89/336/EEC; CE Mark [EN 55022 Class A, EN55024]; RoHS and WEEE compliant North America: Designed to comply with FCC Class A, Part 15
Power Supply:	Dual redundant Auto-sensing 100-240V, 50-60Hz EMI suppressed male IEC320 connector
Notes:	[1] Audio measurement made using a 0dBfs 1kHz sine wave sampled at 48kHz, 20-20kHz A-weighted

Applications for UpMix

- UpMix is used on OB vans that create 5.1 content, by upmixing archived material and commercials that need to be inserted by the van at a sports event, e.g. replays.
- UpMix is used extensively at sports events. For example, the World Cup in 2010 was transmitted in 41 different languages in DTS Neural Surround Sound using DaySequerra UpMix.
- DaySequerra UpMix was used to add tracks to the Rocky Horror Picture Show DVD/Blu-Ray release, by taking a 3.1 mix and converting it to a 7.1 surround project.
- UpMix is used at Lucas Films and the Skywalker Ranch for their high latency work to get the very best audio mix possible.
- In the USA, over 750 stations broadcast 5.1 surround sound in HD Radio using UpMix to create the content from a stereo master in real time.

UpMix HD Higher Definition Surround Audio Decoder

The new UpMix HD is a high latency version of the UpMix which provides a higher quality near-discrete, artifact-free 5.1 performance from stereo (latency 38ms).

An Auto-UpMix feature provides full-time 5.1 output from a mix of stereo and 5.1 content input. Additionally, the unit can be upgraded to accept an HD-SDI input and use the de-embedded audio channels from SDI groups 1 or 2.

UpMix NLC Higher Definition Surround Audio Decoder With Loudness Control

Combining the features of the popular UpMix HD and DTS Neural Loudness Control products, UpMix NLC will upmix stereo or encoded stereo content to near-discrete 5.1 surround sound using the latest DTS Neural Surround™ UpMix HD process, and maintain the loudness target and ensure CALM compliance using industry-standard ITU-R BS.1770/1 loudness measurement and the proprietary DTS Neural Loudness Control.

DownMix - Surround 5.1 & 7.1 Encoder

DownMix Surround Audio Encoder

The DTS Neural Surround™ DownMix encodes 5.1 surround sound to a stereo mix that accurately represents the 5.1 original. Unlike standard summing methods, the DTS Neural Surround™ DownMix creates stereo that contains surround steering information. This stereo audio can be stored, mixed, transported and monitored just like any other (Lt/Rt or Lo/Ro) stereo mix.

Key Features of the DownMix:

- Perfect mapping of 5.1 or 7.1 surround elements into the stereo field
- Produces a DTS Neural LtRt from a 5.1 or 7.1 source.
- A higher latency algorithm allows more processing time which produces a higher quality DTS Neural Lt/Rt output.
- Active correction to fix issues such as comb filtering, spatial location and distortion.
- The conveniences of a matrix surround encode/decode system with the performance of modern day perceptual audio codecs.
- Backward compatibility with matrix decoders (Dolby® ProLogic II, SRS Circle Surround, etc.).
- End-to-end surround delivery system when combined with the DTS Neural Surround™UpMix ■

Technical Specification For DownMix

Inputs:	4 x AES-3 inputs (5.1 and 7.1) 1 x AES-3 or AES-11 SYNC
Outputs:	3 x AES-3 outputs (2.0 and 5.1) 1 x AES-3 or AES-11 SYNC pass-through
Digital Inputs & Outputs:	AES/EBU, 75Ω, unbalanced BNC
Sample Rate:	32kHz to 96kHz
Latency:	<16 msec
Dynamic Range:	140dB, DR any input to any output; 135dB, any input to any output [1]
GPIO:	DB-9 female connector 0-5V TTL
Ethernet:	10/100-BASE-T for remote logging and field software updates
Dimensions and Weight:	1U, 19” [482mm] W x 8” [203mm] L x 1.75” [44mm] H; 7 lb [3.2kg]
Environmental:	Convection cooled; Operating: 0 to 60 degrees C
Regulatory:	Europe: LV Directive 73/23/EEC and EMC Directive 89/336/EEC; CE Mark [EN 55022 Class A, EN55024]; RoHS and WEEE compliant North America: Designed to comply with FCC Class A, Part 15
Power Supply:	Dual redundant Auto-sensing 100-240V, 50-60Hz EMI suppressed male IEC320 connector
Notes:	[1] Audio measurement made using a 0dBfs 1kHz sine wave sampled at 48kHz, 20-20kHz A-weighted

Applications for DownMix

- DownMix is used to create a stereo transport stream for 5.1 mixes as a stereo LtRt pair, in combination with UpMix.
- DownMix is used to archive 5.1 content as LtRt for later broadcast, reducing storage requirements.
- DownMix is used to transmit stereo radio from 5.1 mixes.

MultiMerge2 - Surround 5.1 & 7.1 Encoder & Decoder

MultiMerge2 5.1 Surround Audio + 2.0 Processor With Loudness Control

The second-generation MultiMerge2 is a complete surround and loudness solution for your transmission facility.

Combining the features of the popular DTS Neural Surround™ UpMix with an auxiliary second stereo channel, MultiMerge2 will automatically upmix stereo or encoded stereo content to near-discrete 5.1 surround sound using the latest DTS Neural Surround™ UpMix HD process; on-board DTS Neural Loudness Control maintains the loudness target for both 5.1 main and auxiliary stereo outputs and ensures CALM compliance using industry-standard ITU-R BS.1770/1 loudness measurement and the proprietary DTS Neural Loudness Control – all in 1U.

Key Features of the MultiMerge2

- Simultaneous processing for 5.1 main and 2.0 auxiliary stereo inputs.
- UpMix HD algorithm for near-discrete 5.1 performance from stereo .
- Auto-UpMix provides full-time 5.1 output from mix of stereo and 5.1 content input.
- Mute-less 5.1 pass-through to guarantee critical audio is never lost, such as center channel dialog.
- DTS Neural Loudness Control and ITU-R BS.1770/1 Loudness Measurement - ensures CALM compliance.

MultiMerge2 - Surround 5.1 & 7.1 Encoder & Decoder

- EAS/CAP input for transmission of Emergency Alert messages.
- Dual redundant power supply provides operational backup in the event of single mains supply failure.
- Ethernet interface for loudness-compliance logging and field updates.
- Optional HD/SDI input for 8 de-embedded audio channels from SDI group 1 or 2.

MultiMerge2 accepts 5.1-channel network audio and two-channel local audio or SAP/DVS along with EAS/CAP audio and automatically upmixes only when necessary, enabling broadcasters to deliver compelling 5.1-channel surround sound while saving time, money and space. The new UpMix HD algorithm outperforms all existing upmixing methods for channel discreteness, envelopment, and original spatial location regardless of source (Lt/Rt, Lo/Ro, ProLogic, ProLogic II or SRS Circle Surround).

The breakthrough DTS Neural Loudness Control MultiMerge2 maintains the broadcaster-defined loudness level without the annoying side-effects of traditional energy-based volume management solutions; industry-standard ITU-RBS.1770/1 loudness measurement maintains broadcaster-defined loudness level across all audio programming to minimize viewer complaints while ensuring CALM compliance.

A set of user-definable alarms can alert an operator of input loss and hardware faults. An Ethernet interface provides PC loudness compliance logging and field software updates.

Other options include HD/SD-SDI input for 8 de-embedded audio channels from SDI group 1 or 2 making MultiMerge2 the mostcost-effective, total content delivery solution. ■

Technical Specification For MultiMerge2

Inputs:	3 x AES-3 PCM inputs for 5.1 surround main program 1 x AES-3 PCM input for 2.0 AUX Stereo 1 x AES-3 PCM input for EAS/CAP 1 x AES-3/AES-11 input for external AES sync 1 x HD/SDI Input (optional)
Outputs:	1 x AES-3 PCM output for main program Lt/Rt down mix 1 x HD/SDI Pass-through output with option
Audio Input Output Interface:	AES/EBU, 75Ω, unbalanced BNC and HD-SDI input for SMPTE 259M, SMPTE 292M, SMPTE 424M, ITU-R BT.656 and ITR-R BT.601 input with option
Loudness Measurement and Correction:	ITU-R BS.1770/1 Industry Standard Loudness Measurement DTS Neural Loudness Measure, Leq(A), Leq(B), Leq(C) and Leq(M), DTS Neural Loudness Control
Sample Rate:	32kHz to 96kHz, 24-bit
Latency[1]:	0ms Lt/Rt DownMix 48ms DTS Neural Loudness Control <6ms for Hardware Interface
GPIO:	Opto-Isolated DB-9 female connector, 0-5VDC TTL
Ethernet:	10/100-BASE-T for field software updates, logging and remote control
Dimensions and Weight:	1U, 19” [482mm] W x 8” [203mm] L x 1.75” [44mm] H; 7 lb [3.2kg]
Environmental:	Convection cooled; Operating: 0 to 60 degrees C
Regulatory:	Europe: LV Directive 73/23/EEC and EMC Directive 89/336/EEC; CE Mark [EN 55022 Class A, EN55024] RoHS and WEEE compliant North America: Designed to comply with FCC Class A, Part 15
Power Supply:	Dual redundant Auto-sensing 100-240V, 50-60Hz EMI suppressed male IEC320 connector
Options:	HD/SDI input for 8 de-embedded audio channels from SDI group 1 or 2
Warranty:	One year, limited parts and labor
Notes:	[1] Does not included added latency of any options

Mono2Stereo - Mono To Stereo Processor

Mono2Stereo 4 Channel Stereo Synthesizer Processor

Why is the Mono2Stereo so Important For Sports Broadcasts?

Using only one Mono2Stereo and a mixer, you can create an entire in-depth surround sound experience for your viewers and listeners using only a couple of mono shotgun mics. See the application note overleaf for more information.

Dealing with mono content in today’s live sports surround sound environment presents a number of problems for engineers who are working to deliver the highest quality mixes. Simple mono sound sources just don’t always cut it.

DaySequerra’s four-channel synthesizer uses a DTS proprietary algorithm developed to produce an artifact-free, wide soundstage with natural, spectrally balanced stereo from mono content.

Applications for the Mono2Stereo

This is used extensively at sports events for making surround beds from mono mic sources. At the Winter Olympics 2010 in Vancouver, for NASCAR on SPEED and Major League Baseball to name a few, Mono2Stereo units were used to create the surround bed and there are weekly TV broadcasts by ABC, HBO and Turner totalling over 3,600 sporting events broadcast per annum which use the DaySequerra Mono2Stereo hardware.

DaySequerra

dts Neural Technologies

Mono2Stereo - Mono To Stereo Processor

Key Features of the Mono2Stereo™

- Four stereo output channels; each channel has both balanced analogue and 75Ω unbalanced AES3 digital audio outputs.
- User presets simplify setup and allow consistent, easy recall of favourite settings.
- Fast responding limiter maintains desired output threshold level.
- Built-in low shelf and high shelf filters to help compensate for microphone frequency response and tailor overall mix balance.
- Full-time 24 bit Analogue-to-Digital, Digital-to-Analogue and Sample Rate Conversion

Technical Specification For Mono2Stereo

Inputs:	4 x mono AES PCM channels via four 75Ω BNC connectors 4 x mono analogue +24dBu max balanced channels via XLR D-Sub breakout cable AES Sync input
Outputs:	4 x stereo AES/EBU PCM channels via four 75Ω BNC connectors 4 x mono analog +24dBu max balanced channels via XLR D-Sub breakout cable passing through for AES SYNC
Analog to Digital Performance:	24 bit delta/sigma ADC converter (4 channels) 32kHz to 96kHz, 24 bit
Dynamic Range /THD+N:	140dB DR/135dB THD+N Digital Input/Output [1] 105dB SNE/0.0015% THD+N [2]
Ethernet:	10/100-BASE-T for remote logging and field software updates
Dimensions and Weight:	1U, 19" [482mm] W x 8" [203mm] L x 1.75" [44mm] H; 7 lb [3.2kg]
Environmental:	Convection cooled; Operating: 0 to 60 degrees C
Regulatory:	Europe: LV Directive 73/23/EEC and EMC Directive 89/336/EEC; CE Mark [EN 55022 Class A, EN55024]; RoHS and WEEE compliant North America: Designed to comply with FCC Class A, Part 15
Power Supply:	Dual redundant Auto-sensing 100-240V, 50-60Hz EMI suppressed male IEC320 connectors
Notes:	[1] Audio measurement made using a 0dBfs 1kHz sine wave and A weighted [2] THD+N measured using a 20dB 1kHz sine wave sampled at 48kHz, 20-20kHz A-weighted [3] DB-25 to XLR analog breakout cables not included

Application: Creating a Surround Sound Mix at a Sporting Event Using The Mono2Stereo

This is how the World Cup, French Open and British Open events were shot using a DaySequerra Mono2Stereo (M2S):
In a nutshell, you take one shotgun mic and point it at the crowd in the stands and use M2S input ONE to spread the mono gently into the Ls/Rs channels, making an instant rear surround bed. You then take a second shotgun and point it at the field, taking the input into M2S input TWO and spread it a bit more than the surrounds and there is your instant front L & R surround bed. The Centre position created by the M2S is where you cleanly drop in your announcers, or you can also feed the announcer through input THREE and give a little spread to L & R to make the them sound more like they're on the field of play. Getting even more creative, you can use the GPI to toggle between two presets in channels ONE and TWO when the on-field camera perspective changes, e.g. a shot from the other end of the field. You can also drop parabola feeds from the field e.g. kicks, hits, etc, into L and R after the M2S, right into your mixer for more realistic game sounds. Remember that the M2S has four inputs, to create four separate stereo mixes, so one box can create the whole mix.

Low Bit Rate DAB+ Processors

Eclipse LBR4

Low Bit-rate Digital Radio Loudness Processor

DaySequerra's proprietary Eclipse codec pre-processing engine has been specially tuned for operation at 24 kbps, 32 kbps, 48 kbps, 64 kbps & 96 kbps bit rates to significantly reduce artifacts from lossy codecs & low-bit rate transmission and to improve the audio performance of low bit rate HD Radio multicasts, DAB+ & DRM channels.

Eclipse M2DTV

Mobile Media DTV Loudness Processor

The M2DTV has 5.1 surround, stereo & EAS/CAP inputs & features DTS® Neural Surround DownMix along with two channels of the Eclipse codec pre-processing engine.

Radio Tuner/Monitor

M2A-FM

Off-Air Analogue FM Modulation Monitor

The M2A-FM gives you audiophile grade Class-A biased analogue & AES/EBU audio outputs, a sensitive FM preselector with 20 presets, built-in attenuator, RBDS decode and opto-isolated alarm outputs for audio peak, audio program, carrier loss and RBDS.

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