Audio Processing Techniques With the Texas Instruments TAS3103 3-Channel Digital Audio Processor

Introduction

This document is intended to illustrate some of the many applications and processing techniques possible with the TAS3103 digital audio processor. It begins with a discussion of the basic audio processing structure, which is a combination of hardware and ROM-based firmware in the chip. As will be apparent, the TAS3103 is an extremely versatile audio processing IC.

Multiple TI patents and patent applications cover the information presented in this paper.

The TAS3103, shown in block diagram form below, is a configurable digital audio processor (DAP) that allows great flexibility while allowing very quick development of applications. The part is generally organized as a 3-channel processor, although some options allow more than 3 inputs to be processed in downmix algorithms. The part is also designed to facilitate 2-chip emulations of 6-channel DAPs with no additional glue logic.

The TAS3103 is controlled via an I2C data bus using a table of subaddresses that allow both writing and reading back any of the control parameters used.

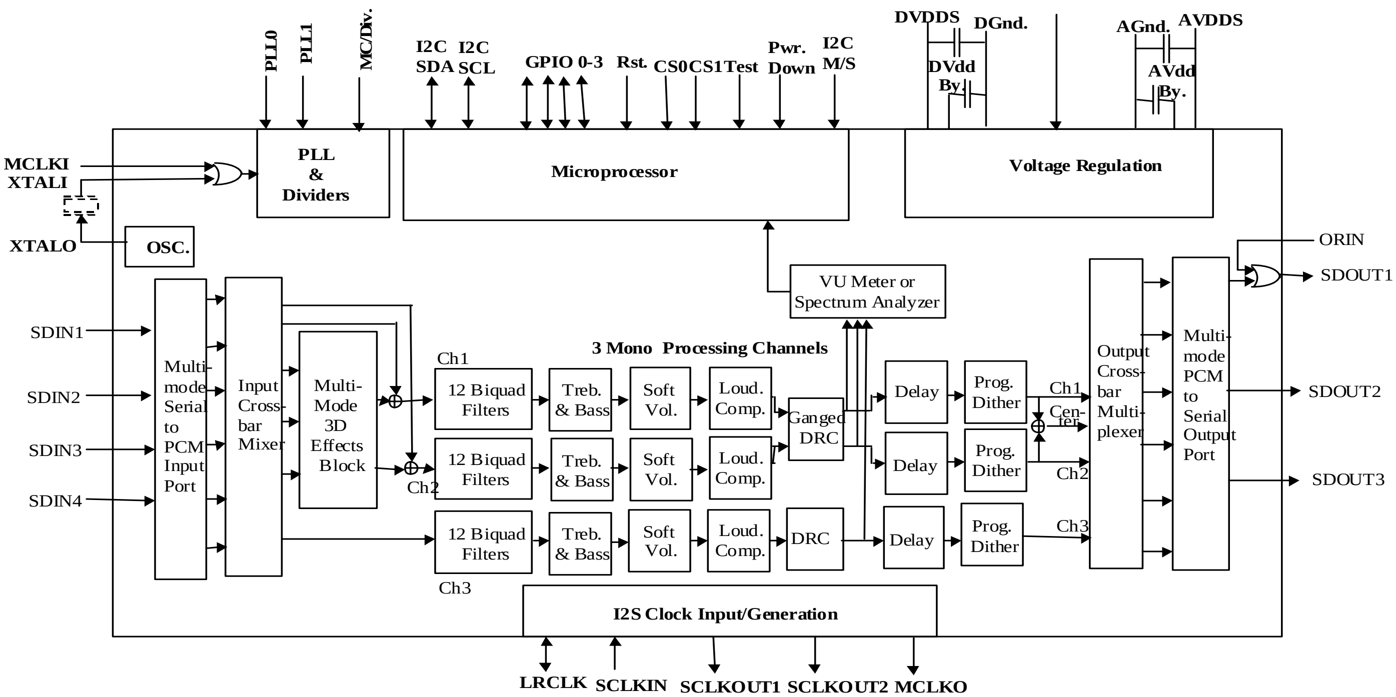

Figure 1: TAS3103 Block Diagram

It should be noted at the outset that this document is being written as an early survey of the intended features of the TAS3103. While the features and applications described below were included in the design of theTAS3103, they have not been laboratory tested at the time of writing. For that reason, this document should be viewed as an early survey of proposed features, rather than a typical application note. Our intent is to convert this document into a series of application notes as soon as the laboratory testing is completed.

The TAS3103 accepts data in various serial data formats including left/right justified and I2S, 16, 20, 24, and 32 bits, discrete or time-division multiplexed (TDM). It allows sample rates from 8 through 96 KHz. It is controlled by I2C commands with subaddresses for each control function. Internal precision is 48 bits parallel (single precision) and coefficients have 28 bit precision (signed 5.23 format). Audio data is input with 8 overhead bits and 8-24 noise bits. (Lower precision inputs have more noise bits in internal computations). The accumulator precision is 76 bits, with truncation to 48 bits after most operations. (Inside each biquad filter, however, the precision remains at 76-bits with truncation to 48 bits at the output of the filter.) The inherently high audio precision allows aggressive processing of audio data, including professional audio, with few concerns about induced artifacts.

The evaluation module can be controlled with a PC-based DAP configuration tool that allows complete configuration of the TAS3103 via a graphic user interface (GUI). Using the evaluation module and DAP configuration tool, application configurations for the TAS3103 can be developed in a few hours.

The block diagram begins with a programmable input serial audio port. Next, a crossbar mixer allows any of the 8input channels to be mixed into 6 nodes, including the 3 monaural processing channels that follow. An effects block facilitates many self-developed and 3rd party 3D algorithms. A biquad filter block implements graphic equalization presets, parametric speaker equalization, phase compensation, and embedding of 3rd party algorithms. A built-in set of treble and bass shelf filters allows soft control of treble and bass over a wide range of sample frequencies with very little I2C control traffic. The soft volume block allows each channel's volume to be controlled without audio artifacts with very little I2C traffic. A loudness compensation block allows compensation for loss of bass at low volumes. A dynamic range control block allows very flexible control of dynamic range with both compression and expansion implemented to allow both boosts and cuts in any of 3 regions. Delay blocks in each channel allow imaging readjustments, hall effects, and other delay functions. Finally, programmable dither and truncation of audio words allows full control over audio output precision.

An additional feature shown in the block diagram is a 10-band spectrum analyzer that outputs band information in log format to registers that can be polled via the I2C bus. (The traffic for 10 bands of spectrum analyzer information at 20 Hz update rate is about 2% of the throughput of a 100 KBit/s implementation of the I2C bus and about ½% of the throughput at 400 KBits/s.) The spectrum analyzer can be a single 10-band implementation with multiple sources and mixes as well as a stereo 5-band implementation. It can also be implemented as a stereo VU meter in log format.

Serial Audio I/O

The input and output serial audio ports of the TAS3103 are very versatile. There are 16 selectable input formats and 16 selectable output formats.

Word Formats

Both input and output ports can handle data word sizes of 16, 20, 24, and 32 bits. Left-justified, right-justified, and I2S formats are supported. Both discrete I/O and time-division-multiplexed (TDM) formatting are supported. In the latter formats, both left-justified and I2S formats are supported with up to 6 channels per input or output channel.

Sample Frequency and Timing

Inputs and outputs support sample frequencies, (e.g. LRCLK frequencies) from 8 KHz to 96 KHz. The TAS3103 can act as a slave, accepting MCLK, LRCLK, and SCLK from an outside source. It can also act as a master, providing LRCLK, SCLK, and MCLK. The master clock can be input from an external source or can be developed with an in-chip crystal oscillator and a crystal.

Format Conversions

The input and output serial audio ports are virtually independent. For example, the word size can be changed from input to output, as can the left/right/I2S justification. The TAS3103 can convert from TDM transfer format to discrete format and from discrete to TDM. The only restriction of this capability is that the input and output SCLKs must be related by a binary ratio. The TAS3103 generates two SCLKs, (one for the input and one for the output), to allow this conversion. The LRCLK must be the same for input and output transfers.

An example of the usefulness of this feature that it allows the TAS3103 to follow a decoder DSP that has only one serial output while providing discrete outputs to DACs that do not have TDM format capability. It can also accept discrete data from ADCs that have only discrete outputs and provide TDM data to a multi-channel DAC that has that capability, reducing board wiring.

Input Crossbar Mixer

The input crossbar mixer provides a mixing function between 8 input channels to the 3 monaural processing channels. It also facilitates a full 6-channel to 6-channel mix in 2-chip implementations of 6-channel applications. It can also provide a 3:1 multiplex of input stereo channels. The mix uses a full algebraic multiply with a 5.23 signed format that allows normal phase or inverted phase gains of up to approximately 16X (24 dB). The fractional part of the coefficient allows very high quality internal algorithms, especially in biquad filter elements.

The crossbar mixer also participates in the effects block described below, providing both precision scaling of algorithm inputs as well as phase inversions where needed.

3D Effects

The effects block, coupled with the crossbar mixer, provides a configurable implementation of a number of 3D algorithms, including stereo enhancement, 6-channel to 2-channel virtual surround synthesis, and other 3D algorithms. The block is designed to implement head-related-transfer functions in the 4-biquad filter blocks and inter-aural delays in the reverb/delay blocks. The outputs from these blocks can be mixed directly or with phase change into channels 1 and 2. Both first-order and second-order filters can be configured. This block can also be used for any algorithm requiring parallel summation of filter blocks.

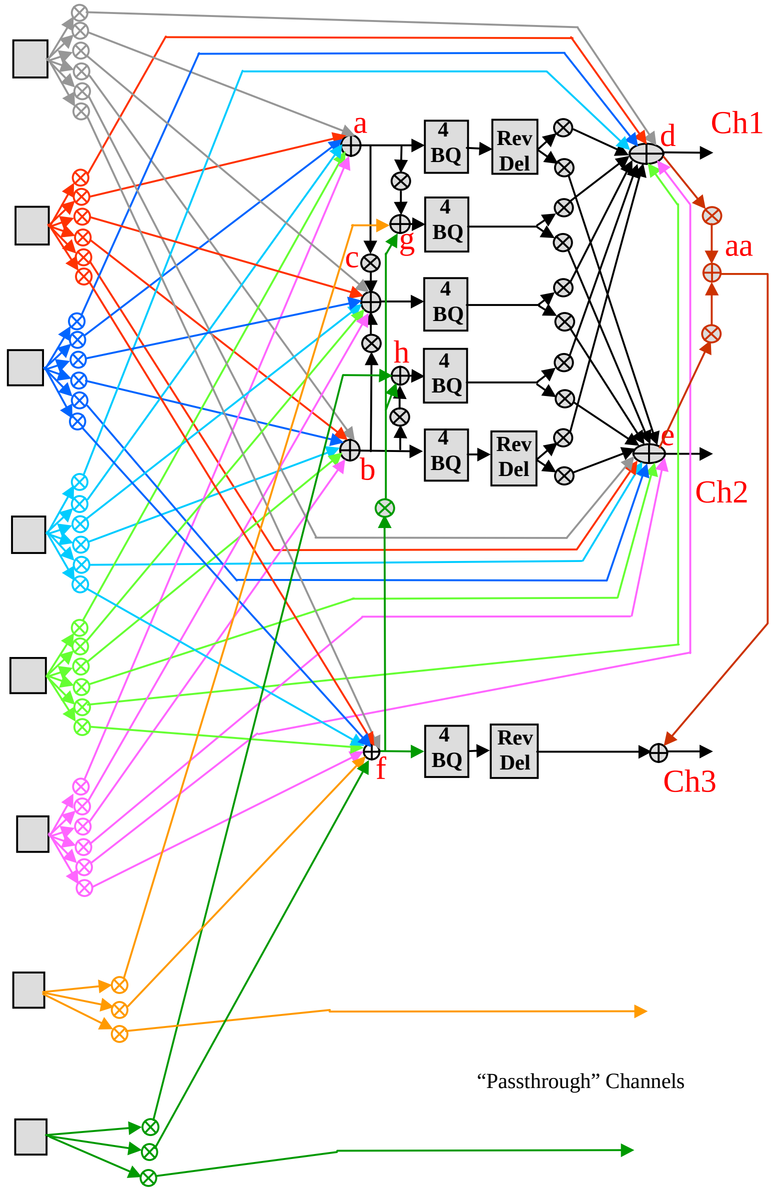

Figure 2: Effects Block (Ch1 & Ch2)

Each of the input pairs, (A and B, C and D, E and F, G and H), arrive through the 4 serial audio inputs and are routed to the 8 PCM nodes shown. 5 of the input mixes allow mixing or multiplexing of each of the input signals to each of the summation points shown. This structure shown in Ch1 and Ch2 is an abstracted overlay that allows the configuration of many 3D algorithms.

An extra mix is provided for each of the inputs, allowing a mix into the 3rd channel, which will be discussed below. The Ch1 + Ch2 summation into Ch3 is used in some 3D algorithms. Inputs G and H participate in some 3D algorithms and can also be "passed through" the TAS5412 to control multiplexing of channels in addition to the 3 processed channels.

Head-Related Transfer Function (HRTF) Filters

Each of the 4-biquad blocks can implement 4 second-order infinite-impulse response (IIR) filters of any type. Typically each of the blocks will be used to implement the HRTF for one of the data inputs, establishing the apparent direction of that sound. The PC-based DAP configuration tool can generate the filter coefficients to fit a desired curve.

Interaural Delay Generation

The reverb/delays in the effects block can be used to generate the inter-aural delay (e.g., the delay from one ear to the other as a sound propagates by from one side, consistent with the HRTF). That delay causes the brain to interpret a sound's direction correctly for a component of the input. This allows the implementation of 3D algorithms that use delays.

3rd Party 3D Algorithms

We are working with 3D algorithm developers to imbed their algorithms in the effects block. As soon as arrangements are complete, relevant applications notes will be published.

L&R Channel Processing Techniques

The processing associated with the main left and right channels is shown in Figure 3 below. The Ch1 & Ch2 inputs shown are the inputs from the effects block shown above. Function and features of each of the processing blocks will be discussed below in a left-to-right signal flow.

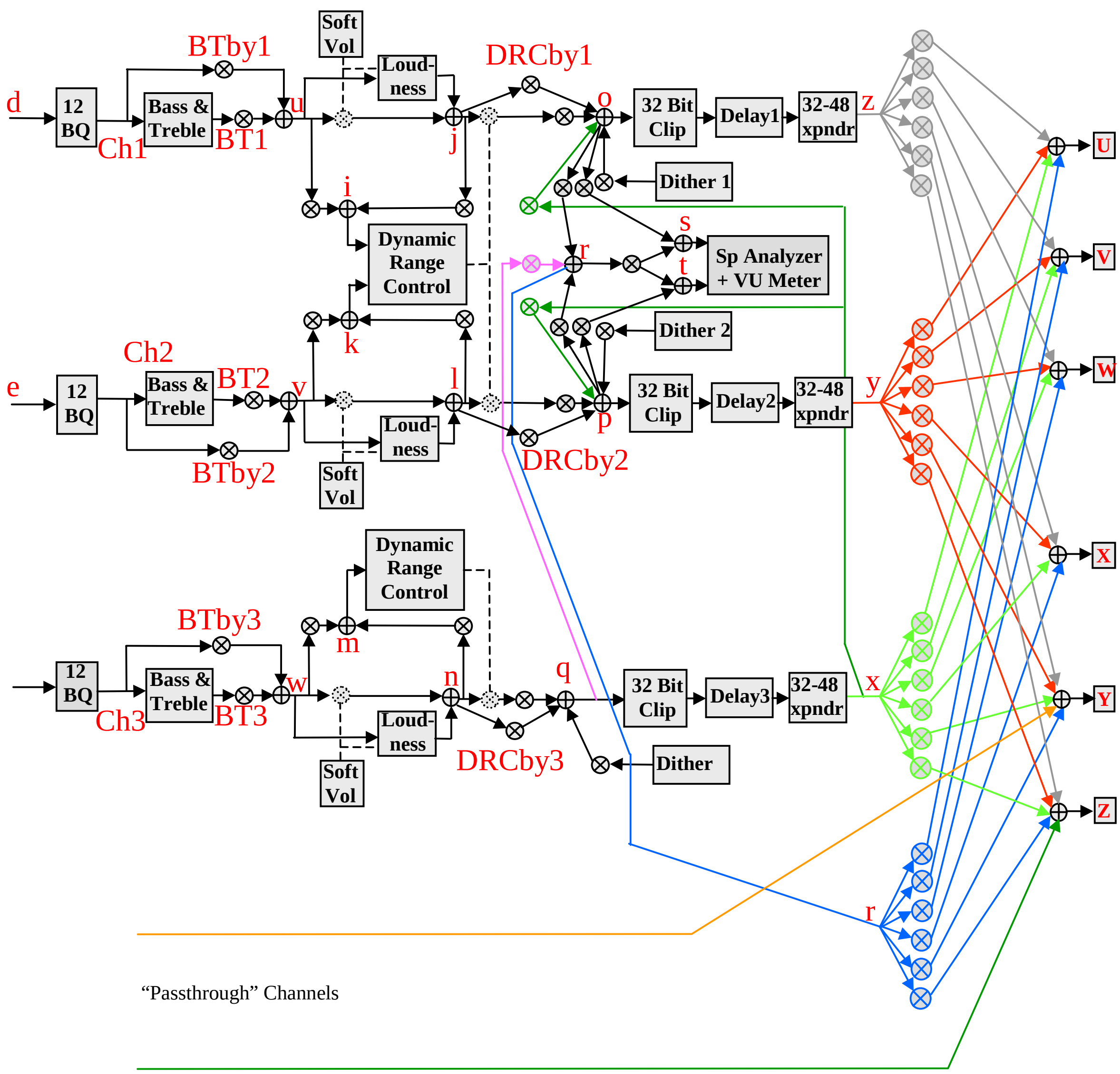

Figure 3: Main L and R Processing (Ch1 & Ch2)

Filter Block

The filter blocks consist of 12 second-order IIR (biquad) filters. They are cascaded to allow sophisticated filtering operations. Each filter can be of any type, and up to 24th order filters can be factored and implemented in the block. In addition, 1 or 2 first-order filters can also be implemented by each biquad.

The accumulation in each biquad remains in the 76-bit accumulator during computation and is truncated to 48 bits between biquads, providing very high quality filter implementation. The 8-bit overhead (48 dB) of the data between biquads allows considerable filter flexibility with minimum concerns about clipping in the filter block. Passband gain control in following filters and pre- and post-filter scaling points can be used to maintain the desired 0 dB point. Some uses of this block are discussed below.

Parametric Speaker Equalization

While high fidelity speakers are generally fairly flat over their usable band, most still have some variation in their output level curve. Lower-quality speakers and crossover networks can have even more variation over the audio band.

Parametric speaker equalization uses a set of filters with different center frequencies and bandwidths to compensate for speaker variability over frequency or to allow the equipment manufacturer to create a desired response curve. This provides the capability to “voice” speakers by adjusting the audio spectrum to any desired shape. Normally, bell shaped filters with either boost or cut function are used to pre-condition the audio so that the combination of the filters and speaker set has the specified output curve. This technique, called parametric speaker equalization can provide dramatic improvements in apparent speaker quality in cost-driven applications. This technique is especially powerful when used with small satellite speakers in 5.1 AV Receiver, DVD Receiver, and HTIB applications. It is also useful in mini/micro component automotive, and TV applications.

Graphic EQ Presets

If the user’s system microprocessor stores several sets of coefficients for the filter blocks, very sophisticated graphic EQ presets can be implemented. This allows precise tailoring of the channel response for different kinds of instrumental music, for vocals and dialog, and for movie sound reproduction. This allows considerably more flexibility than typical 3-band graphic EQ presets.

In an automated approach, an A/D converted microphone can be analyzed by the 10-band spectrum analyzer output, controlling a 10-band graphic EQ block with the same filter characteristics. With this matched set of filters, the algorithm for using a white noise source to get a flat output from a given speaker/surroundings environment is relatively simple. This approach can be used with a low-cost microphone to implement an in-home or in-car speaker equalization approach.

Bass & Treble

The TAS3103 provides a high quality treble and bass adjustment with “analog feel”. Control of the treble and bass is as simple as selecting a filter set based on the sample rate and then selecting one of about 150 shelf filter curves by transmitting an index value.

Soft B&T Control

Changes in treble and bass level involve transmitting the desired final index value, which selects the actual treble or bass shelf filter curve. The TAS3103 transitions smoothly, moving through each of the intervening filter curves at a controlled rate. This provides an artifact-free, smooth transition, no matter how large the transition commanded.

Built-In Filters Match Sample Rate

The figure below shows the relationship between the sample frequency and the treble/bass corner frequency of each of the 5 sets of shelf filters. The shaded cells indicate the “normal” selection for each sample frequency, although any of the filters can be chosen if different corner frequencies are desired. Each filter set is actually about 150 different filters, with shelf boosts up to +18 dB and cuts to – 18 dB. The inclusion of these filters into the TAS3103 greatly simplifies the system implementation of high-quality, "analog feel" treble and bass control.

BASS FILTER SET 3db CORNERS

Fs (LRCLK)		Set2b	Set3b	Set4b	Set5b	Set6b
96		250	500	750	1000	1500
88.4		230	460	691	921	1381
64		167	333	500	667	1000
48		125	250	375	500	750
44.1		115	230	345	459	689
32		83	167	250	333	500
24		63	125	188	250	375
22.05		57	115	172	230	345
16		42	83	125	167	250
12		31	63	94	125	188
11.025		29	57	86	115	172

= Typical Selection

TREBLE FILTER SET 3db CORNERS

Fs (LRCLK)		Set2t	Set3t	Set4t	Set5t	Set6t
96		6000	12000	18000	24000	36000
88.4		5525	11050	16575	22100	33150
64		4000	8000	12000	16000	24000
48		3000	6000	9000	12000	18000
44.1		2756	5513	8269	11025	16538
32		2000	4000	6000	8000	12000
24		1500	3000	4500	6000	9000
22.05		1378	2756	4134	5513	8269
16		1000	2000	3000	4000	6000
12		750	1500	2250	3000	4500
11.025		689	1378	2067	2756	4134

Figure 4: Treble and Bass Corner Frequency Selections

Soft Volume

Each of the 3 monaural processing channels in the TAS3103 has a precision soft volume control that is separately programmable via the I2C bus. The volume can be adjusted from +24 dB to – infinity.

Soft Volume Control

The soft volume controls, like the treble and bass controls use a "fire and forget" approach. That is, the system integrator need only send the desired destination volume and the TAS3103 will make a smooth transition over about 40 msec with no clicks or zipper noise. If multiple commands are sent to a channel during the 40 msec, they are overwritten in the internal register and the last command before the start of the next 40 msec will be used in that transition. This approach avoids the queuing problem of "wind-up" caused by a large number of unexecuted volume commands. This means that a digitized volume control or shaft encoder can be turned continuously and randomly by the user with completely smooth response.

Soft Mute/Unmute

In addition to individual volume control, the TAS3103 provides a soft mute command that individually mutes each of the 3 channels in about 40 msec with an artifact-free transition. The advantage of the mute command over the volume control is that the unmute command restores the previous volume setting with no need for the external control microprocessor to read and re-send the original setting.

Immediate Mute/Unmute

If an immediate mute/unmute command is needed, any of the in-line mixes can provide an immediate mute/unmute function that eliminates the delay for muting. This will produce an audible click, but is useful in some situations.

Loudness Compensation

The loudness compensation block tracks the volume control setting to allow spectral compensation. This is often used to compensate for weak perceived bass at low volume levels. It directs the input to an unmodified path and a path filtered by a single biquad filter. The filtered block is gained and mixed with the unmodified path. The gain of the filtered path is established by a set of gain and offset parameters that use the volume as an input parameter and calculate the amount of the filtered channel to add to the unmodified path. Both linear and log control laws can be implemented and any desired filter can be used. A typical filter is a bell-shaped boost filter centered in the bass region desired with an appropriate bandwidth.

Dynamic Range Control

The TAS3103 provides a powerful dynamic range control capability. The left and right channels have a ganged control capability and the 3rd channel provides a separate dynamic range control. Each block has two programmable thresholds that define 3 programmable slope ranges. These regions can implement both boosts and cuts. The energy integration time, attack time, and decay time are all individually programmable for complete control A few typical uses are shown below.

Simple Compression/Expansion

In the application below, only one of the slope regions is used. In this case, a slope higher than that of the angled dotted line, (which indicates a linear response), is an expansion and a slope lower than that of the dotted line is a compression. Segments above the linear line are boosts and sections below the dotted line are cuts.

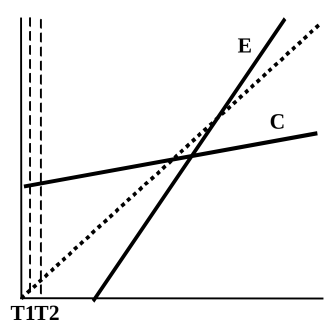

Figure 5a: Simple Compression/Expansion

Late Night Mode

This mode is used for listening whenever volume must be kept low. Loud sounds are compressed with a cut and low sounds are compressed with a boost. This dynamic range compression, coupled with the volume control, allows all of the program material to be heard above the room background noise without the need for loud peaks.

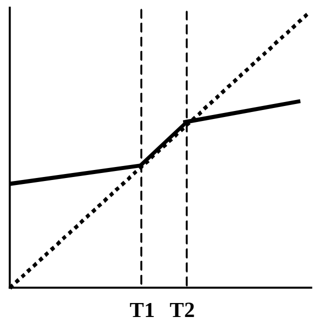

Figure 5b: Late Night Mode

Background Noise Floor Compensation

When listening in a noisy environment, for example in a car travelling at various speeds, it is sometimes necessary to turn the volume up to hear low sounds, resulting in peaks that are louder than desired. This application of dynamic range control involves adjusting the degree of simple compression as a boost of low signals. This boost, coupled with the volume control, allows the program material to be placed between the background noise level and the loudest peak desired. If the VU meter output is used along with a digitized microphone input by the system microprocessor, the degree of compression and volume can be adjusted adaptively to implement a high-quality algorithm that maintains program quality and peak volume as the speed of the car changes. The noise level could also be sensed indirectly by having the system microprocessor use the vehicle speed indication already available from cruise control operation.

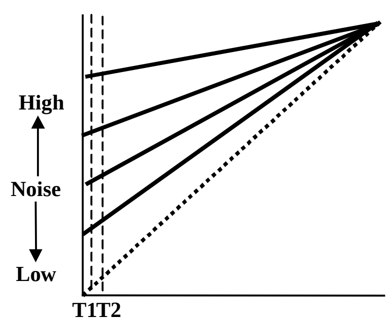

Figure 5c: Background Noise Compensation

Background Noise "Squelch"

In applications like headsets and teleconferencing, it is desirable to mute background noise when the user stops speaking. The configuration below provides a "squelch" of low-level background with an adjustable level. Thus when the speaker stops talking the channel is muted and no sound comes through. This prevents unwanted low-level background conversations and noise from being transmitted. The attack/decay time constants can be set to prevent cycling between words.

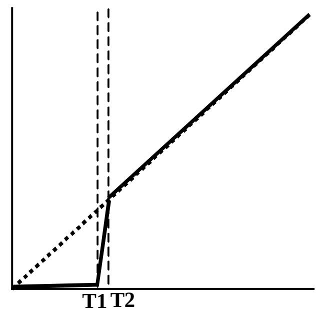

Figure 5d: Background Noise "Squelch"

Peak Limiting/Clipping

High compression levels can be used for limiting the program loudness to protect speakers. The curve below applies a progressive compression that allows the average program level to sound quite loud, while protecting speakers from extended thermal overloads.

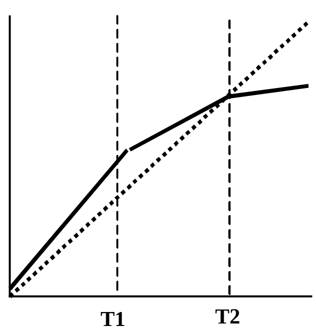

Figure 5e: Peak Limiting/Clipping

Center Channel Synthesis

There are two ways to form a center channel from the left and right channels in the TAS3103. The 3^{rd} channel can be used, as described in a later section, to form a completely independent channel with complete audio control. Figure 3 above shows a low-cost alternative that keeps the 3^{rd} channel free for other applications. The summing point "r" in the diagram mixes left and right proportionally. The sum of these two signals is then provided to an output multiplexer that allows this signal to be output from any of the 6 outputs to be formatted by the serial output port (see Figure 7). The output multiplexer function will be discussed below.

Programmable Dither/Final Truncation

The 48-bit output is truncated back to 32-bit I/O precision in the block shown in Figure 3. In the same concept used in the input, the top 8 bits and the bottom 8 bits are removed. There are a number of scaling points in the audio chain, including one immediately before this block, that allow preservation of the 0 dB level. Since each scaling multiply has both gain and attenuation capability, it is easy to control the flow of audio data through the channel.

The output serial audio port may further truncate the output precision depending on the output word size chosen (16, 20, 24, or 32 bits). Because a truncation introduces artifacts in the output due to quantization noise, a programmable output dither can be summed into the output before truncation to eliminate audible artifacts. Both triangle-weighted and uniform distribution dither outputs are possible, and the dither level is adjustable from the LSB of an 8-bit audio signal to below the LSB of 32 bit data , including an “off” position.

Preserving 24-Bit Audio

The 48-bit data path of the TAS3103 preserves data quality by placing the truncation level at 32 bits (-192 dB) below a typical 0 dB point. This means that even 24-bit data that has been dithered can be processed without damage. 24-bit input data is input with 16 noise bits beneath it, so multiplication operations preserve initial dithering.

Uses For 32-Bit Audio Transfers

This also means that TAS3103s can be cascaded end-to-end with a dithered 32-bit transfer between them to allow considerable audio processing while maintaining 24-bit dynamic range in the dithered output of the final TAS3103. This makes the TAS3103 a very useful IC for professional audio applications in digital audio workstations and mixer panels. This is augmented by the fact that all multiply operations provide bit-perfect outputs when set to “1”, “0” or “-1”, or any integer or fractional power of 2.

Delay

Each channel has two delay points. The first, listed as “reverb/delay” in Figure 2, is implemented with 48-bit precision and can be either a single reverberation or a delay. The second occurs after the 32-bit truncation and has 32-bit precision. The possibility of regenerative reverberation is discussed below in the 3rd channel discussion.

Delay “Pool” Concept

In order to provide the maximum flexibility, all of the included delay memory is available in any mix to the 6 reverberation and delay blocks. The total memory is 4K words of 16-bit precision. Each 48-bit reverb/delay step takes 3 of the 16-bit memory words and each 32 bit final delay step takes 2 memory words. For this reason, simple delays are more efficient at the end of the channel processing. The 6 memory pointers can be programmed to allocate part or all of the memory among the 3 channels. Each step is one sample period. For example, a sample rate of 48 KHz has a step period of 20.83 microseconds, giving the pool a maximum 32-bit delay of 42.7 msec.

Audio Image Modifications

The output of each channel can be adjusted in image location using the delay blocks. The example sample size above at 48 KHz allows adjustments of a fraction of an inch in imaging location. A 1-foot adjustment of an output with respect to the others uses 42 sample intervals of delay. This is about 2% of the delay pool capability at 48 KHz sample rate.

One example of this process can be seen with a center channel created in the 3rd processing channel. The filter block or tone controls can be used to create a passband for the desired center performer (for example, a lead singer or soloist). The delay blocks can then be used to move the image of the lead performer several feet in front of the band or orchestra, creating a much more “live” soundstage. If the filtering operation is not needed, the same process can be achieved with the back-end center channel syntheses, saving the 3rd channel for subwoofer processing.

Dispersion Compensation

As will be discussed below, the 3rd channel can be used to synthesize a subwoofer channel. The group delay of bass notes in the subwoofer channel can be quite different from the main program material. The delay block allows compensation for group delay differences.

Output Mixer/Multiplexer

This block is provided to allow selection of where each of the channels and the synthesized center channel occur in the output serial data. This block, along with the input crossbar mixer, provides complete freedom to reformat data as part of the processing performed. This can be used for general re-ordering, for left/right reversal, and for phase inversion of outputs.

The output mixer/multiplexer occurs after the 32-bit truncation. For this reason, any multiplication applied at this point can disturb the dithering applied before the truncation. For this reason, in critical applications this block should be used only as a multiplexer with phase change capability. This means that the multiply values should be “1”, “0”, or “-1”, or any integer power and some fractional powers of 2. In these cases the dithering is preserved.

Spectrum Analyzer/VU Meter Output

The TAS3103 provides internal spectrum analyzer and VU meter implementations with the block diagram shown in Figure 6. It consists of 10 fully-configurable biquad filters that will typically be configured with center frequencies at 1 octave or 2 octave separation with variable band width to cover the audio frequency range. The outputs of each biquad are absolute value converted and filtered through a first-order filter to detect the envelope. The time constant of the first-order filter can be set for desired spectrum analyzer dynamics. The output is then log converted and the output is provided to 10 registers as 1-byte, 4.4 format binary numbers.

The 10 bytes of information can be polled efficiently via the I2C data bus. Reading all 10 bands at a 20 Hz update rate via the I2C bus results in bus loading of under 2% throughput at 100 KBits/s and ½% throughput at 400 KBytes/s.

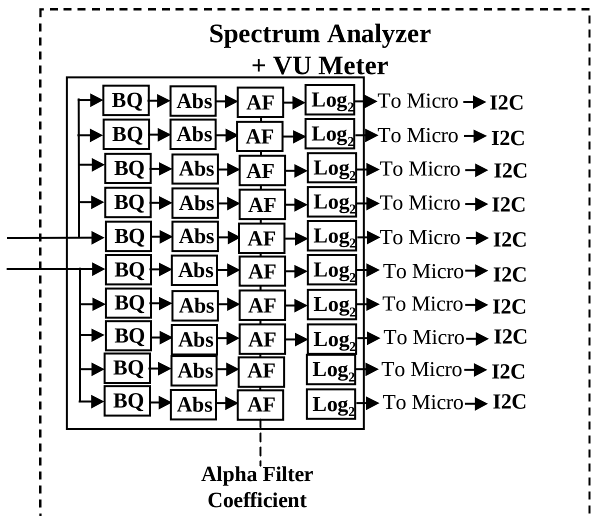

Figure 6: Spectrum Analyzer/VU Meter

Spectrum Analyzer Options

Several mix points before the spectrum analyzer inputs (shown in Figure 7) allow several options for data routing.

First, any of the 3 channels or any combination of them can be applied equally to both of the inputs shown, allowing a 10-band (1 octave spacing) spectrum analyzer output of the selected program material.

In a second approach, the left channel can be supplied to one of the inputs and the right channel to the other input to create a stereo 5-band (2 octave spacing) spectrum analyzer.

VU Meter Operation

A second read-only subaddress selects only the two central channels. If those two biquads are placed in an "allpass" configuration, the two outputs implement a stereo VU meter with one-byte logarithmic readout of each channel. The user can select either the spectrum analyzer or the VU meter function, but not both.

3rd Channel Techniques

Figure 7 below adds the 3rd channel to the previous view in Figures 2 and 3. The 3rd channel can be used for a number of support techniques, as well as providing a 3-channel processing capability that can be used in chip pairs to implement 6-channel processors. As before, the input serial audio processor converts the I2S formats into a 6-channel PCM input to the crossbar multiplexer. The 6 PCM outputs shown in figure 7 enter the output serial audio processor to be converted to I2S serial audio output formats.

As can be seen, the 3rd channel can receive a mix of any of the 6 input channels. Further processing is similar to channels 1 and 2. The following examples indicate some of the ways that the 3rd channel can be used. Note in Figure 7 that the output of the 3rd channel can either be multiplexed to any of the outputs or can be summed into the output of the left and right channels. This provides considerable flexibility in use of the 3rd channel.

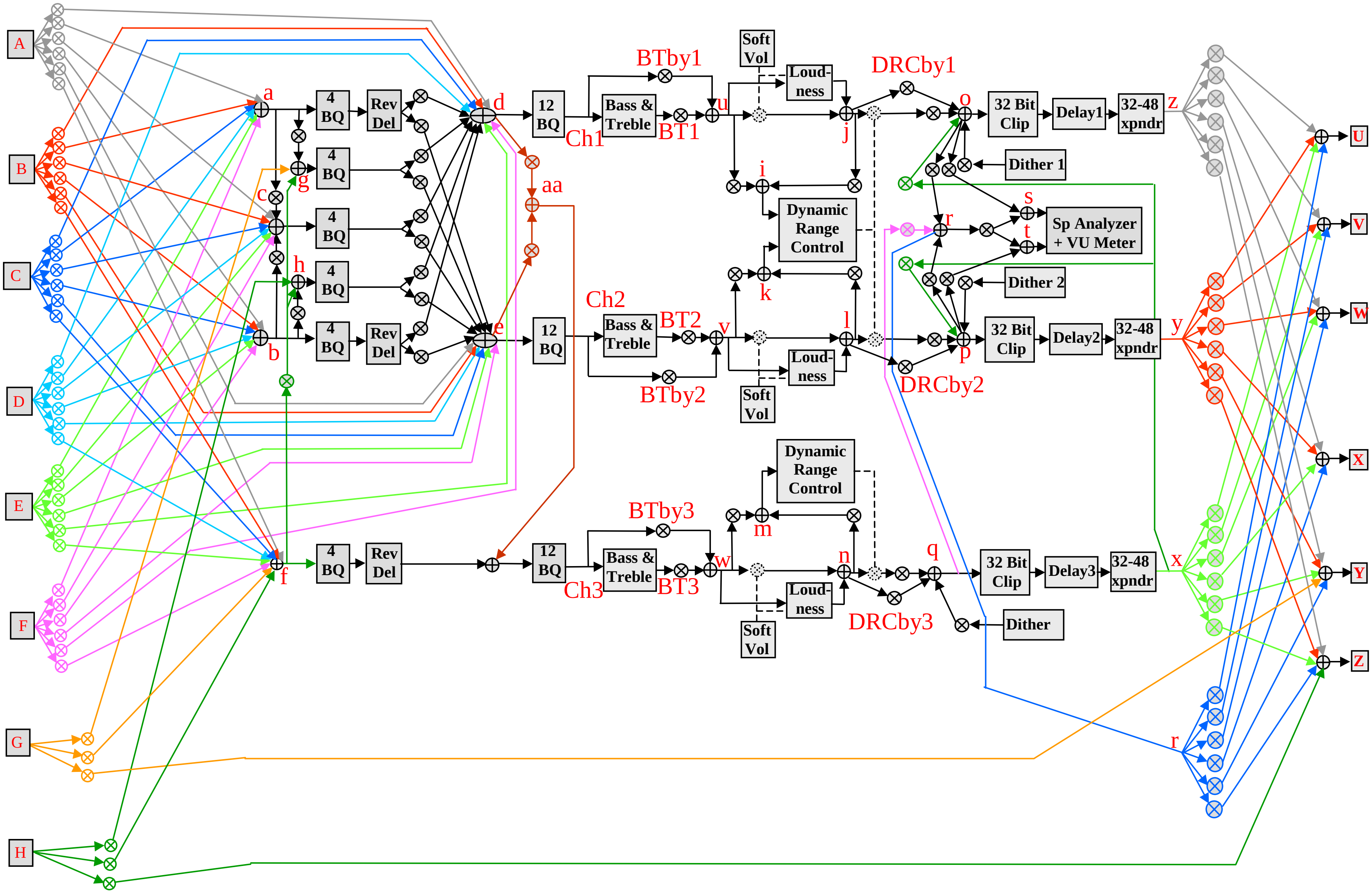

Figure 7: TAS3103 Functional Block Diagram

Subwoofer Channel Synthesis

One of the most common uses of the 3rd channel is the synthesis of a subwoofer or low frequency effects channel. With a 2 channel input, the 3rd channel implements a 2.1 channel output. The filter block in each channel can act as a precision crossover removing the subwoofer input from the left and right channels if they are to be sent to small speakers that would have a distorted output or be damaged with the low frequency signal.

Processing of the subwoofer output is not limited to the low-pass filtering in the filter block. The loudness block can be used as a simple bass enhancement at low volume settings. An even more sophisticated bass enhancement capability is provided by using the filter block to shape the desired bass response overall along with a separate dynamic range compression configuration. This can be set up to adaptively enhance the bass with decreasing output levels, greatly enhancing perceived bass without any of the normal distortion effects encountered at volume peaks. The loudness control can be added to further augment the bass dependent on the volume setting. This provides a very flexible and powerful tool for bass enhancement.

Separate Subwoofer Output

The 3rd channel can then be sent out as the left or right channel of a stereo output. (Note that the synthesized center channel can be sent out on the other channel of this output.) This can be used with a DAC to produce a line output for powered subwoofers.

Summed Back Into L/R Channels

It is also possible to sum the output of the subwoofer channel back into the left and right channels for reproduction by wide range speakers. This allows the capability to provide all of the bass enhancement features discussed above in simple stereo systems. This is very useful in mini/micro component systems, televisions, and conventional stereo systems. It is easy to switch back and forth between both of these implementations or to mix the desired amounts of the 3rd channel into both configurations at the same time.

Hall Effect Synthesis

The 3rd channel is also very useful for pseudo-surround and hall effects. The left and right channels are mixed into the 3rd channel. The techniques vary from simple simulated reflections from side and rear walls of a small room to regenerative reverberation.

Filtering Techniques

First, the filter block of the 3rd channel is used to shape the spectrum to simulate reflected sound. Typically, this involves a low-pass filter with a 6-8 KHz bandpass simulating high frequency loss in the reflection.

Delay/Reverb Techniques

For the longest single reflection times, all of the delay is allocated to the 3rd channel's final delay, which then multiplexes its output to two rear speakers, one in phase and one with inverted phase. This creates a simple surround effect. A more sophisticated effect can be generated if the reverb block generates a first reflection with a short time constant and the final delay creates a reflection with the rest of the delay memory. This simulates both side and back wall reflections in a small room. The total time available at 48 and 96 KHz is rather limited, however. The following section shows how to use regeneration to get longer reverberation time constants.

Regenerative Reverberation

In this approach, the output of the 3rd channel is wrapped around to its input via external wiring and configuration of the input mixer and the output multiplexer. If the gains are greater than one, the channel will oscillate. If the gains are set appropriately less than one overall, the loop will regenerate audio signals, creating reverberation. Signals are mixed into the loop with the input mixer and taken out with either mix into the effects block or the mix into the back end of the left and right channels. With all other parameters set properly, the soft volume can be used as a smooth regeneration control, preferably with the maximum point set below oscillation. If the regeneration percentage is set to 80-90%, regeneration times of 1-2 seconds are possible. Use of the output dither is useful in this case to prevent build-up of tones.

Other elements in the 3rd channel can enhance the effect. If both the final delay and a short reverberation delay are used, each pass through the loop adds 2 reflections, quickly simulating the many reflections heard in typical room reverberation.

The 4 biquad filters before the first reverb/delay block can be configured as a phase-dispersive allpass filter, simulating the dispersion effects that occur during long reverberations. Each pass will further disperse the sharp “reflections”, smoothing the general reverberation and preventing the problems normally associated with simple reverberation schemes.

The filter block can provide a progressive filter function, causing successive passes to have either a tendency toward lower frequencies in the reverberation or preventing this effect if it is not desired and occurs naturally within the regeneration loop.

6-Channel Systems

Two TAS3103s can implement 6-channel systems with no added glue logic. Several features facilitate this capability:

First, each TAS3103 accommodates a 6-channel input and has a crossbar mixer that allows access for each of the 3 channels in each chip to all of the input signals, assuming the serial inputs are wired in parallel to the inputs of both chips. This means that cross-channel mixing can be easily accomplished.

Next, one of the stereo outputs in each chip is routed to one side of an internal OR-gate feeding SDOUT1, as shown in Figure 1. This is important, because the 3-channel format means that one of the stereo pairs is split between the two chips. One input of the OR gate is one of the monaural channels, the other is an external pin called ORIN. This gate logically ORs the two inputs onto SDOUT1. The OR gate is used with the serial formatting feature below.

Finally, each of the chips can be designated as a "chip A" format or "chip B" format with one of the subaddress configurations. This designates which data location or locations are active and which are zeros in the output serial data. Because of this formatting, a half-output from one chip can be overlayed, or logically ORed without interference with the output of the second chip, reconstituting a full stereo output. Thus, one stereo output comes from one chip and two outputs come from another. This feature is available in both discrete and time-division-multiplexed serial formats.

The combination of these features allows a chip pair to accept and output serial data in a full emulation of a 96 KHz 6-channel, single chip format with no additional glue logic.

When the internal processing shown in Figure 7 is doubled in a 6-channel application, the full complement of normal audio processing is available in all 6 channels. As can be seen below, there are further advantages to this approach.

The question might be asked: "Why not just create a 6-channel version of the TAS3103, rather than requiring two chips?" The TAS3103 is designed to operate at sample rates up to 96 KHz. This means that the processing shown in Figure 7 must be executed in about 10.4 microseconds. Doubling the channel count would mean that only half as many instruction cycles are available. Other versions of the TAS3103, with different channel counts and features are currently under design.

5.1 Channel Processing

Both Dolby Digital and DTS outputs typically have 5 main channels and a subwoofer or low-frequency effects channel. A pair of TAS3103s is ideal for providing the end user considerable control over how the program material sounds. A typical system implementation will be to place the pair of TAS3103s in the serial digital stream between a decoder and an output DAC or digital amplifier. This simplifies the decoding routines considerably, provides much more audio processing than would otherwise be available, and allows more decoding options to be stored in the decoding device.

The TAS3103 provides all of the functions discussed above, plus features normally provided by "hall effect" processors following decoders in AV Receiver, DVD Player, and HTIB applications. Several of these features will be discussed below. Note that with 2 devices, either a stereo 10-band spectrum analyzer or a 4 channel, 5-band spectrum is possible.

5.1 Channels with 3D Algorithms

There are a number of 3D algorithms that can be implemented with two TAS3103s. Several third-party algorithms appear feasible and are under discussion with the developers. Once testing and arrangements with the developers are complete, future application notes will describe them.

5.1 Channel to 2 Channel Downmix

One application of the TAS3103 pair (or even of a single TAS3103) is to downmix a 5.1-channel input to a stereo output. There are two ways this can be done. One is a basic downmix to stereo without regard for spatial orientation except for left and right stereo separation. The second is a virtual surround synthesis that uses head-related transfer functions and possibly inter-aural delays before mixing to encode spatial cues on the 5 main channels so that the surround channels are still perceived in the stereo output. Several 3rd party approaches to this feature appear promising and will be discussed in future application notes.

5.1 Channels with Reverberation (Hall Effects)

The regenerative reverberation approach discussed above in the single chip 3rd channel discussion can also be used to enhance 5.1-channel program material. In this case, the center channel is synthesized using the left/right mix near the end of the main left/right processing channels. This frees up the 3rd channel on that chip to do the regenerative reverberation approach, generating hall effects through early reflection and reverberation synthesis. The user has the choice of applying reverberation to the main left and right channels or using the 3rd channel on the chip with the surround channels and applying reverberation to the rear channels. If a separate subwoofer channel is not needed, independent reverberation can be applied to both the main and surround channels with different characteristics, providing almost unlimited flexibility in reverberation effects.

Combinations of Effects

As one example of the degree of sophistication available with two TAS3103s, consider the following: The main left and right channels can be modified with a stereo “spreading” or “augmentation”, of which there are a number of 3rd party choices. Similarly, the surround channels can be spread with the other effects block and delayed in the second chip’s delay memory. This has the effect of providing a very wide stage, front and back and of simulating a reflection from the rear wall.

If the front and rear reverberation approach discussed above is added to the stereo spreading, the combined algorithms should create the impression of a very large sound stage or hall in the 5.1 audio image.

2 Channel to 5.1 Channel Synthesis

One final approach that is possible with 2 TAS3103s is the synthesis of believable 5.1 data from a stereo input, even one without matrix surround. While this approach will not generate the precise imaging of a true 5.1 decoded program source, it will greatly augment the portrayal of stereo in a sound system with a 5.1 speaker complement.

The approach starts with processing of the left and right channels, which are sent to the left and right speakers. If the separate subwoofer channel is desired, a high-pass filter can be implemented to divert the low frequencies from the left and right channel speakers. The stereo spreading algorithm can be used if desired.

The final mix of the left and right channels synthesizes a center channel. This output tracks the volume and tone control of the left and right channels.

The 3rd channel of the same chip is used to generate a regenerative reverberation effect that can be summed into the left and right channels and externally to the surround channels to be created in the second chip.

Surround channels are created in the second chip in left surround and right surround channels that are low-pass filtered to 6-8 KHz and each delayed by about 21 msec (at 48 KHz sample rate) to simulate the high-frequency loss and travel time of rear reflections. The reverberation generated in the other chip is mixed into these channels. The stereo spreading algorithm can also be used for the surround channels if desired.

Finally, a subwoofer channel is synthesized by mixing the left and right inputs into the 3rd channel of the second chip. This channel incorporates a low-pass filter that is essentially the other half of a crossover between the subwoofer and the other speakers. All of the dynamic bass boost features described in the single-chip sections are available for this channel.

This configuration should provide a much better presentation of stereo information in a 5.1 sound system than the normal “5-channel stereo”. Many variations of the processing parameters of this approach are possible.

Conclusion

The TAS3103 provides an incredibly versatile audio processing capability. While all of the individual features are available in other devices, it is unique in the number of features, the high sample rates possible, and the audio quality supported. It is designed to be inserted in almost any I2S data stream to provide all required audio processing needs. Two TAS3103s seamlessly emulate a 6-channel processor without glue logic at up to 96 KHz sample rates. It is an appropriate circuit element for applications from low-end TV and mini/micro component systems to high-end AV receivers and pro audio. This allows the TAS3103 to span complete product lines with a single hardware and configuration approach. Finally, when applications are developed with the evaluation module and PC-based GUI configuration tool, application development times can be reduced to hours, rather than weeks.