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Editing 8-channel DSD Signals Using NPC SM5951A as Example

(Part.1)

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NPC offers the following three products in its lineup for super audio CD (SA-CD) LSIs, which were already introduced in this magazine.

- SM5866AS, high-performance mono-channel DAC supporting both DSD and PCM
- SM5816AF, 6-channel DSD-to-PCM ($8f_S$, $2f_S$) converter
- SM5819AF, 6-channel DSD-to-PCM ($4f_S$, $2f_S$, f_S) converter

As mentioned in the Confidential column in the April issue of this magazine, NPC released an 8-channel DSD (Direct Stream Digital) signal editing LSI, SM5951AF, in cooperation with Sony, in February. This LSI inputs four DSD signals through each of its eight channels, mixes these DSD signals, and converts them into one DSD signal for output.

This article introduces this LSI, starting from an explanation of DSD signals, and continues onto the next issue of the magazine. This issue discusses the present status of DSD editing and features of DSD signal processing.

Present status of SA-CD software production

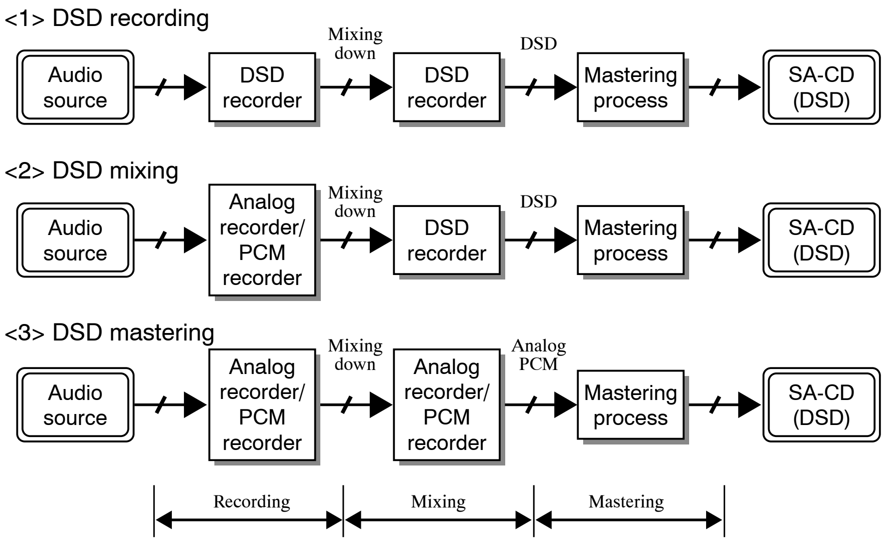
SA-CD software currently available on the market can be divided into the following three production methods, as shown in Figure 1.
(reference: http://www.superaudiocd.com/, etc.)

<1> DSD recording
<2> DSD mixing
<3> DSD mastering

The DSD recording SA-CD in <1> is, as its name implies, was recorded by the DSD method at a studio.

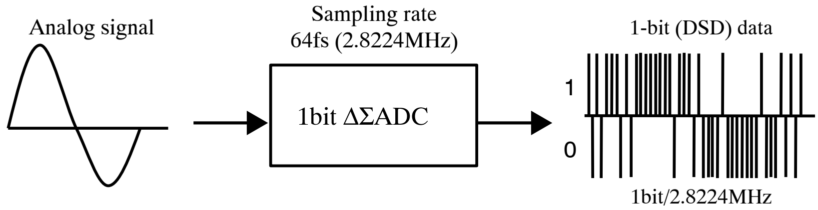
The DSD mixing SA-CD in <2> was recorded to an analog or PCM multi-channel recorder at a recording studio or live, and directly mixed down to DSD. While DSD recording is used to record the source with DSD, DSD mixing is for recording with analog or PCM and then converting the mixed sound into DSD.
For the initial digital recording, a 24-ch or 48-ch multi-track recorder (MTR) for professional use is used. An SA-CD is produced by mixing down the source from MTR to a master recorder (2 to 5.1 ch). Subsequently, in the mastering process, the order of music, sound quality, and volume of the master tape are adjusted or modified to create master data for the SA-CD.

The DSD mastering SA-CD in <3> uses a source that was directly converted from an analog master tape to DSD, or a digital master tape of high bits or high sampling frequency converted into DSD. To convert a CD of 44.1kHz/16 bits to DSD, x64 oversampling is performed and then converted into 1 bit by a noise shaper (ΔΣ converter).



<Figure 1> Three Types of DSD Software Editing Process

The number of SA-CD software titles available on the Japanese market exceeded 1,000 at the end of the last year. However, the quantity of DSD recording SA-CD in <1> is still not much, and many SA-CDs have been created by re-mastering sound sources of the past or DSD mixing from analog sources or PCM digital recording sources. In this sense, DSD recording SA-CD in real sense of the term has been expected.



<Figure 2> DSD Signal Generation (DSD signal at $64f_s$ rate)

According to Mr. Kunisaki of "Sound & Recording Magazine" in his lecture at the 1-bit Consortium Assembly at Waseda University in February 17, DSD recording of a single recording is insufficient for the needs of current software production for which over dubbing and punch-in/out are necessary, and for DAW (Digital Audio Workstation) using a computer is necessary. At present, PCM recording by the Protools HD 192 kHz/24 bits seems to be the most widely spread.

Therefore, realizing of multi-channel DSD recording, punch-in/out, computer-controlled DSD mixing, signal processing with EQ and compressor, and DSD mastering are urgently required for the future DSD recording.

At present, DSD8 of SADiE of England is available as DAW for DSD editing. This DSD8 is an 8-ch DSD/PCM mastering workstation and can perform EQ and Dynamix processing (such as compression and limiting) at DSD rate ($64f_s$). It also supports down conversion from DSD to PCM (16 bits/44.1 kHz).

As multi-track DSD recorders supporting both DSD and 24 bits/192 kHz, GX9000 and GX-9048 of Genex of USA are available. Both these DSD recorders support 8-ch and 48-ch.

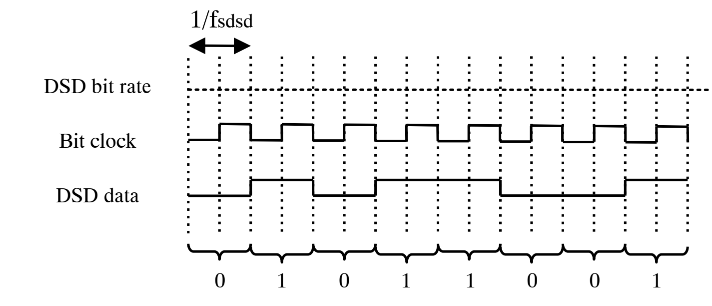
These workstation and recorders are designed to process signals at DSD ($64f_s$) rate and to maximize the sound quality in the internal DSD signal processing process.

Features of DSD signal

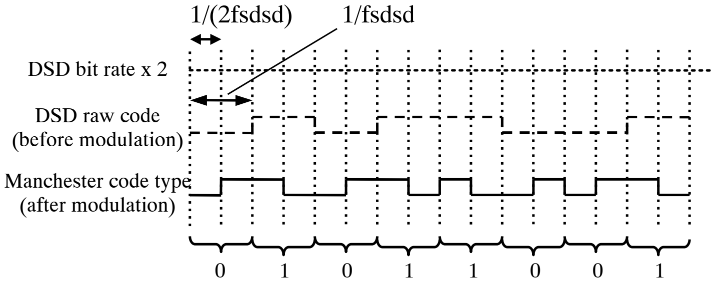
Now, let's see the features of DSD signal processing in connection with DSD signal mixing.

(1) DSD signal format

A DSD signal is originally a digital code that is created by sampling an analog signal with a 1-bit delta-sigma (ΔΣ) A/D converter of the fifth to seventh order at a rate of $64f_s$ (64 x 44.1 kHz = 2.8224 MHz), as shown in Figure 2. However, the source signal of a DSD signal is not always limited to an analog signal.



<Figure 3> Waveform of DSD-raw Signal



<Figure 4> Waveform of SDIF-3 (Manchester code type) DSD Signal

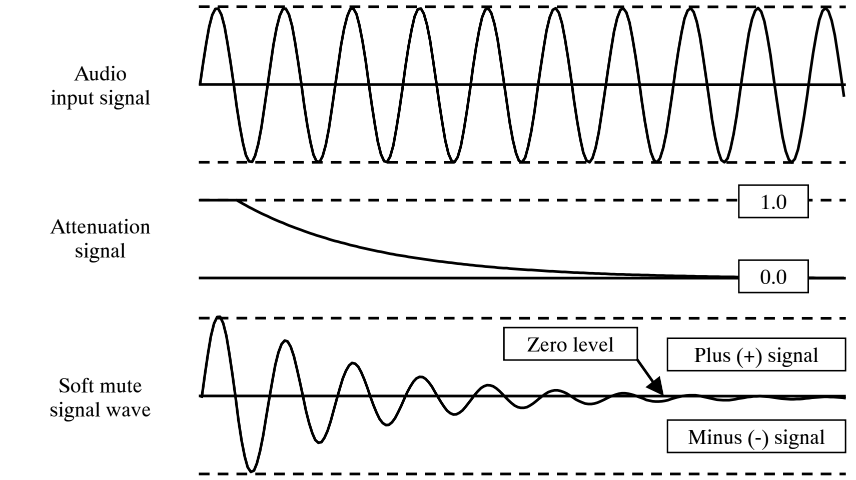
The DSD signal that transmits a 1-bit bitstream on which ΔΣ conversion has been performed from one piece of equipment to another is currently in two transmission formats.

The most commonly used format is "DSD-raw" shown in Figure 3. This format was once called a prototype format. This format is, where DSD rate is f_{sdsd}, consists of a bit clock with 50% duty of f_{sdsd}, and 1-bit data of a ΔΣ converter synchronized with the falling edge of the bit clock.

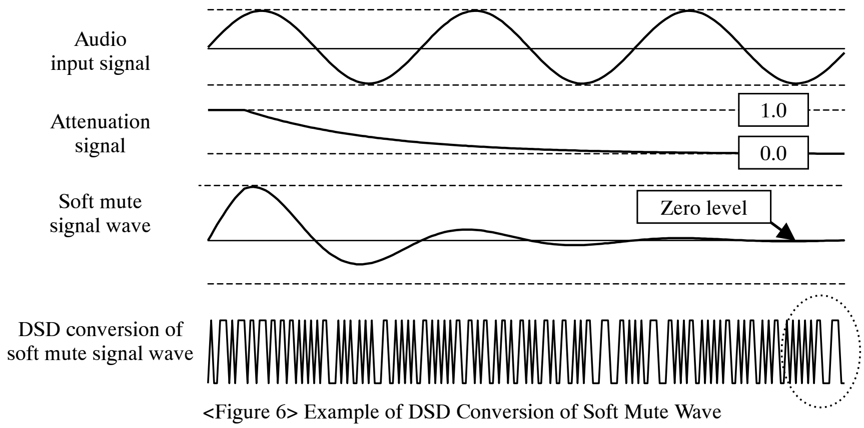
The other format is recommended especially for equipment for professional use and is called SDIF-3, as shown in Figure 4. This format may also be called Manchester code type DSD because of its modulation method.

By this modulation method, even if codes that are 0 or 1 continue, they are modulated so that bit inversion always takes place at the center of bit data. As a result, it has merits of reducing the influence of the transmission cable on the sound quality and, because a break in the cable can be detected, of decreasing troubles of connection. This Manchester code type uses a bit clock rate two times higher than that normal, at $2f_{sdsd}$.

In both the above formats, Separate 44.1kHz Word Sync is necessary as a clock that synchronizes equipment if it is for professional use.



<Figure 5> Soft Mute Operation Processing of Analog Signal (PCM)
(0 when original signal is completely muted)



<Figure 6> Example of DSD Conversion of Soft Mute Wave
(1 and 0 are repeated when original signal is completely muted)

(2) Muting DSD signal

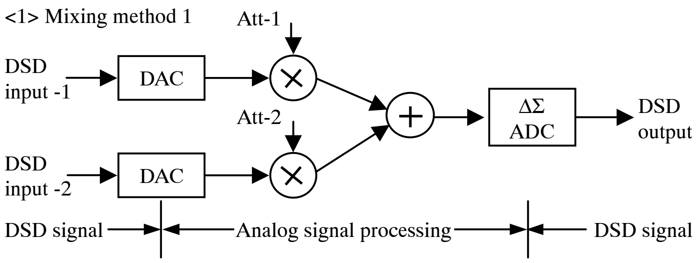
If an audio signal is occasionally muted, it sounds like large tics. Generally, therefore, soft muting that gradually attenuates the audio signal is employed.

Figure 5 shows the waveforms of an audio input signal and a gain curve for soft muting of an analog signal, and of a soft mute signal resulting from multiplying them. As can be seen, the sine wave is gradually attenuate toward ±0.

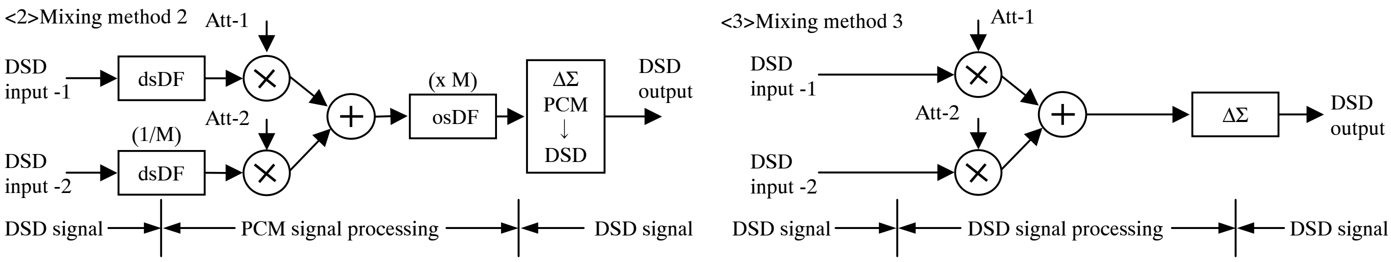
Even if the audio input signal is PCM, it takes a center value 0 of 2's complement after soft mute and thus is the same as when an analog signal is input.

Figure 6 shows the result of simulating the waveform after DSD conversion of the original analog signal on which soft mute has been performed.

As is evident from this figure, the value of the converted signal alternates between 1 and 0 at a ratio between 1 and 0 of about 50% even when the original signal is completely muted, and does not converge to a specific DC level unlike when an analog or a PCM signal is muted. This is a very important point in handling the DSD signal.



◀▼ <Figure 7> Three Types of Mixing for DSD Signal



Example of DSD signal processing

Before explaining the functions of SM5951AF, NPC newly released, some basic examples are given of handling the DSD signal that has the above features.

(1) Mixing of DSD signal

When creating DSD software, mixing is an important signal processing process except when a DSD signal is directly recorded. Mixing is a process to multiply at least two signals by a specific weight and add up the signals. Figure 7 shows three types of general methods of mixing two DSD signals into one.

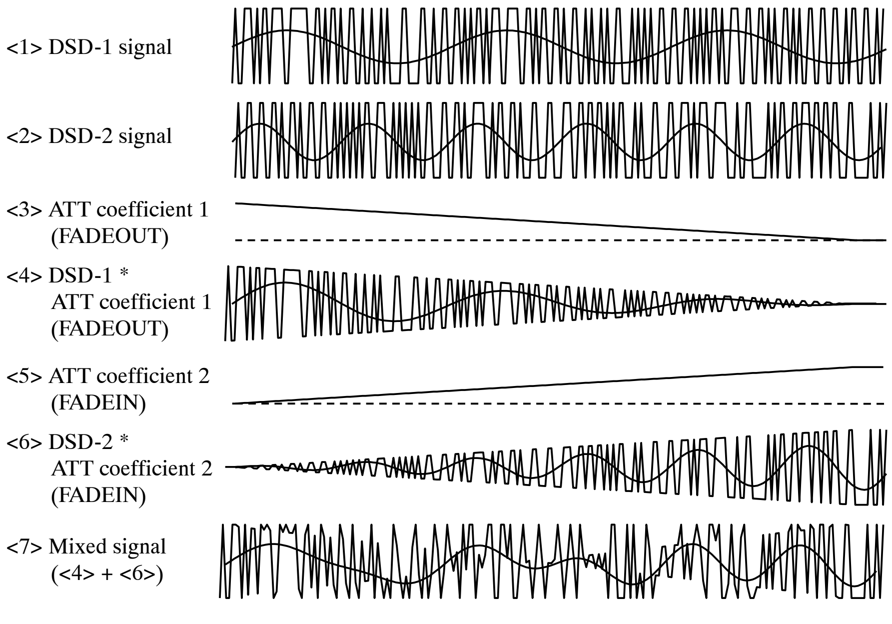
Method <1> is the orthodox method. Two DSD signals are once converted to analog signals by DACs. There analog signals are attenuated and mixed in analog domain. The resultant mixed analog signal is converted into a DSD signal by a 1-bit ΔΣ ADC converter for output.

Method <2> uses a downsampling DF (dsDF) to convert each of two DSD signals into a PCM signal at a DSD rate of 1/M (integer). The resultant PCM signals are then multiplied by each attenuation coefficient with a DSP or operation processor and mixed. Upsampling is performed on the mixed signal with an over-sampling DF (osDF) of xM, and then converted into a DSD signal by a digital ΔΣ converter. This method has a merit that a system can be configured by using the existing DF, DSP, or processor.

Method <3> execute digital signal processing at DSD rate entirely. One of the advantages of 1-bit signal processing is that multiplication of 1bit by multi bit coefficient such as 24 bits can be executed by flipping 1s and 0s in coefficient alternatively. In addition, DAC and ADC for which the influence of minute analog noise and clock jitter cannot be ignored, and dsDF and osDF for which the influence of minute digital noise cannot be ignored during operation processing are not used. Therefore it can be said that 1-bit signal processing is the simplest and ideal method.

(2) Cross-fade of DSD signal

By controlling mixing of the DSD signals well, the two DSD signals can be cross-faded (faded in/faded out).



<Figure 8> Cross-fade of Two DSD Signals. <7> is in multi-bit status at DSD signal rate.

Figure 8 shows the concept of cross-fade of two DSD signals. Assume that the mixing circuit in <3> is used for circuit diagram. The following explanation is given based on Figure 8.

Firstly, two DSD signals, <1> DSD -1 signal and <2> DSD-2 signal, that have resulted from DSD conversion of two analog input signals of sine wave are prepared. In addition, attenuation coefficient <3> ATT coefficient 1 (FADEOUT) and <5> ATT coefficient 2 (FADEIN) by which DSD-1 and DSD-2 are respectively multiplied are prepared. Then <1> DSD-1 signal and <2> DSD-2 signal are multiplied by <3> ATT coefficient 1 (FADEOUT) and <5> ATT coefficient 2 (FADEIN), respectively.

As a result, <4> DSD-1 x ATT coefficient 1 that gradually fades out the sine wave and <6> DSD-2 x ATT coefficient 2 that gradually fades in the sine wave are obtained.

These are signals of DSD rate. However, their resolution is extended to few hundreds times that of the original DSD signal because multi-bit coefficients are used for operation. It can be reasonably assumed that this operation is pretty hard for the ordinary digital audio processor because of its ultra high sampling rate.

By adding <4> DSD-1 x ATT coefficient 1 and <6> DSD-2 x ATT coefficient x 2, a mixed signal can be obtained. <1> DSD-1 signal and <2> DSD-2 signal are gradually mixed together and crossed over at the center of the time line (horizontal axis) as shown in the figure. (cont'd)

(Nippon Precision Circuits, Inc.)