

DESIGN CONSIDERATIONS FOR A LINE STAGE

The design of the DSA Pre I linestage presented a number of challenges and considerations that had to be resolved within the context of the basic design principles of DSA. Many people would think that the design of a line stage was easy; after all it only needs to have some selectable inputs and a volume control to meet the basic functionality of a line stage. However, it is those very aspects that seem so simple which turn out to be the most difficult when designing a world class line stage such as the DSA Pre I.

Some of the basic design considerations that had to be addressed are:

- How many inputs and of what type—balanced or unbalanced?
- What type of output—balanced or unbalanced-- and how many?
- Throughput gain and ability to accommodate different levels of inputs?
- Type of volume control and its relationship to left/right balance?
- Other features that add to overall flexibility and utility of the unit?
- Remote control operation and what functions to be remotely controlled?

There are other considerations that could be added to the above list, but these represent the major ones that went into the design of the DSA Pre I. Each of these brought challenges for the designer and solutions had to be found that were true to the basic principles of DSA, including:

- Maintaining an audio bandwidth that went from a few Hz to several octaves beyond the upper limits of audible sound.
- Extremely high dynamic range over the full bandwidth.
- No loop or global feedback in the design.
- Fully balanced audio input and output capability

To round out the list it was also desired to maintain the same basic chassis dimensions and appearance that DSA introduced with its renowned Phono II. The following sections will discuss each of the design considerations mentioned above and how they were implemented in the DSA Pre I.

1. Number And Type Of Inputs

It was desired to support both unbalanced (RCA) and balanced (XLR) inputs for the Pre-I. The number of each, or having both available for each input is largely a matter of available real estate on the chassis. For the Pre I, this was dictated by the desire to maintain the same chassis dimensions as for the DSA Phono II. High quality RCA and XLR connectors each require a minimum amount of area plus there must be sufficient space for fingers to add and remove cables easily. Since it was also necessary to allow space for the outputs and a power cord with power switch, it was decided to have six (6) inputs for each channel available on the back panel of the Pre I. And, rather than attempt

to overly crowd the back panel with an RCA and XLR connector for each input, there are three (3) unbalanced and three (3) balanced inputs for each channel. These are interleaved to maximize the accessibility to each set of inputs. However, knowing that it is sometimes desirable to be able to make a quick connection to an input to test a source, or connect another type of audio device, the Pre I has a seventh input which is located behind the drop down door on the front panel. This input is unbalanced (RCA) only and is selectable from the front panel, but not the remote control.

2. Number And Type Of Outputs

It was decided in the beginning that the main audio output would support both balanced (XLR) and unbalanced (RCA) connections. This is in keeping with the same capability on the DSA Phono II; however, to allow for a wider range of output loading options for the Pre I, a higher bias current driver stage was designed. Like the driver for the Phono II, this driver is fully balanced and operates totally in a Class A mode.

In addition to the main audio output, an output that can be used for recording the audio signal was also desired. Due to space considerations, this output has unbalanced (RCA) connectors only. This output is extracted from the output of the first gain stage which is before the volume control; however, it is affected by the throughput gain selection for the input being used. (See the next section for a discussion of the throughput gain selection.) This output is driven by a Class A source follower instead of a balanced driver and is capacitively coupled. Thus, to maintain the frequency response at the very low end of the audio spectrum, this output should not be loaded with anything less than 10kOhms.

3. Throughput Gain And Different Input Levels

The basic throughput gain of the DSA Pre I is 12dB. In terms of voltage gain this is a factor of four (4) which is adequate for virtually all source material while providing sufficient output voltage to drive all but the most insensitive power amplifiers. However, there may be situations where either the source is a lower than expected voltage, or the power amplifier requires additional voltage drive. To accommodate these instances, there is a toggle switch at each input on the back panel which permits selecting an additional 3dB of throughput gain—making a total gain of 15dB—for that particular input. This same switch also permits the reduction of the gain by 6dB, giving a total gain of 6dB. This is intended to be used primarily with inputs from digital sources where the outputs are typically a standard “line level” that can exceed the output levels from a phono preamp, or other source, by several dB. While the volume control can always be used to reduce or increase the output levels as required, the inclusion of this gain selectable feature was intended to reduce the need to make large volume control adjustments when switching between inputs. The gain selection for each input can be set as required, and when that input is selected the appropriate gain is also chosen. Note that there is no gain selection for the input located on the front panel; this input has a fixed 12dB of throughput gain.

Because these gain changes are implemented in the first input amplifier stage, the record output will be affected by the throughput gain selection. However, this will ensure that the record output level will not vary excessively when switching between inputs with differing input voltages.

4. Volume And Balance Controls

There are many accepted methods for designing volume and balance functions. For the DSA Pre I it was decided that the volume control would be digitally controlled—this to allow for a digital remote control—and that the balance function would be managed by incrementing or decrementing the level of the audio channels. Along with this approach, it was also desired to maintain constant impedance through the volume/balance control, regardless of level. This best supported the goal of very high bandwidths and dynamic range for the Pre-I. Other desires were to minimize the number of actual resistive components in the audio chain as the volume or balance was varied, and to maintain a constant total audio output when the balance between channels was changed. This latter desire immediately defined the minimum step size for the volume control at 0.5dB. With a 0.5dB step the channel balance can be altered in 1dB steps by incrementing one channel by 0.5dB and decrementing the other channel by the same amount. Thus keeping the total audio output unchanged.

The final need to maintain constant impedance and to ensure the necessary precision throughout the volume/balance control range led to the choice of discrete precision “attenuator blocks” based on the use of “T-pad” or “Pi-pad” designs. These designs are well recognized throughout the audio and RF and communities for fixed attenuators, but we believe that this is the first time that they have been employed for a volume control application. It was decided to use seven of these blocks in a binary sequence: 0.5dB, 1.0dB, 2dB, 4dB, 8dB, 16dB and 32dB. Various combinations of these blocks permit attenuation from 0dB to 63.5dB in 0.5dB steps, or 127 possible attenuator settings. The attenuator blocks that were designed for the Pre I employ 0.1% precision SMD resistors and each block is accurate to 1% of the step value. Each attenuator block is designed to maintain the same impedance level and they can be cascaded in any combination to achieve the desired attenuation while preserving a constant impedance level throughout. However, for the attenuator blocks to perform as desired they must be either “in” or “out of the audio chain. This requires seven DPDT relays which permit the audio signal to pass through the attenuator block, or through a “bypass” short circuit. Simply short-circuiting the attenuator block is not sufficient to preserve the proper attenuation or the constant impedance. These relays are driven by a pair of 4-bit asynchronous up-down binary counters connected as an 8-bit counter; however, the 8th bit is not used. Matching counter circuits are used for each channel, but with a common “up/down” pulse to maintain synchronization.

It was necessary to use mechanical reed relays for the switching operations for two reasons: a) They have very low resistance, and b) unlike optical relays they are totally linear, even at very low voltage levels. For this application, DSA selected a low power precision DPDT reed relay with very low contact resistance (typically < 0.1Ohms) and

very short duration contact bounce when operating (typically <0.5msec). The result is a precision attenuator that meets all of the design goals for both attenuation and balance control, and is virtually silent in operation.

5. Other Features

One feature that is often desired by audiophiles is the ability to invert the phase of the audio signal. The DSA Pre I is a non-inverting amplifier, however, like the DSA Phono II, it permits phase inversion of the audio signal. The inclusion of phase inversion is easily accomplished due to the differential amplifier topology of the Pre I. Also, because not all audiophiles will have a phono preamplifier that has a mono mode (where the two channels are summed) it was decided to also include this feature in the Pre I. The mono mode can be selected with any input regardless of source material.

The Pre I also has a “ ” headphone jack located on the front panel behind the drop down door. This jack is driven from the same driver stage that is connected to the main outputs on the back panel. However, when the headphone jack is in use, the main output is muted, but all other functionalities of the Pre I remain.

6. Remote Control Functionality

To develop the remote control functionality for the DSA Pre I, I turned to my good friend John Chapman of Bent Audio. For many years John has developed remote controls for his line of audio equipment, including attenuators, as well as providing remote controlled stepping motors for those using stepped attenuators. John modified his artfully designed 13-button remote control to address the desired remote functions for the DSA Pre I.

These include:

- Selection of all six back panel inputs
- Volume up/down
- Balance left/right
- Mute
- Run
- Phase

John also programs the necessary microcontroller within the Pre I to operate these functions as well as the microcontrollers that decode the digital attenuator bits for the digital level displays. The beauty of using the microcontroller is that, unless it is being used to operate a function, it is “sleeping” and hence adds no interference to the audio signal when not being used.

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