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Extron DMP 128 Plus Matrix Processor

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Digital Matrix Processor Extron DMP 128 Plus CV AT

With the DMP 128 Plus series, Extron offers a series of modern digital matrix processors with a range of functions that make them suitable for use in conference centers, universities, shopping centers, and many other common AV installations. The devices are available in various configurations and can be equipped with Dante network interfaces, echo cancellation, or VoIP functions as required.

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The heart of the DMP 128 Plus is a large matrix with, depending on the configuration, up to 52 or 84 inputs and 48 outputs. All device variants have twelve balanced/unbalanced mic/line inputs with preamps and phantom power (1 to 8) as well as eight balanced/unbalanced analog outputs. They also feature a configurable USB audio interface with a total of 8 combined USB channels, and USB HID control for synchronizing volume and mute with the host PC.

Depending on the model, optional inputs and outputs are available via a Dante interface (AT versions) or up to eight VoIP lines (V versions). Acoustic Echo Cancellation is available (C Versions), which is particularly important for conference functions with far end rooms. All variants of the DMP 128 Plus also have a DMP expansion connection, allowing to exchange 16 channels bidirectionally with another DMP processor or other Extron device with an integrated audio processor (e.g., DTP Cross Point series). If required, additional DMP Plus processors can be added to a system without special configuration of devices or an audio network. The DMP 128 Plus is configured using the Extron DSP Configurator Software, which accesses the device via the Ethernet or the front USB interface.

Use Cases

FIG. 01 shows a simple but very typical example of the use of a fully equipped DMP 128 Plus CV AT with all interfaces. The matrix processor receives signals locally via its analog inputs from four wireless microphone receivers and additional eight microphones via the Dante network. A PC connected via USB recognizes the matrix processor as a sound card and can play or record signals. The PC is also used for configuration via the Ethernet connection. Up to eight remote participants can be connected via the telephone line using the VoIP interface. The DMP 128 Plus manages all incoming phone lines and can perform all the necessary functions. The example in FIG. 01 is only one of

many possibilities. In practice, significantly larger and more complex systems can also be managed. External audio connections via the Dante network and control via Audio Control Panels via the ACP Port are also possible.

Structure

The internal structure of the DMP 128 Plus Digital Matrix processor is built around a central matrix as the core of the system. All inputs and outputs are connected to the matrix and can be routed and mixed as desired. The device provided for testing featured AEC, VoIP and Dante and had the following input and output paths for the matrix:

Inputs:

- 8x Analog with preamp and phantom power
- 4x Analog with preamp
- 8x Aux In for USB, VoIP or internal player
- 16x Virtual returns for internal send buses
- 48x Inputs for audio network
- 48x Dante or 32x Dante + 16x Expansion Inputs

Outputs:

- 8x analog and Dante audio network
- 8x Aux Out for USB or VoIP
- 16x Virtual Send buses (internal subgroups)
- 16x Expansion outputs for audio network
- 24x Dante and 16x Expansion inputs

A total of 84 inputs and 48 outputs are available for the matrix. Of the 48 expansion inputs, the first 32 are permanently assigned to the Dante network and the next 16 inputs are assigned either to the Dante network or to the DMP Expansion port for direct connection to other Extron DMP devices. On the output side, the 16 expansion outputs are available (in parallel) on both Expansion port and the Dante network. The signals from the eight analog outputs are also available on the Dante network. With this generous equipment and the all-encompassing matrix, all options are open to the user. For a quick start to configuring the processors, Extron has created templates for various typical applications with setup files for the device, including the associated macros as well as documentation about their function and the control assignments available for download on the homepage. Starting from such a template, it is easy to create your own configuration. Together with a good manual and a 24-page documentation in which all functions are explained, you can quickly reach your goal even when using it for the first time.

Network Connectivity

A look at the back of the device shows a row of RJ-45 network connections. In the simplest model of the DSP 128 Plus, there is initially only one Ethernet connection for configuring the device and another one for the DMP Expansion connection. A second Ethernet connection is added for the V variant with VoIP, which is used for control functions and VoIP operation. Another four network connections are added in the AT variant with a Dante Interface which can support redundancy thru the primary and secondary ports. If redundancy is not required, the internal switch can be used in a so-called daisy-chain mode to connect one device to another with a simple Dante interface.

Signal Processing

There are firmly defined functions in the input and output paths for signal processing. FIG. 03 shows these for →

Image 04: Rear view of the DMP 128 Plus C V AT. In the left half, the twelve analog inputs and eight GPIOs, each with one input and two outputs, whose function can be defined via the software. In the middle you can see the eight analog outputs and in the right half all network connections.

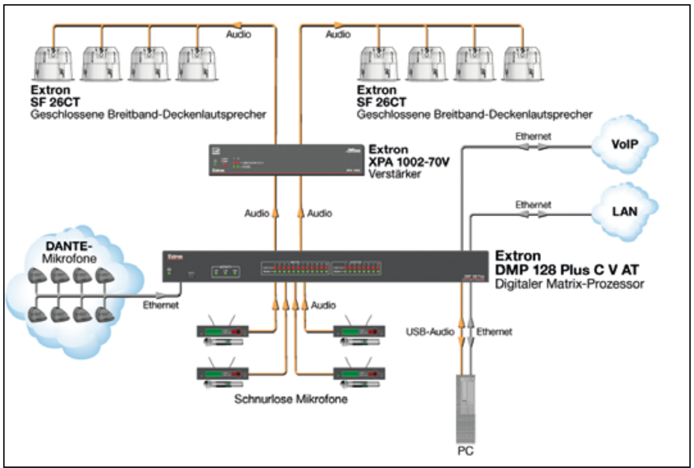


FIG. 01: Simple application example for a DMP 128 Plus matrix mixer

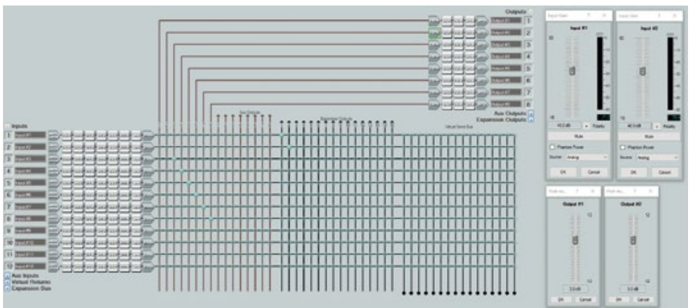


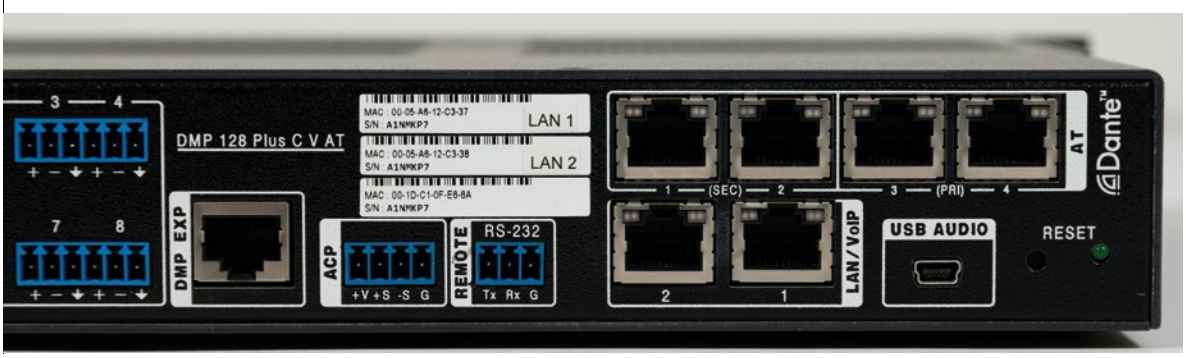
FIG. 02: Section of the 84x48 matrix in the DMP 128 Plus C V AT

the analog inputs. The associated Dialog Boxes can be found in FIG. 04 . For analog inputs 1-12, the signal chain begins with the Preamp Gain setting, which allows for values ranging from -18 to +80 dB, as well as Mute, Phase Reverse, and Phantom Power. The broad adjustment of the gain values takes place from 0 to +40 dB on the analog side and beyond that in both directions as digital gain. Alternatively, inputs 1-12 can also be linked to the Dante/expansion inputs, where the gain range is then reduced to purely digital -18 to +24 dB. After the gain setting, there is a freely adjustable bank of five filters, in which you can choose between bell filters, various high and low passes with up to 48 dB/oct. slope, shelving filters and notches, as well as a loudness function.

The next module is an Acoustic Echo Canceller (AEC). This is always required when communicating with other rooms in which loudspeakers and microphones are located. If the signal from the far end room is reproduced via loudspeakers and picked up again by the microphones, it returns to the far end room as an annoying echo. This effect is known to most people when a person calls with a hands-free device and the echo canceller does not work well there. You then hear yourself later as an echo, loudly, which can be very annoying. To do this, the Echo Canceller compares the incoming signal from the far end room with the captured microphone signals, which also contain this signal in the case of loudspeaker playback. The echo canceller adapts to this and eliminates these signal components by subtracting them. If this is successful on both sides, then you can communicate bi-directionally hands-free in both directions without interference and without echoes.

Next in the Signal path you find two dynamic processors, which can be used as a compressor, as a limiter, as a noise gate or as an Automatic Gain Control (AGC). The combination of two dynamic processors makes a lot of sense, as it allows you to achieve good leveling of the input signals. Different distances to the microphone and speaking volume can be compensated, which is particularly important for conferences with far end rooms.

The following delay is a standard function, but it is still worth taking a quick look. The setting can be made with superior flexibility in samples, milliseconds, feet, or meters and is also linked to a setting for the air temperature, with which the change in the speed of sound propagation depending on the temperature can be considered.



Network interfaces of the AT V variants with a primary and secondary Dante interface, both are equipped with a switch. Below are two Ethernet ports and a USB audio interface for a maximum of eight configurable channels. On the far left with another RJ45 socket is a DMP extension connection. In between, with 3.5 mm screw terminals, an RS-232 interface and the connection for external control panels.

FIG. 03: Module for signal processing for the analog input channels and building blocks. The inputs (Flexinputs) 1–12 can either be assigned as analog inputs or switched to a Dante channel. This is particularly useful when processing Dante microphones

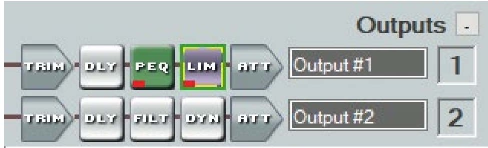


FIG. 05: Modules for signal processing in the analog output channels and expansion outputs

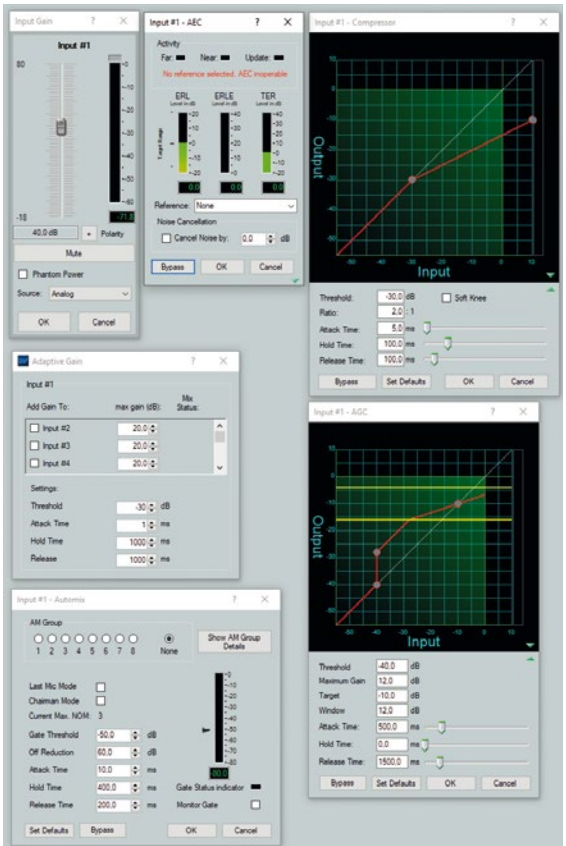


FIG. 04: User interfaces for the modules in the input channels

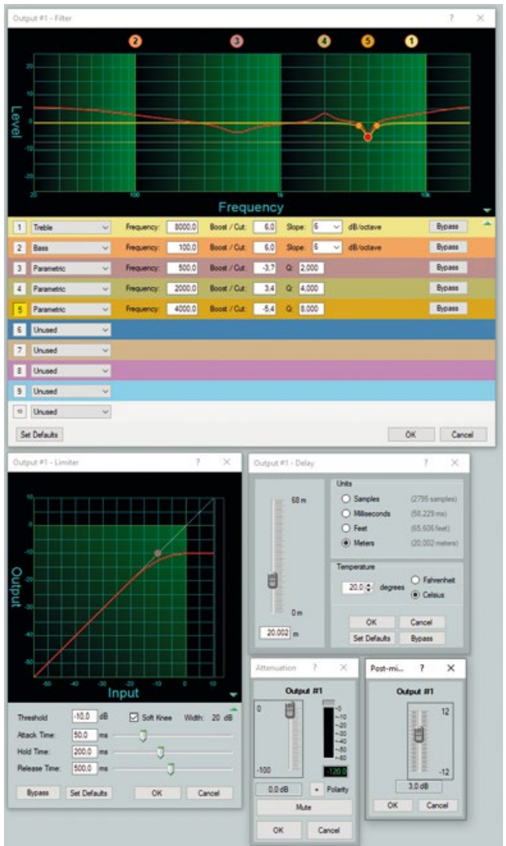


FIG. 06: User interfaces for the modules in the output channels

The delay block is followed by a ducker that can be used to reduce the signal in one path when other inputs are receiving a signal. For example, an announcement on an input can ensure that the level of other feed paths for background music are reduced by a certain value or switched off completely. As an alternative to the ducker, an adaptive gain can also be activated in this function block, where the gain is adjusted via the signal level on another input, e.g., from an interference level microphone.

The last block before the pre-mix gain trim in the input signal chain is the automatic mixer (AM). All external input paths can be assigned to one of eight AM groups in the automatic mixer. Here, the gain is adjusted depending on the open microphones (in one of two operating modes, Gain Sharing or Gated, depending on the active microphone, so that there is no feedback even with many open microphones. The opening and closing of a microphone path are done via noise gates, so that no noise is captured by unused microphones.

The Aux Inputs differ slightly from the input paths described above. The acoustic echo canceller function and the delay are not available here. In addition to the filter block and a dynamic function as well as a delay, the virtual returns are also equipped with a feedback sup- ➔

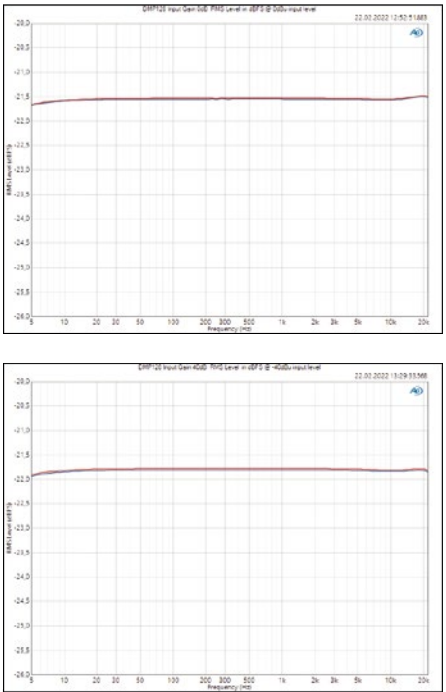


FIG. 07: Frequency response of the preamps and ADCS at 0 dB gain (above) (TOP) and at +40 dB gain (below). The input sensitivity at 0 dB gain is at +21.5 dBu for 0 dBfs on the digital side

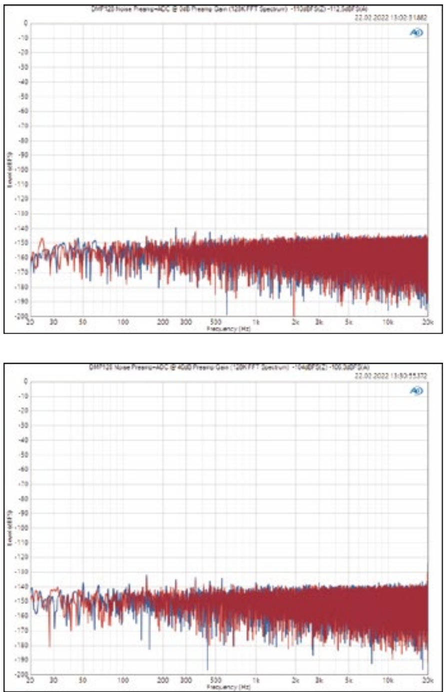


FIG. 08: Noise at 0 dB gain (above -110 dBfs) and at 40 dB gain (below -104 dBfs)

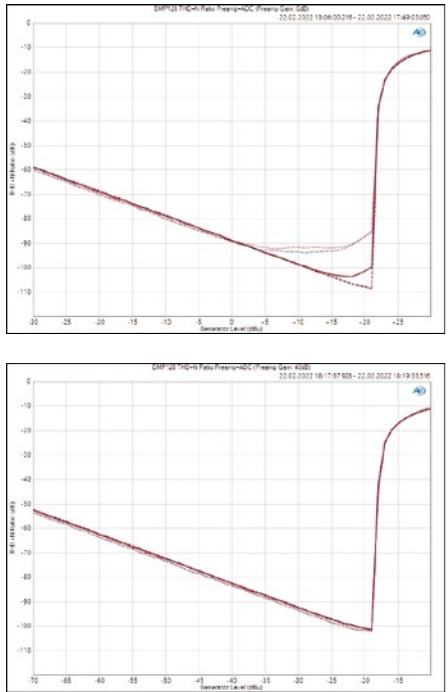


Fig. 09: THD+N depending on the input level (x-axis) at 0 dB gain (top) and 40 dB gain (bottom). Measurements at 100 Hz(– – –), 1 kHz( ) and 6.3 kHz(···).

pressor (FBS) that works with five fixed and 15 variable notch filters. The FBS is only available in virtual returns 1 to 4. The expansion inputs 1 to 48 only have the option of feeding into the automatic mixer.

Turning to the outputs, you find filtering and dynamics on all paths, as well as delays except for the AUX outputs. Complete processing for loudspeakers can also be easily configured with these functions.

Image 06: The DMP 128 Plus is built with a large base board and various plug-on modules. The separate power supply unit is on the bottom right.

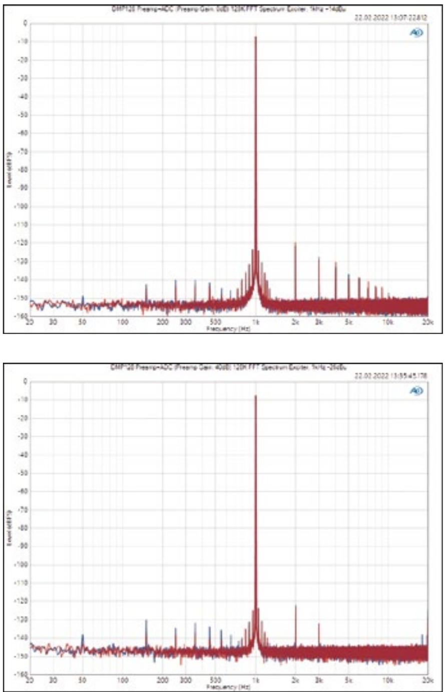


Fig. 10: FFT spectrum at –8 dBfs modulation with 0 dB gain (above) and 40 dB gain (below). In both cases, the distortion values are very low.

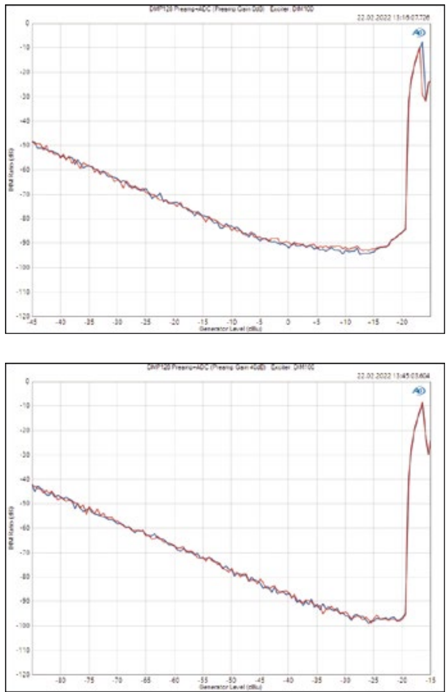


Fig. 11: Transient intermodulation distortion (DIM) at 0 dB gain (top) and 40 dB gain (bottom).

In summary, the DMP 128 Plus offers a huge range of functions around the all-encompassing matrix, with which a wide variety of tasks can be solved. Often, devices with such a wide range of functions run the risk of becoming confusing to operate. That is not the case here. The Extron DSP Configurator Software displays the entire device with all inputs and outputs and their functions so clearly that a look at the well-made manual is only rarely necessary, if at all.

Extensions

In addition to the DMP 128 Plus devices, there is a range of control elements that are available in various formats for wall installation. The ACP series control panels include modules with 5 buttons, 6 buttons with volume controls and indicators, control panel with rotary knob and mute button and the combined control panel ACP 100. The control panels are connected to the matrix processor via the ACP port and configured via the Tools menu of the DSP Configurator software. The control panels are also powered via the ACP port.

Audio Quality

A device like the DMP 128 Plus Digital Matrix Processor is usually at the center of a conference or public address system. All analog input signals, including the sensitive microphone signals, must pass through the preamps and AD converters, and all analog output signals through the DA converters and output stages. These two points are therefore crucial for the audio quality of the device and thus also for the entire system.

The primary requirement is to be able to transmit high signal dynamics, on one hand, without noise or other audible interference in quiet passages and, on the other hand, distorted or limited loud signal peaks. In extreme cases, this may require mapping a signal dynamic range of 100 dB or more with headroom at the upper and lower end. If you consider that a few dB loss can quickly occur in a signal chain due to possible mismatches and necessary headroom, then input and output stages should offer at least a signal-to-noise ratio (S/N) of 110 dB or more. However, the standard, be it for power amplifiers, controllers, breakout boxes or mixers, is a few dB below this, which is sufficient for most applications. If you look at more demanding tasks in theatres, concert halls or clubs, then that may no longer be enough. In addition to the microphone preamp, special attention is also given to the output stages, as these are located directly before the power amps in the signal chain, where there is no downstream fader.

The following measurements were therefore carried out for the DMP 128 Plus in our test: Frequency res- ➔

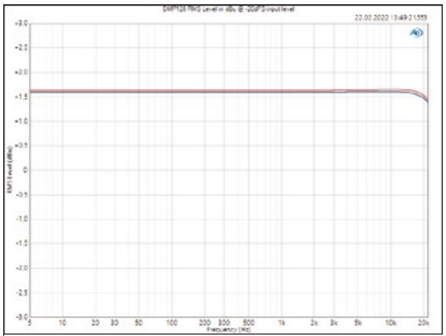


FIG. 12: Frequency response of the DACs and output stages at –20 dBfs corresponding to an analog output level of +1.6 dBu .

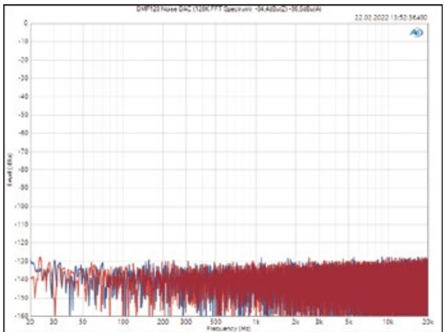


FIG. 13: Noise at the analog outputs with an overall level of –94.4 dBu and –96.6 dBu (A).

ponse, noise level, THD+N and TIM for the analog inputs with preamp at minimum and maximum gain as well as for the analog outputs. The measurements were carried out with the measurement signal fed via the Dante network. All measurements were performed with a sampling rate of 48 kHz. The audio analyzer used for this is an APx 555 from Audio Precision.

Preamps and AD converters

The measured frequency responses (FIG. 07) run at minimum and at maximum gain of the preamps from 20 Hz to 20 kHz with a fluctuation range of less than ±0.1dB. For the analog inputs, the S/N depends on the gain setting of the preamp. At 0dB gain (FIG. 08 above) an S/N of 110dB unweighted and 112.5dB dB-A weighted is achieved. At maximum analog gain of 40dB (FIG. 08 below), the A-weighted S/N is 106.3dB with a maximum-scale input sensitivity of 18dBu, which results in a calculated equivalent input noise (EIN) of a good -124.3dBu (20-20k, A-weighted). TAB. 01 provides an overview of the preamp Gain setting and the associated values for the S/N and for full-scale.

The analog gain adjustment range extends from 0 to +40 dB. A digital gain is added for all values above and below, which means that the clip limit at the upper end or the background noise at the lower end of the usable control range can't be shifted further. However, a real gain setting of 0 to +40 dB with a clip limit of +22 dBu or -18 dBu can be considered completely sufficient in practice. In terms of S/N, the DMP 128 Plus meets all expectations. These values are achieved on the input side with an eight-channel Cirrus CS5368 AD converter and upstream TI 1612 and LM833 OPVs.

Gain [dB]			Max.In.	Noise [dBfs]	
total	analogous	digital	[dBu]	lin.	A review
-18	0	-18	22	-126.7	-128.9
-10	0	-10	22	-118.9	-120.9
0	0	0	22	-110.5	-112.6
10	10	0	12	-110.5	-112.6
20	20	0	2	-109.2	-111.3
30	30	0	-8	-109.2	-111.3
40	40	0	-18	-104.0	-106.3
50	40	10	-18	-94.0	-96.3
60	40	20	-18	-84.0	-86.3
70	40	30	-18	-74.0	-76.3
80	40	40	-18	-64.0	-66.3

Table: Noise and maximum input level depending on the gain values. The ON is -124.3 dBu (A)

On the topic of distortion of the analog inputs, THD measurements were carried out at minimum (0 dB) and at maximum (+40 dB) gain for frequencies of 100 Hz, 1 kHz, and 6.3 kHz. For this measurement, the signal was fed into the analog inputs and routed output via the Dante outputs without using internal signal processing and back to the measuring system via the DVS (Dante Virtual Soundcard) on the APx's PC. At 0 dB gain, the results of the THD+N curves from FIG. 09 above show a very good progression. The THD value falls below the -100 dB line at the minimum directly before the clip limit. It is at 6.3

Certified for Microsoft Teams Rooms

Extron recently announced that several of its digital audio processors, amplifiers, and speakers are now certified for Microsoft Teams Rooms, including the DMP 128 Plus digital matrix processor. The products that are certified for Teams Rooms have been thoroughly tested for performance and usability. In addition to the DMP 128 Plus series, the following other Extron products also have this Microsoft certification: DMP 64 Plus series digital matrix processor, XPA U 1004 SB and XPA U 1002 amplifiers, SF 26CT series as well as SF 3CT LP and SF 3C LP ceiling loudspeakers.

kHz where the curve runs a little higher, but still below the -90dB line. Somewhat surprisingly and even better, the values drop at maximum gain (FIG. 09 below) of +40 dB, which may be related to the internal gain structure of the preamp.

In addition to the total value of the distortion components, their spectral composition also counts. In ABB. 10 the FFT spectra of the 1 kHz measurement are each measured 8 dB below the clip limit. Irrespective of the fact that the distortions are very low overall, their composition with k 2 as the highest line and higher-order harmonics that fall evenly and rapidly, is also exemplary. Such an FFT spectrum corresponds to what is commonly referred to as optimal in terms of sound.

In contrast to the THD measurement, which is carried out with a sine wave signal in the steady state, a mixed signal consisting of a 3.15 kHz square and a 15 kHz sine wave is used for the subsequent measurement of the transient intermodulation distortion (TIM). This type of signal with the square steep edges, have significantly higher demands on the electronics and shows their behavior with rapidly changing transient signals. The TIM measurement is therefore said to be particularly relevant to the tonal qualities of a device. FIG. 11 shows the associated measurement, again for 0 dB (above) and for +40 dB (below) gain, where the curves in large areas run at a very good -90 dB and well below. The results at maximum gain are slightly better than at 0 dB.

DA Converter and Output Stages

The last measurements still apply to the DA converters and the subsequent output stages. A CS4398 chip from Cirrus is also used here as a DA converter with NE5532 operational amplifiers in the subsequent output stage.

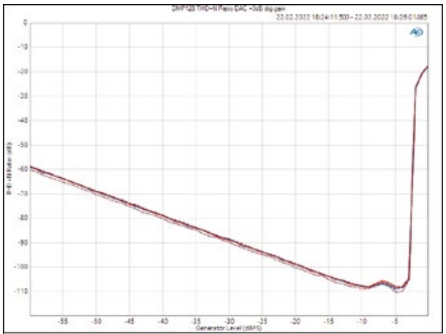


FIG. 14: THD+N as a function of the digital output level in dBfs (x-axis). A digital gain of +3 dB was set for a better representation of the clip limit. Measurements at 100 Hz(– – –), 1 kHz( ) and 6.3 kHz(···).

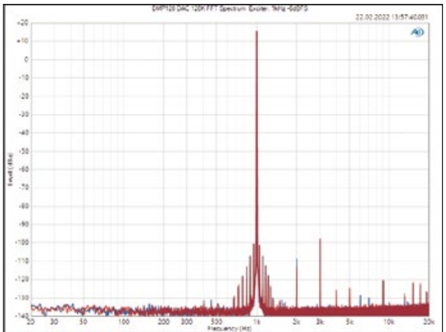


FIG. 15: FFT spectrum at –6 dBfs modulation on the digital side and +15.6 dBu output level.

The internal resistance of the balanced outputs is 100 Ω. Of course, the frequency response here is also perfectly straight, as if drawn with a ruler (FIG. 12) . The measurement was made at -20 dBfs on the digital side, resulting in a level of 1.6 dBu at the analog output. The maximum output level is therefore 21.6 dBu. The measured value for the S/N on the analog outputs (FIG. 13) is a very good 116 dB unweighted and 118 dB A-weighted.

The THD+N measurements at 100 Hz, 1 kHz and 6.3 kHz from ABB. 14 also paint a very good picture with values from almost -110 dB to just before the clip limit for all three frequencies. In this case, the FFT spectrum for 1 kHz at -6 dBfs shows a little more k3 than k2, but at such a low level overall that there is probably no need to discuss whether this is good or bad. The k3 line is at -99 dB compared to the fundamental of 1 kHz at +16 dB. The k3 distortion component is therefore -115 dB (=0.00018%).

Prices and Models

As explained before, the DMP 128 Plus is available in different variants, so that a model with more or fewer features can be selected depending on the requirements.

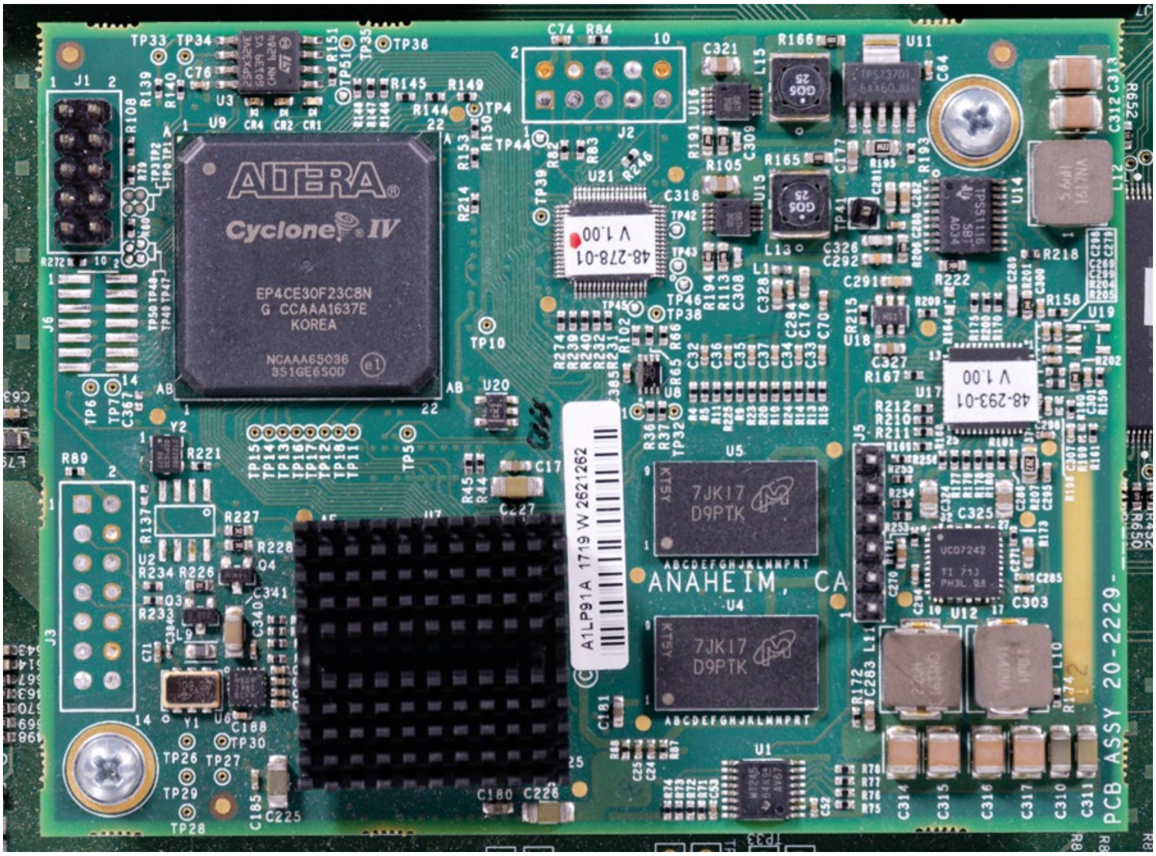


Figure 10: The heart of the DMP 128 Plus. The ProDSP board is populated with Altera FPGAs

The DMP Plus 128 processors can also be easily integrated with a control system.

DMP 128 Plus (basic device)
DMP 128 Plus AT (with Dante Interface)
DMP 128 Plus C (with AEC)
DMP 128 Plus C AT (with AEC and Dante)
DMP 128 Plus C V (with AEC and VoIP)
DMP 128 Plus C V AT (with AEC, VoIP, and Dante)

AEC = Acoustic Echo Cancellation
VoIP = Voice over IP connections
AT = Dante / AES67

Conclusion

The Extron DMP 128 Plus Digital Matrix Processor is a well-equipped device. In its basic model with 12 analog inputs and eight analog outputs, as well as a USB audio interface and a DMP expansion connection, it is ideal for smaller conference systems, and offers more than comprehensive functionality. If you add the optional features with the Dante interface, acoustic echo cancellation and a VoIP interface, the DMP 128 Plus is also ideally equipped for larger installations, either individually or in combination with other devices. As expected from Extron, the audio measurements of the analog inputs and outputs are consistently convincing. The very clear and intuitive DSP Configurator software and its ready-made templates for typical tasks, which make it easy for newcomers to get started, are also a great success. It should also be mentioned that other DMP Plus models are available in different configurations, the DMP Plus 64 (as a little brother) and the DMP Flex Plus 128 with fewer analogue inputs •

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