PRODUCTION PARTNER

Fachmagazin für Veranstaltungstechnik

Review
from issue 09/2017

Digital mixer

Yamaha Rivage PM10

Yamaha can be regarded as a pioneer in the development of digital mixing consoles. The company's new premium console is the Rivage PM10: can this console again serve as a benchmark?

» ... The PM10 is Yamaha's flagship digital mixing console. ...«

Digital mixer

Yamaha Rivage PM10

Yamaha can be regarded as a pioneer in the development of digital mixing consoles. The company's new premium console is the Rivage PM10: can this console again serve as a benchmark?

Copy, images and measurements: Anselm Goertz | Drawing: Yamaha

Yamaha's mixing consoles can be found in virtually every rental company's inventory – and these digital consoles have had a decisive influence on event technology. What was missing for some time was a prestigious "top model", which Yamaha now offers with the PM10. The "Rivage PM10 mixing system" is the flagship amongst Yamaha's mixing consoles and – as is common practice today – is based on three components: console, DSP engine and I/O rack. The connection between these components is based on two ring-shaped networks.

Networked mixer structure

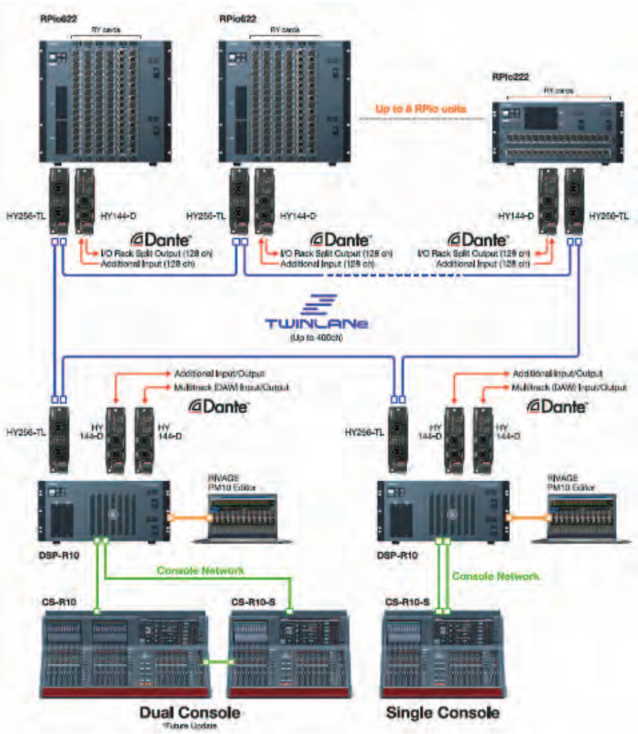
The simplest and also smallest configuration consists of a DSP engine as the core of the system, a console connected via network cable and an I/O rack connected via the optical TWINLANe network. Both types of network are designed as a ring and thus – with only two devices to be connected – consist of one outgoing and one return line. The ring structure ensures redundancy. Stand-alone operation of the console by itself is not possible, since this is designed as a pure control unit with a few inputs and outputs for peripheral tasks while the entire signal processing takes place in the DSP engine.

Fig. 1 shows a somewhat more complex system with three consoles, two DSP units and three I/O racks. The DSP engines and the three I/O racks are connected via the blue TWINLANe fibre optic ring. The consoles are connected to the DSP engines via standard network cables, the Console Network cables. Not available yet in the current version, but planned in the future, is the operation of two consoles using the DSP engine, which are then connected via the console network ring using cables. The cable length can be up to 100 m. This can be used as an extension or as a mirror to backup the user interface. If two consoles are used at separate locations, for example FOH and monitor, where only one TWINLANe connection is available, they also require a local DSP engine respectively. Within a TWINLANe ring, a

I/O rack RPIO622 *with six RY card slots and two slots for MY cards on the front panel*

maximum of two DSP engines and eight I/O racks can be integrated in the current version. The TWINLANe's capacity is 400 channels with a 96 kHz sample rate and 32-bit resolution. The console network transports a maximum of 109 channels. Also as a future option, the mirroring of the DSP engine is envisaged, which is of course only logical if one already wants to build a redundant system using only two consoles.

The well-known Yamaha MY card format was no longer suitable for handling the large number of channels. For this reason, the new HY card format was developed for the Rivage system, transmitting up to 256 channels bi-directionally. Nevertheless, all Rivage system devices also have two MY card slots each, in which all cards from the previous consoles can be inserted. For the new HY format, the TWINLANe card is available in two versions for multi-mode (up to max. 300 m)



PM10 Rivage system *in a large sample configuration with three consoles, two DSP units and three I/O racks. The core system is networked via a Yamaha TWINLANe ring. Connections to external devices with audio signals are made via Dante network. The consoles are connected to the DSP engines via the console network, also as a ring. (Fig. 1)*

and single-mode (up to max. 2 km) fibres as well as a Dante card. The Dante card can process up to 144 channels bi-directionally at 96 kHz, thereby also allowing connections to larger Dante networks. The DSP engine has four HY card slots, while the I/O racks have two.

In addition to the proprietary networks, all devices are also equipped with classic Ethernet network connections. If the system is thus integrated into a PC network, a notebook or tablet with Rivage PM10 Editor software can additionally be used for operation.

Modular I/O boxes

In addition to many other innovations, two new I/O racks and new slot cards have been developed for the PM10 system. The racks come with the RPio622 and RPio222. The type designation also directly indicates which slots are available in which number. The type designation “622” thus stands for six RY slots, two HY slots and two MY slots. The smaller RPio222 I/O rack has two RY slots. For the RY slots, cards are available with 16 analogue inputs, 16 analogue outputs or with 16 digital inputs and outputs in the AES/EBU format with SRC (Sample Rate Converter) respectively. The cards can be used in the I/O racks in any combination. It is not possible to change this combination during operation. If one wants to integrate additional devices or recording equipment on site, this can be done using an HY-Dante card or an MY-

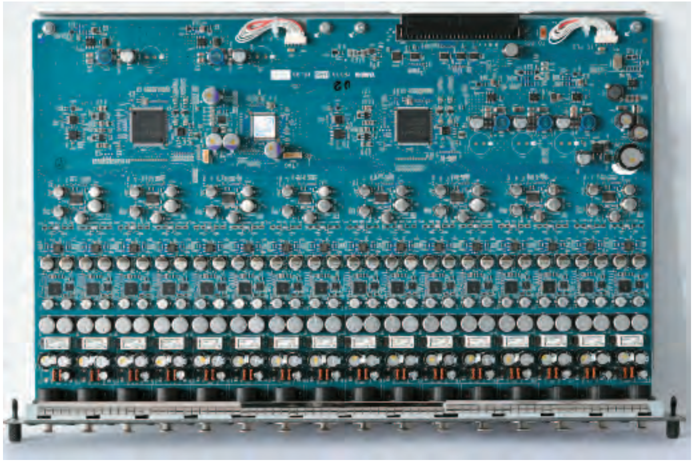
DSP engine *from the front and rear. On the front side, there are two slots for cards in the familiar MY format. Up to four HY cards for TWINLANe or Dante cards can be used on the back. The HY cards can transmit up to 256 (TWINLANe) or 144 (Dante) channels. The DSP engine is of course equipped with two power supplies*

I/O rack RPio622 *from the rear with two HY slots; also here one will find the obligatory two power supplies*

Dante card – depending on the desired number of channels. However, the MY cards are limited to a maximum of 16 inputs and outputs.

ADC and DAC as interfaces to the analogue world

Microphones and loudspeakers continue to operate in the analogue world. For this reason, every digital device, network or mixer system inevitably requires an appropriate A/D and D/A converter (usually referred to as a "transducer", which is theoretically not quite correct: ultimately, one sticks to electrical signals. A microphone or loudspeaker, on the other hand, are real transducers that convert acoustic or mechanical energy into electrical energy or vice versa). There is no question, however, that the A/D and D/A converters are of crucial importance. It is not without reason that external high-quality A/D converters with appropriate pre-amps are used in recording studios for special, particularly sensitive sources. Since the majority of a console's signals are of an analogue nature in live applications, the A/D converters play an important role: if quality is lost here, it is irretrievably gone. The other side of the signal chain with the D/A converters is not so critical. The components for D/A conversion have all been available in very good quality for a long time and the subsequent analogue circuitry only has to deliver a



RY16-ML-SILK *card with 16 analogue inputs with mic pre-amps and Silk processor*

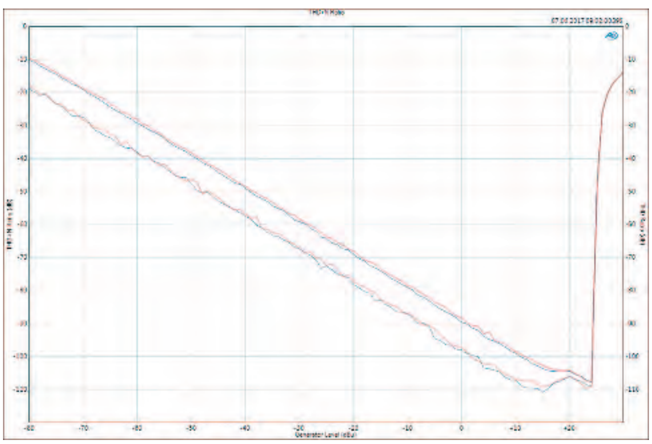
sufficiently high signal level with balanced stability. There are still a lot of mistakes to be made here, but the task is much easier than developing a good remote-controlled pre-amp for the A/D converter. Additionally, in live applications, the components following the console, such as system controllers or power amplifiers, often already have digital inputs in AES/EBU format or with Dante and the analogue outputs thus become obsolete. Meaningful measurements of pre-amps, ADCs and DACs are therefore an important part of every PRODUCTION PARTNER mixer test.

To the extent possible, we record the signal-to-noise ratio, the harmonic distortions (THD) depending on the level and frequency, the distortion spectrum and the transient distortions (DIM or TIM). For the A/D converter with pre-amplifier, measurements are taken once at low gain for typical line level signals and once at high gain for microphone signals.

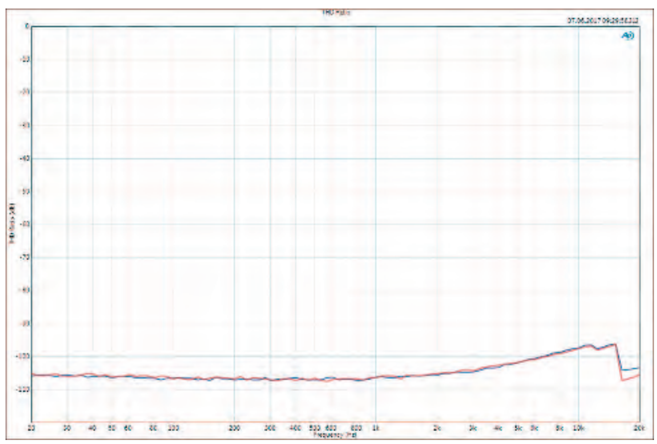
Preamp Gain	max. input	Noise
[dB]	[dBu]	[dBfs]
–6	+30	–115
0	**+24**	**–113**
6	+18	–111
12	+12	–115
24	0	–114
36	–12	–111
48	–24	–101
60	–36	–90
66	–42	–84

Table of the gain values *for the pre-amp with maximum input level and the noise level to be measured on the digital side in dBfs (Tab. 1)*

Table 1 initially shows the gain settings that are possible in a very wide range from -6 to +66 dB in 1 dB steps. For 0 dB



THD and THD+N *from pre-amp and ADC at 1 kHz and 0 dB gain depending on input level. The clip limit is +24 dBu. The solid line shows the THD+N values; the dashed curve shows THD only (Fig. 2)*



THD *of pre-amp and ADC at 1 kHz and 0 dB gain depending on the frequency at -6 dBfs modulation corresponding to +18 dBu analogue input level (Fig. 3)* gain, full output is achieved on the digital side (0 dBfs) at +24 dBu. With gain values of up to approximately 30 dB, the ADC dominates the signal-to-noise ratio and amounts to approximately 114 dB. If the gain in the pre-amp continues to increase, the pre-amp's signal-to-noise ratio begins to have a significant influence on the value. For the extreme value of 66 dB gain, this is an S/N of 84 dB. Together with the -42 dBu gain limit, this results in an equivalent input noise of -126 dBu. With A-weighting, the noise level is 3 dB lower. The values correspond to the information given in the data sheet.

The curves in Fig. 2 show the THD and THD+N values depending on the input level. Only the harmonic distortions are taken into account for the THD measurement, while all components not belonging to the excitation signal and thus also the noise (+N) were considered for THD+N. The THD-only values are thus always slightly better. In Fig. 2, the THD values reach a minimum of very good -110 dB and remain below -105 dB even until the clip limit is reached at +24 dBu. The measurement was carried out at 1 kHz. The behaviour of the pre-amp and ADC at other frequencies is shown in Fig. 3, where a constant level of +18 dBu with variable frequency was measured. At 1 kHz, the -107 dB from Fig. 2 can be found again. For lower frequencies, the value remains largely constant and there is a slight rise to -98 dB at 10 kHz at high frequencies. The jumps above 10 kHz are caused by the FFT-based measurement method with a 96 kHz sample rate, where the k_3 component falls out of the measuring range above 16 kHz. The basic increase in THD values towards the high frequencies is inevitably caused by the declining feedback in the analogue circuitry.

Even if values of -100 dB and less are considered to be negligible distortions, a quick look at the distortion spectrum in Fig. 4 should not be missing. The spectral lines shown here refer to 0 dBfs. The basic wave at 1 kHz is -6 dBfs and the maximum harmonic k_3 at -113 dBfs is thus -107 dB. Other harmonic distortions are even less pronounced. The slight jitter bell around the fundamental wave at 1 kHz may have been caused by the SRC in the signal way.

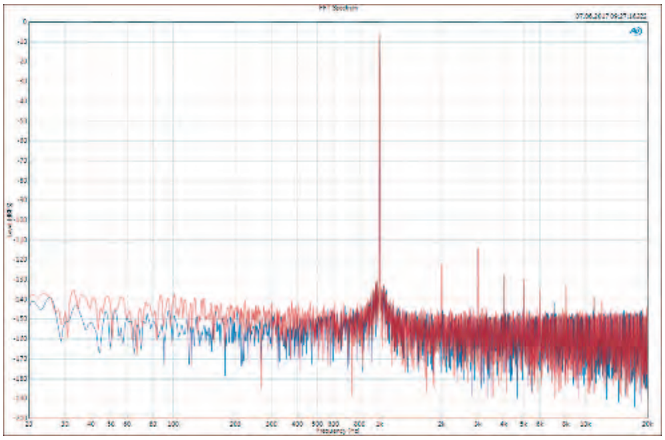
The last measurement in this setting concerns the dynamic intermodulation distortions (DIM). Also here, Fig. 5 shows the values depending on the analogue input level. A minimum of -100 dB and -92 dB directly at the clip limit are excellent results.

An identical series of measurements was then carried out at a very high gain of +60 dB in the pre-amp. In Fig. 6, the clip

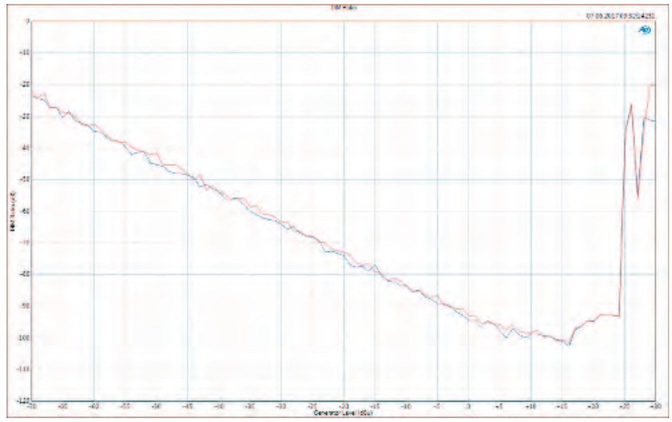
30 Years of Yamaha Mixers

Yamaha has been one of the world's market leaders in professional mixing consoles for over 40 years. The PM series' analogue consoles were milestones in Yamaha's history, as were Yamaha's digital consoles, which have been available for 30 years. The first – of course, still analogue – PM console was launched in 1972 and commenced a series that has been in use ever since and which has just found its new flagship product in the PM10. The forerunner of all PM consoles – the PM200 – was followed by a number of smaller models as keyboard mixers or for the small combo, until the still highly regarded PM1000 was introduced in 1974 as the first large modular FOH console. Further large live consoles followed with the PM2000, 3000 and 4000. The PM4000 in particular has become a standard on large stages alongside well-known English brands. In 2003, and thus already well into the digital age, the analogue PM series was once again crowned with the PM5000. Already two years before, the PM1D was presented as the first large and fully digital live console. The digital origin was the DMP7 compact keyboard mixer. In 1991, the DMC1000 was the first digital studio console to be introduced – a console, which is still in use today and is appreciated for its good audio quality. The 02R, which was launched in 1995, could be described as the breakthrough on the broad market. In its modernized form as the 02R96 it is still, together with the DM1000 and DM2000, included in Yamaha's current portfolio and is still used by many studios worldwide.

The PM series was extended in 2004 with the slightly more compact PM5D. After that, it took twelve long years until a new "big" live console was presented: the PM10 Rivage in 2016. The "D" in the type designation is omitted nowadays, since a special mention of the fact that this is a digital console is no longer needed. To simply call it a mixing console, doesn't actually fit anymore: if one takes a closer look at the PM10's scope and features, "mixing system" is probably the more appropriate term. The PM series' history thus continues and we look to the future with great interest. The current live consoles can be found in the TF, QL and CL series.



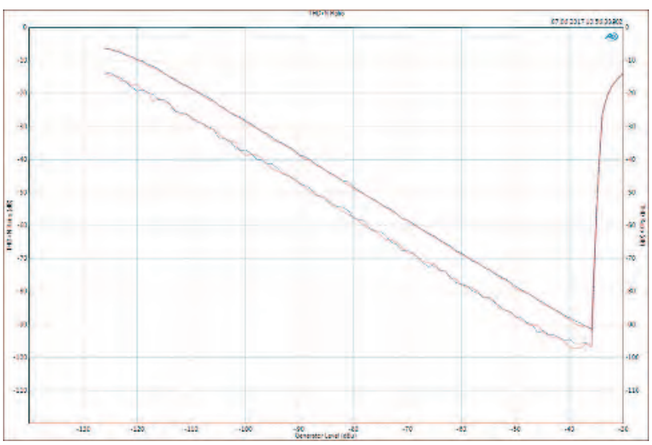
Distortion spectrum *of pre-amp and ADC at 1 kHz and 0 dB gain. 6 dBfs modulation according to +18 dBu analogue input level (Fig. 4)*



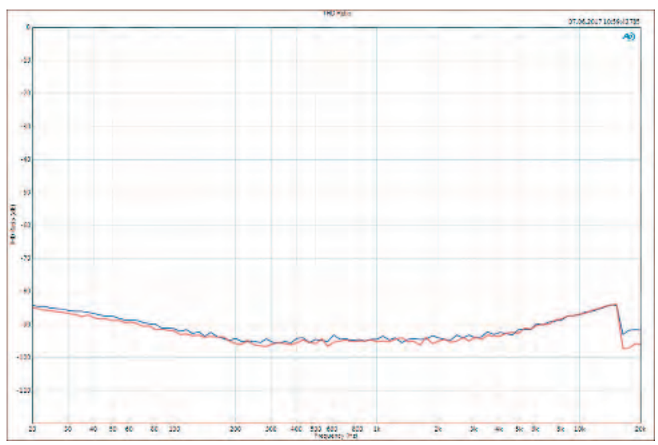
DIM *(dynamic intermodulation distortion) of pre-amp and ADC at 0 dB gain as a function of input level (Fig. 5)*

limit is now reached at -36 dBu on the analogue side. As expected, the THD and DIM values are slightly poorer than expected, but can still be described as very good. The THD value at the clip limit is -97 dB and the DIM value is -90 dB. Both values would be considered very good even without the 60 dB gain in the pre-amp. The clean distortion spectrum in Fig. 8, with only k_2 and k_3 components, confirms the good overall result as does the THD measurement as a function of frequency in Fig. 7.

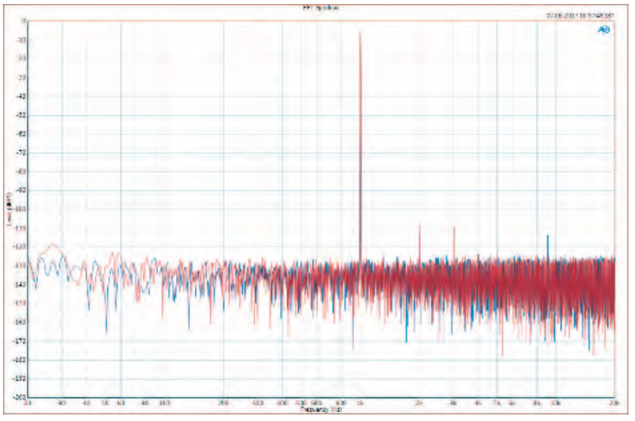
For the circuit technicians among the readers, it should also be mentioned that Cirrus CS5381 was used for the ADCs and that the remote pre-amp was constructed with a THAT 1570 amplifier module and a corresponding THAT 5171 controller



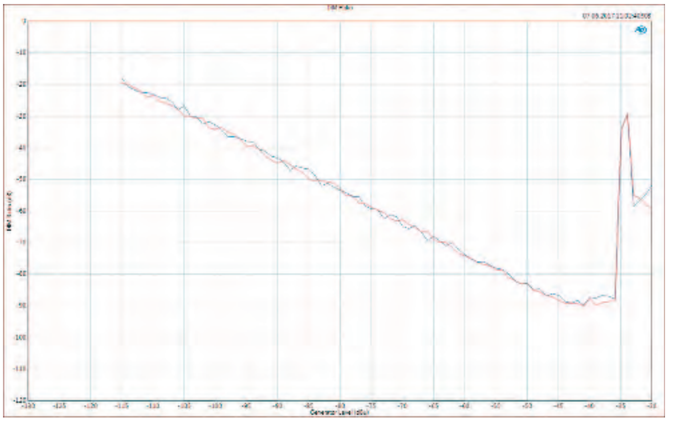
THD and THD+N *from pre-amp and ADC at 1 kHz and 60 dB gain depending on input level. The clip limit is now -36 dBu. The solid line shows the THD+N values; the dashed curve shows THD only (Fig. 6)*



THD *of pre-amp and ADC at 1 kHz and 60 dB gain depending on the frequency at -6 dBfs output level corresponding to -42 dBu analogue input level (Fig. 7)*



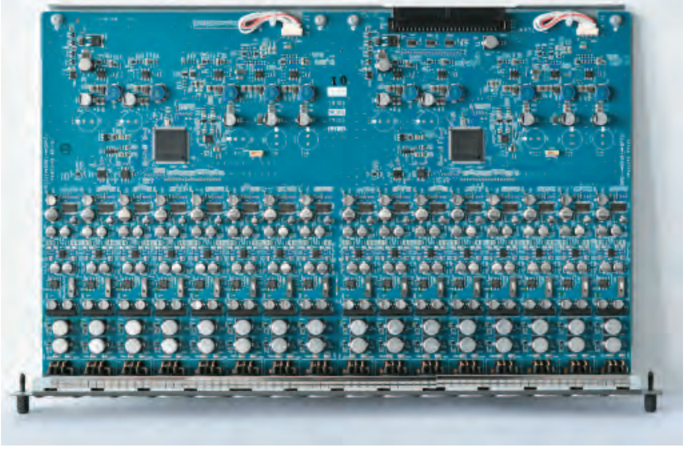
Distortion spectrum *of pre-amp and ADC at 1 kHz and 60 dB gain. Output level -6 dBfs corresponding to -42 dBu analogue input level (Fig. 8)*



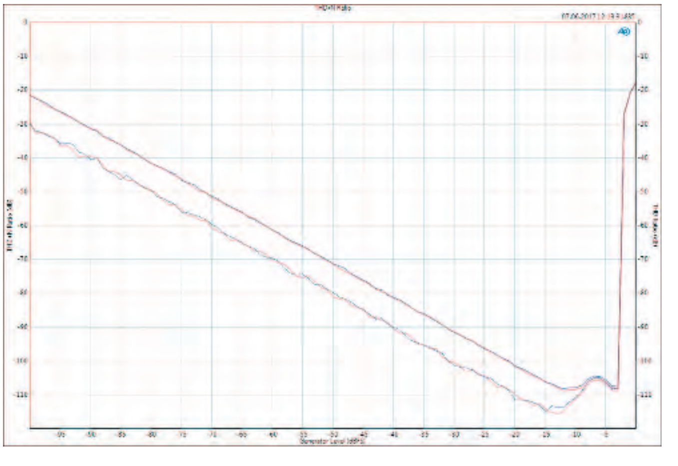
DIM *(dynamic intermodulation distortion) of pre-amp and ADC at 60 dB gain as a function of input level (Fig. 9)*

module. In addition, a relay-switched passive 18 dB attenuator was also used. The RY16-ML-SILK card's special Silk processor is discussed in a separate topic box.

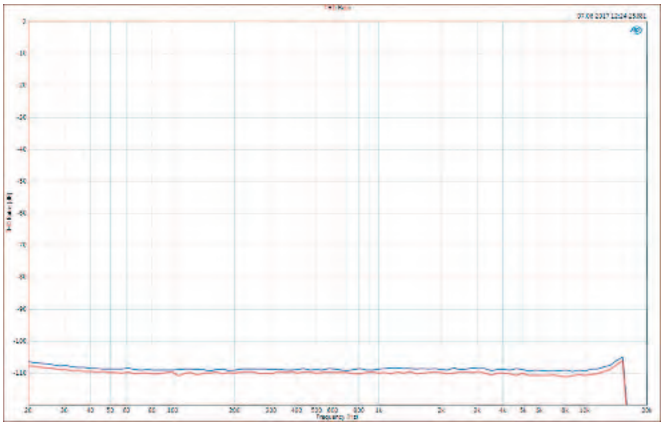
On the output side, the DACs with the following analogue output stages take over the signal transmission. Here too, we will take a quick look at the hardware. CS 4398 from Cirrus are used as D/A converters, while a two-channel DAC is used for each output. The S/N can be improved by 3 dB. Small switches on the circuit board allow the maximum output voltage to be set to 15, 18 or 24 dBu. The measured values were slightly higher. In the +24 dBu setting, the maximum output level was +25.2 dBu. The measured noise level at the analogue outputs was -93 dBu or -95 dBu (A) respec-



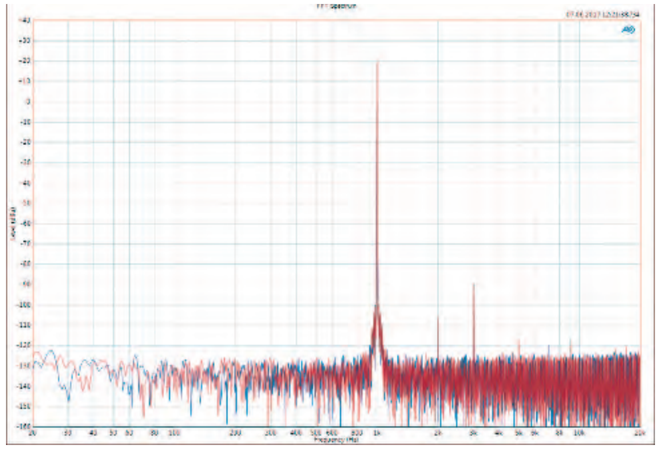
RY16-DA *card with 16 analogue outputs; with the small slide switches, the maximum output voltage can be set individually for each output and adapted for the following device*



THD and THD+N *of the DAC and the analogue output stage at 1 kHz depending on the level on the digital side. The measurement was performed with 3 dB gain on the digital side. The solid line shows the THD+N values; the dashed curve shows THD only (Fig. 10)*



THD *of the DAC and analogue output stage as a function of the frequency at +19 dBu output level (Fig. 11)*



DAC and analogue output *stage distortion spectrum at 1 kHz and +19 dBu output level. The slight jitter bell at 1 kHz is probably caused by the SRC (Sample Rate Converter) in the signal path (Fig. 12)*

tively, which results in a very good S/N of 120 dB. The good S/N can then be largely maintained for subsequent devices due to the possibility of level adjustment.

For the DACs, in principle measurements comparable to the ADCs were made, however only for one level setting – in this case for +24 dBu. Fig. 10 to 12 show the THD and THD+N as a function of level, THD as a function of frequency and the distortion spectrum at 1 kHz. The latter two were each measured at 6 dB under full modulation. Values of -110 dB over the entire frequency band can, with good certainty, be described as perfect. The distortion spectrum again shows the slight jitter bell around the fundamental wave. A SRC in the signalling pathway could be the possible cause here too.

In short, the PM10 system's A/D and D/A cards have excellent audio quality. The analogue inputs offer a very wide gain range, so that values between -42 and +30 dBu are possible

PM10 system *at our measuring station*

One of three 12-channel fader bays *on the PM10 console's interface; the channel section can be seen in the top right of the picture*

Rivage PM10 Editor software *settings for a channel EQ (Fig. 13)*

for full-scale modulation. This allows adaptation to any possible source. The measured values are excellent in all settings. The same applies to the analogue outputs, which – thanks to the adjustable output levels – now also allow good adaptation to all types of subsequent devices.

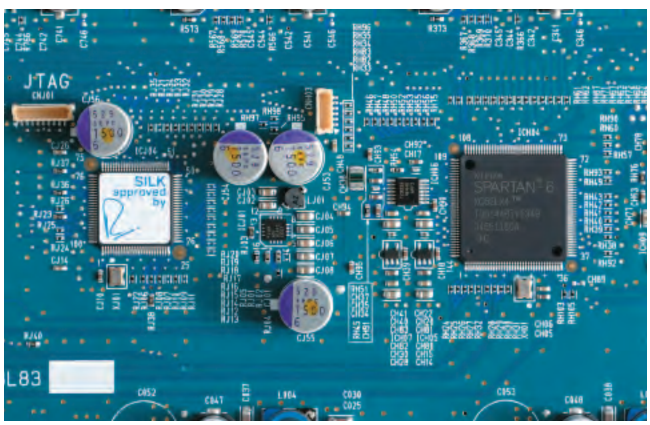
Consoles and remote

A lot of DSP power and great functions in one console are only really helpful if they are fast, safe and easy to use. This applies even more to time-critical live applications than it does to studio productions. The design of a good user interface is equally important. Large and easy-to-read screens, a not-to-high density of controls and clear structures are just as much a part of this as are a good and high-quality design of the controls and of the console as a whole. Optically and haptically, the PM10 does not disappoint. The console comes with a noble appearance and immediately gives a clearly arranged impression. Three fader bays with twelve channels each, two large 15" touch displays and the generously proportioned channel section (as is already known from the PM1D) provide a good overview. For each channel selected, the channel section contains all functions and setting options in direct access, creating the familiar feeling of having everything in view from analogue consoles.

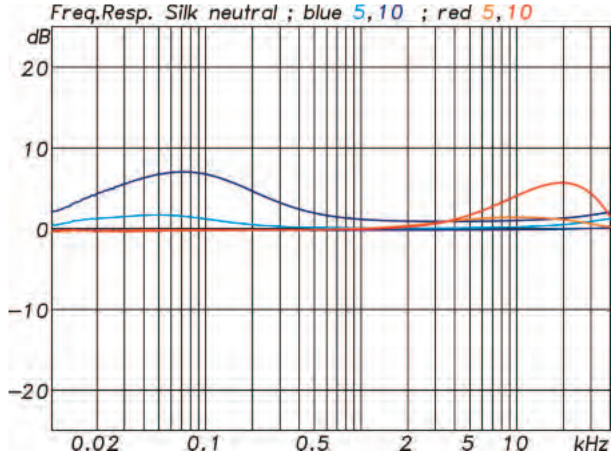
The test conditions in the laboratory are always a further yardstick for the operability of a console: this is often the first contact with the device and the required constellations for the measurements are not standard, so they have to be explicitly created. If one can do this more or less intuitively, without external help and without having to search through manuals or YouTube videos for a long time, then this is a first plus. A lot can be edited on the PM10 system's surface: 144 channels, 72 mix buses, all with rich features, a plug-in system with more than 50 well-known "devices", an auto-mixer and a large-scale scene management. It would be impossible to list everything here.

However, the many special Rupert Neve plug-ins with EQs and compressors as well as the Eventide Ultra-Harmonizer

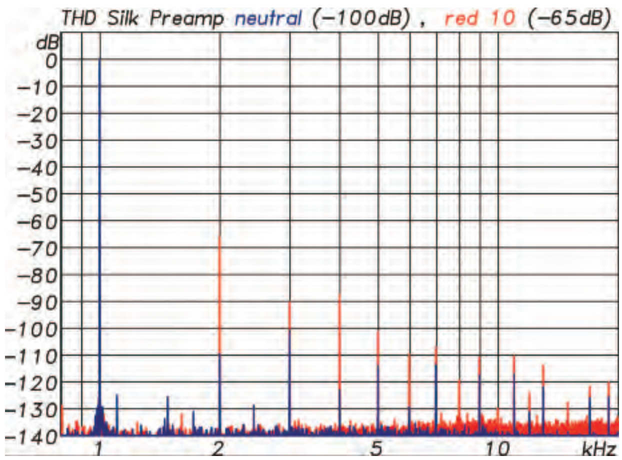
Silk setting *with the selection "blue" for low-frequency and "red" for high-frequency as well as the texture setting for the intensity of the Silk processing*



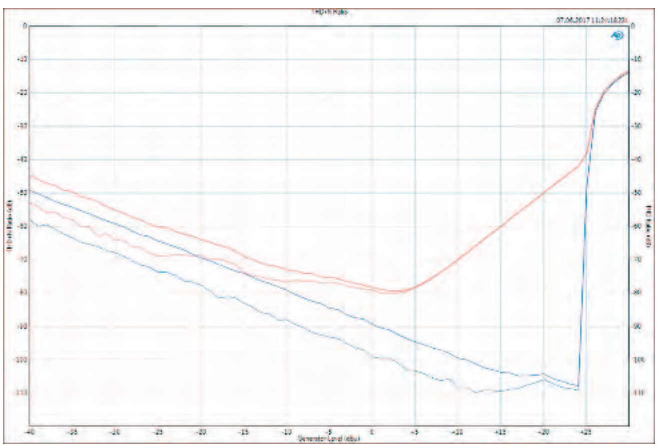
Silk processor *on the analogue input board; the Silk DSP simulates the behaviour of the analogue pre-amp with transformer developed by Rupert Neve*



Influence of Silk *on frequency response (Fig. 14)*



Distortion spectrum *without (blue) and with (red) active Silk; plenty of k_2 ensures the desired sound (Fig. 15)*



THD and THD+N *from pre-amp and ADC at 1 kHz with (red) and without (blue) Silk depending on input level. The respective solid line shows the THD+N values; the dashed curve shows THD only (Fig. 16)*

and tc electronic Hall processors should not go unmentioned. The auto-mixer design by Dan Dugan is of course legendary. The console is now available in two versions: the CS-R10, which was provided for the test, with three fader bays and two screens and a width of 1.55 m as well as the CS-R10-S with two fader bays, one screen and a width of 1.13 m. In addition to the consoles, a PC or tablet can also be used together with the editor software as a remote control or for advance configuration.

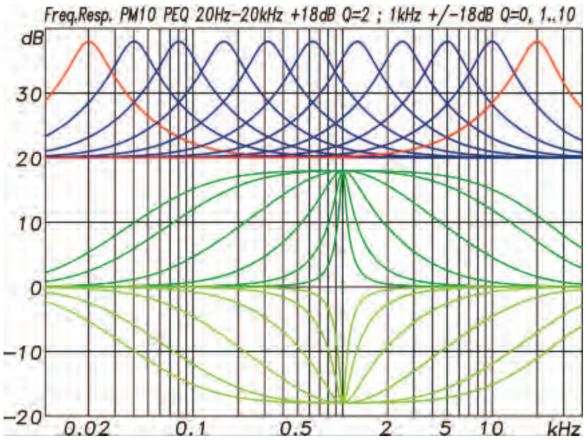
For live recording, the PM10 offers the possibility to easily record a stereo mix on a USB flash drive or to transfer the full programme with all channels via Dante card directly to a DAW. The PM10's HY Dante cards can transmit up to 144 channels. Depending on the Dante interface on the DAW with a PCIe interface or the Virtual Soundcard, 128 or 64 of these can be processed.

The practical gain compensation function is also available for processing signals from the I/O racks in several DSP engines or recording devices. This is always necessary if one source is used for several applications. For a live act, this can be the FOH mixer, the monitor mixer and maybe a recording. If the analogue pre-amp gain would be changed by the FOH, this might have undesired effects on the monitor mix and recording. The gain compensation now ensures that the I/O rack's output level remains constant at all times. If the analogue gain is increased by the FOH, this is automatically compensated by a subsequent digital gain in the I/O rack. In order to change the input level for the respective channel at the FOH, the digital gain in the local DSP engine can be used for this channel.

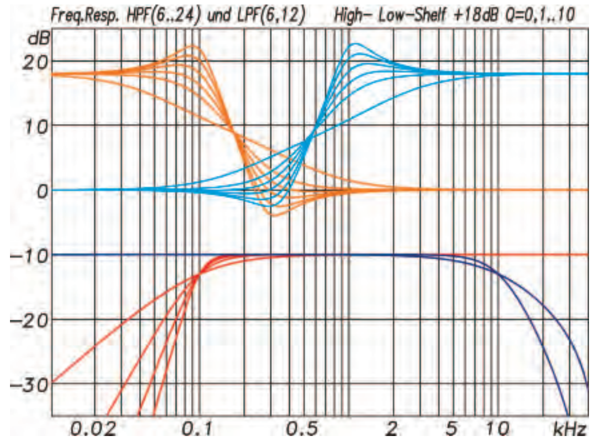
Another important function in the background is delay compensation. Depending on the channels' signal paths with or without inserts, plug-ins or EQs, the runtimes will slightly change. The delay compensation calculates the runtimes in the paths, compares them and inserts small delays if necessary, so that the runtimes remain the same regardless of the individual settings. An activated insert in the input for example causes an additional delay of 1.08 ms. Delay compensation can be activated separately for the inputs, the output bus structure and the outputs.

Silk pre-amps

As the measurement results have impressively shown, the PM10's pre-amps and ADCs process the analogue signals to perfection. Now it is possible that one may miss a historical



Bell filter *in parametric EQ; the filter curves remain unchanged in their course even at high mid-range frequency (Fig. 17)*

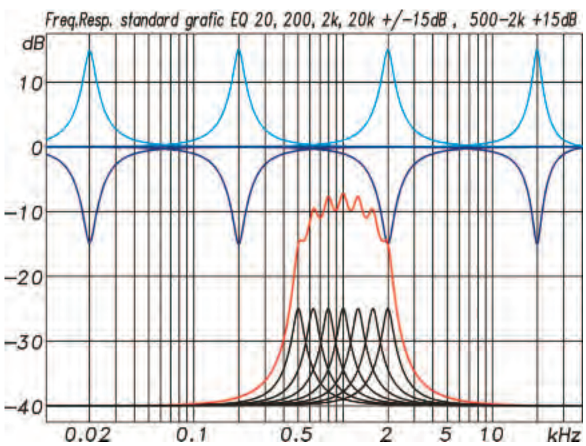


High and low pass filters *as well as shelf filters in the inputs. The high-passes can be selected between the 1st and 4th order. The low-passes are only 1st and 2nd order. The shelf filters allow a wide adjustment range for the filter quality from 0.1 to 10 and a maximum gain of ±18 dB (Fig. 18)*

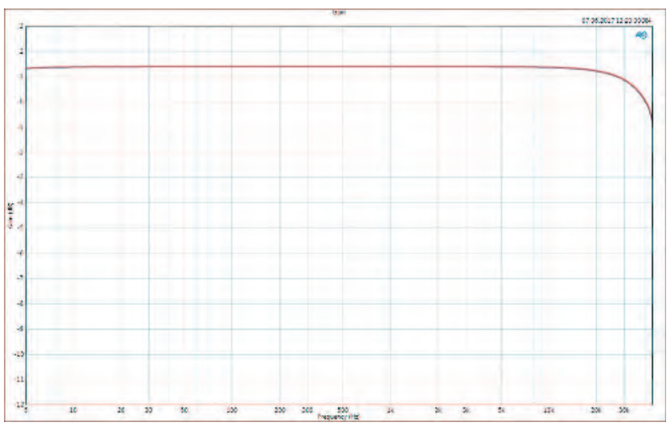
console's special sound or even separate pre-amps. In a similar way to analogue recording technology, where band saturation was used to create a special sound, this was also done with transformers that were used in all inputs and outputs of professional equipment for a long time. Depending on the level, one could make "sound" with the transformers.

In concrete terms, this means: one could add harmonics and slightly manipulate the frequency response.

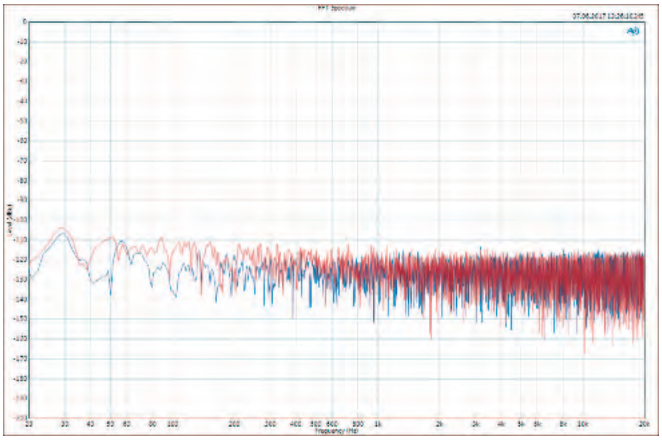
Rupert Neve, now 91 years old, is considered to be the expert in the field of pre-amps and transformer technology. Under the brand name Rupert Neve Designs, a range of exclusive



Graphical one-third octave band EQ. *The filters become slightly narrower towards the high frequencies. If several adjacent filters are used, this can lead to a large increase in the cumulative curve (Fig. 19)*



Frequency response *across all from analogue input to analogue output measured at 0 dB gain (Fig. 20)*



Interference spectrum *at the analogue output with a total level of -87 dBu or -89 dBu (A-weighting) respectively; for the measurement one input channel was routed to each output (Fig. 21)*

pre-amps, EQs and compressors are offered. The “Silk” feature found in some devices can also be found in the PM10’s input paths. The PM10 console’s channel section includes a field with a rotary encoder and two buttons as well as Rupert Neve’s signature. The Silk function can be activated with one button; with the other, the user can select between blue and red Silk. The rotary encoder controls the intensity. Since it is always difficult to describe the sound effects, here’s Yamaha’s original text: „red for sparkling energy, and blue for solidity and power.“ Silk in the analogue original affects the carrier’s saturation behaviour via the feedback and thus creates the desired effects.

In the PM10, the Silk effect is implemented using Yamaha's digital VCM technology, which is already integrated in a dedicated processor on the RY16-ML-SILK card. The fact that even the processor carries the Rupert Neve signature shows that Yamaha is extremely proud of this technology. VCM stands for “Virtual Circuitry Modeling” and not only stands for an imitated user interface and similar parameters, but also for a complete digital processor. This was developed by the team around Toshifumi “Dr. K” Kunimoto a few years ago and exactly reproduces analogue circuits up to the characteristics of passive components such as capacitors and resistors. “Exactly” means here also including all shortcomings, which partly constitute the special character of these devices. Already during the development of the VCM processor some years ago, there was close cooperation with Rupert Neve Design, during which the 5033 EQ and the 5043 compressor from Neve’s Portico series were first implemented. Since Yamaha also has plenty of its own analogue classics to offer, some of Yamaha's own devices such as the EQ-1A and the Compressor OPT-2A have brought back to life. The Silk function has now been implemented for the first time in the PM10 and is now available for all analogue inputs in the I/O racks and also for Omni inputs on the console.

What exactly happens here is not quite so easy to measure. Fig. 14 shows the frequency response with Silk blue and red for the intensities 5 and 10. A slight increase in the high and low frequencies can be identified in setting 5. In the maximum position 10, both are significantly intensified.

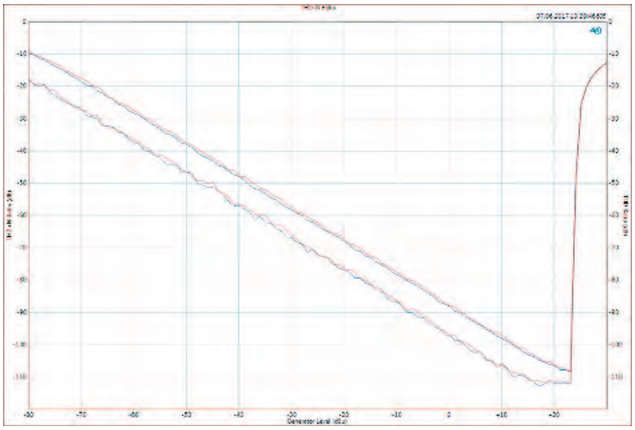
In the setting red 10, the signal’s harmonic content increases noticeably. If these are -100 dB or less without silk, then the even harmonics increase strongly with Silk. Fig. 15 shows the distortion spectra with and without Silk. The typical level-dependent saturation behaviour is shown in Fig. 16 with the THD curve as a function of input level with and without Silk.

From a metrological point of view, Silk red thus produces well-dosed, even-numbered harmonic distortion components depending on the intensity and modulation. At high intensity, the effect is amplified by increasing the high frequencies. Silk blue is particularly noticeable in an increase in low frequencies. Both of these are therefore well suited to the sound description quoted above.

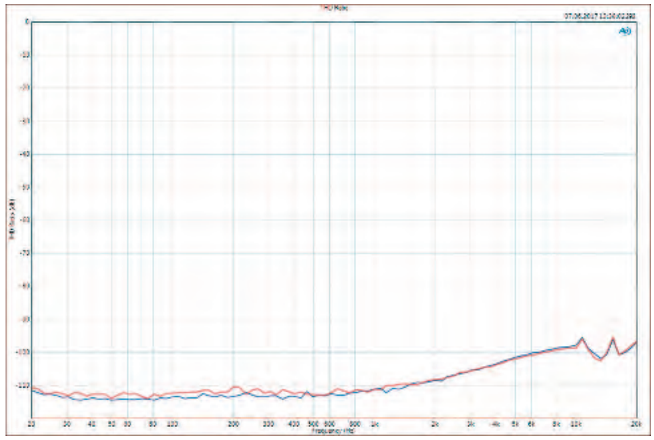
Graphical and parametric EQs

The Yamaha consoles’ filters have been the subject of long discussions due to their special characteristics. The unusual pattern was caused primarily by the lack of compensation at high frequencies, where the filters became noticeably narrower. Fig. 17 shows the filter curves of the PM10’s parametric bell filters. As can be seen from the upper measurement series with blue and red curves, this behaviour can no longer be seen here. The filters practically behave in an “analogue” way up to the highest frequencies, in other words without compression of the filter curve.

Each input channel includes a parametric 4-band EQ, where the filters can also be defined as low or high shelves in addition to the bell characteristic. In addition, there is also a high-pass and low-pass filter, in which the slope for the high-pass filter can be set between 6 and 24 dB/Oct and between 6 and 12 dB/Oct for the low-pass filter. In addition to the EQ



THD and THD+N *as a function of input level at 1 kHz over all from analogue input to analogue output measured at 0 dB gain. The solid line shows the THD+N values; the dashed curve shows THD only (Fig. 22)*



THD as a function of frequency over *all from analogue input to analogue output measured at 0 dB gain and +19 dBu input level (Fig. 23)*

functions, each input channel is also equipped with two dynamic sections, two inserts for plug-ins and a delay. There is a similar amount in the outputs. The parametric EQ even has eight bands, but here only one dynamics section is included.

For the plug-ins, the PM10 also provides plenty of graphical EQs with 8, 15 or 31 bands. Fig. 19 shows the filter curves of the graphic third-octave band EQ, where the previously described filter curve compression effect can still be seen to some extent. No compensation was used, however this is not as noticeable at 96 kHz. The filter curve at 20 kHz is therefore somewhat narrower than the other curves with lower mid-range frequencies. If several adjacent filters are used, the individual filters' relatively large bandwidth results in a significantly higher increase or decrease, as shown in Fig. 19 below. As all filter curves are also shown on the PM10's display, however, it is always immediately clear how the selected settings affect the sum.

Measured overall performance values

All measurements discussed so far referred to the console's individual sections such as pre-amps with ADCs, the filters or the DACs with the analogue output stages. This type of measurement is convenient and allows for a specific evaluation of the individual areas that the audio signal passes through. As a user, however, one is more interested in the general behaviour. With some final measurements, the complete signal path through the PM10 – from the analogue input to the analogue output – should thus be described.

If all faders and the pre-amp gain are in a 0 dB position, then an almost perfect frequency response is measured as shown in Fig. 20. The slight gain of approximately 1.2 dB is due to the difference between full-scale gain in the input at +24 dBu for 0 dBfs and the maximum output voltage of +25.2 dBu in the output.

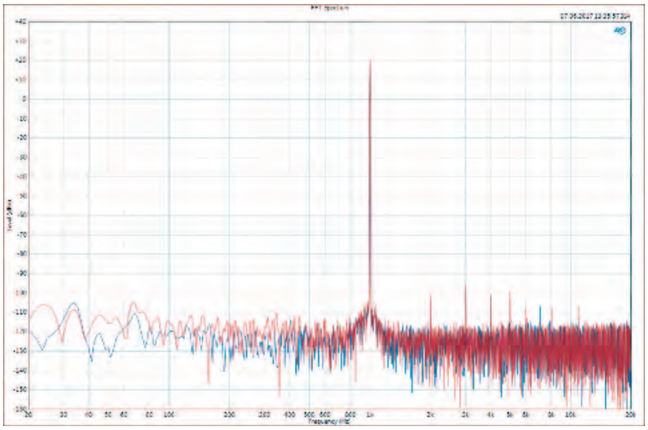
The sum of the interference spectrum at the output from Fig. 21 is -87 dBu or -89 dBu with A-weighting. A dynamic range of 112 dB is thus available for the complete signal path. The value is derived from the 113 dB in the input at 0 dB pre-amp gain and a slight deterioration by 1 dB due to the 120 dB S/N in the output paths. As expected, the very good distortion values measured separately for the inputs and outputs are also reflected in the measurement. Depending on the level, the THD curves run perfectly evenly up to an equally perfect minimum of -112 dB. At higher frequencies, the THDs increase slightly, but we are still talking about values around -100 dB at 10 kHz, where there is no need for discussion at all. It hardly gets better than this. Interesting is the distortion spectrum from Fig. 24, where, despite the measurement now

performed using analogue to analogue measurements, the hint of a jitter bell around the fundamental wave at 1 kHz can still be seen. The SRCs from the AES/EBU inputs or outputs can no longer be identified in the signal path.

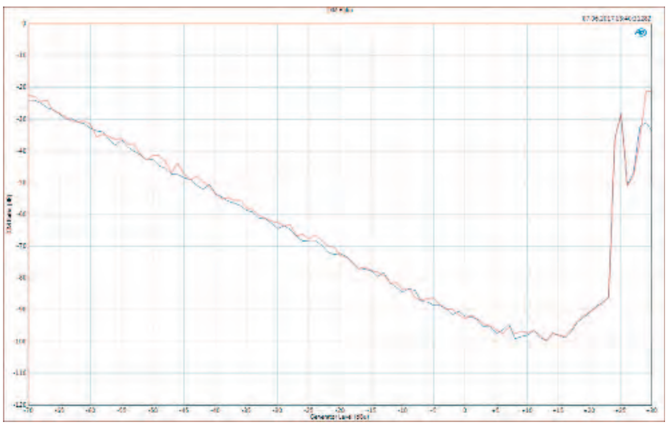
Fig. 25 shows another final DIM measurement, which fills a small gap in the measurements. Unfortunately, as a DIM measurement with the APx555 from digital to analogue is not possible, this curve is still missing. The measurement from Fig. 25 shows the combined behaviour of inputs and outputs. The inputs' very good values of -102 dB in the minimum and -93 dB directly at the clip limit deteriorate only slightly to -99 dB in the minimum and -87 dB at the clip limit when adding an additional output channel in the signal path. Thus, everything is perfect here too.

Summary

The PM10 is Yamaha's flagship digital mixing console. The mixing system – consisting of consoles, DSP engines and I/O racks – is operated using the new TWINLANe fibre optic network and the wired console network. Both networks are designed as a ring structure and thus offer redundancy and a simple set-up. With only a few components and simple cabling, an extremely powerful and flexible system can be created, while there is hardly any demand in a live application that the PM10 system is not up to. The new 16-channel cards with analogue or digital inputs and outputs also allow a flexible configuration of the I/O racks and offer excellent audio quality. Added to this are the new HY cards with up to 400 channels for the TWINLANe fibre optic network and with up



Distortion spectrum *over all from analogue input to analogue output measured at 0 dB gain and +19 dBu input level (Fig. 24))*



DIM *(dynamic intermodulation distortion) over all from analogue input to analogue output measured at 0 dB gain as a function of input level (Fig. 25)*

to 128 input and output channels for Dante connections. The PM10 system's consoles, which are currently available in two sizes, can be operated quickly and intuitively for everyone in a familiar form with channel bays for twelve channels each and a very generous channel section. As expected, all components of the PM10 mixing system make a solid and high-quality impression, just as one would expect from a mixing console system whose entry-level configuration is already clearly in the six-digit price category.