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PRODUCTION PARTNER

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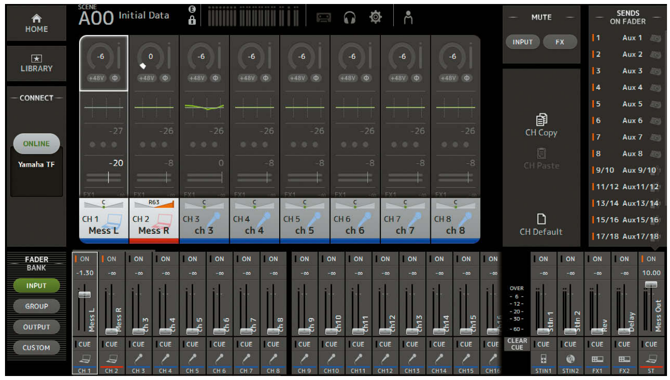
Digital Desk

Yamaha TF Series

Yamaha's Starter Series, comprising three digital desks, features 1-knob and GainFinder thus offering new solutions ensuring fast and safe handling.

Text and Measuring: Anselm Goertz | Photos: Dieter Stork

Central screen *with one fader block visible in the iPad Stage Mix app (figure 1)*



TF Editor *on an external computer (figure 2)*

Due to the favorable price development of DSPs, converters and other electronic components, lower-priced digital mixing desks have now become available, too. So it is even more important to offer the less experienced user a user interface that can be handled easily and provides a clear and simple layout of its rather complex functions. This is what Yamaha had in mind when it created the novel TF series featuring as many standard functions as possible to be accessed directly and a distinct channel assignment. A real innovation utilized in the TF series is the "1 knob" function for EQs and compressors enabling the user to use just the one knob instead of adjusting lots of single parameters. In addition to that, there are libraries containing presets for certain applications and microphone types (the latter created in cooperation with Audio-Technica, Sennheiser and Shure).

So, is the system expandable? Due to cost concerns a permanent integrated Dante audio network interface, which is standard in the CL and QL series, is not included in our basic version of the TF series. For the first time, though, the TF series features an NY expansion slot allowing the bi-directional transmission of 64 channels. The old Y-format was limited to a bi-directional transmission of a maximum of 16 channels and thus less suited for the network connection of stage boxes. Perfectly corresponding will be the introduction of the new Dante card NY64 scheduled in the spring of 2016, featuring a Dante Brooklyn module suited for setting up a redundant network. Springtime 2016 will also see the launch of the new stage box Tio1608D with 16 inputs and 8 outputs and, also, fully remote capable preamps. It will also be equipped with a primary and secondary network connection and can either be operated in redundant or in daisy-chain mode with additional stage boxes.

The user interface is absolutely state-of-the-art now; its huge color display is fully touch-sensitive and can be operated using one or two fingers to swipe, rotate, pinch and pull. If you don't like that or prefer the precision of an incremental position encoder you can, of course, make adjustments or navigate using the central rotary encoder on the right below the display or the four user defined knobs. As you can't connect a gooseneck lamp you have to make sure that – in dark environments - you will still be able to identify inactive keys (as they won't be backlit then). For remote control there are the customary apps for iPads and – in simplified form – for iPhones. The TF StageMix App offers you a nearly complete remote control of the desk and can also be used as an expansion to the desk.

The MonitorMixApp is suited for smaller displays and allows access to the monitor mix via the aux busses for up to 10 external devices. The TF editor, which allows a complete configuration of the desk including scenes and presets, is available for Windows and OS X, as well.

Hardware and Structure

In 2015 the TF series started with 3 consoles, models TF1, TF3 and TF5. They are basically identical and only differ when it comes to the number of available input mixing channels.

Aux mixing *for aux bus 1 performed using the Stage Mix app (figure 3)*

Connector panel *with analog inputs and outputs, card slot, network and USB connector; DANTE can be integrated using a NY64 card*

The TF1 has 32, the TF3 and TF5 40 each. The number of faders and of analog inputs also differs. The TF1 has 16, the TF3 24 and the TF5 32.

The analog outputs on the rear panel are defined as omni outs and can be Matrix-assigned to any desired sources.

There is no matrix on the inputs. Here the assignment equals the analog inputs of the desk or the respective USB channels. The optional Dante card adds a potential Dante channel.

Further rear panel connections are two stereo line-level inputs which are assigned to the two stereo input channels of the console. Alternatively you can also assign the stereo playback from the USB flash drive on the front panel or the USB channels 33 and 34 of the rear USB host for recording.

Direct recording to a USB storage device is only possible in dual channel mode and only via the front USB port. This port can also be used as a direct connection with an iPad that then can be used as a two-channel player or recorder. If you want to record all 32 signals or play them back you will have to use a computer with suitable software, connected via one of the host ports.

Coming soon: Dante network and Stage Box

Today's audio world is digital: Streaming services, music servers, networked devices – digital technology is everywhere. The interfaces to acoustic signals, i.e. converters, though, are still analog. Microphones constitute one side of the signal chain, speakers the other. Microphones convert an acoustic signal into electric signals of very low voltage, speakers then reverse the process, needing high voltages and lots of power to reconvert electric signals into audible sound pressure.

So, what's that got to do with digital networks? The low voltage output of microphones and the high power speaker signals have one common enemy: long cable runs causing interferences and signal losses.

So, the overall aim is to transfer signals from the analog to the digital domain as soon as possible and to reverse this process as late as possible.

Stage box, *in 2015 still a preview model with Dante interface and remote head amps*

TF1, TF3 und TF5 *differ in channel count and inputs and faders*

That's when audio networks come into play, aiming to pick up or send audio signals where they are created or needed. This used to be the task of multicore cables with lots of conductor pairs for back and forth transmission. These multicores were heavy, expensive and failure-prone. Using audio networks enables us to reduce all this to one or two network cables and transmit signals to wherever we want them to be without any losses or retrieve them from the place they are generated in (using big or small stage boxes). Nowadays, some controllers or power amps already have interfaces for audio networks onboard.

Many users, though, deeply distrust this technology because of distressing experiences with PC networks etc. Avowed adversaries also often argue that, in the good old days, failures in analog systems or multicore setups could easily be remedied by just using a spare channel. If, though, a digital network fails, the worst case scenario will mean: total disaster, game over!

To enhance the acceptance of network technology you have to meet, at least, the following user demands: networks must be easy to configure and must be protected against malfunctions and failures. Not least because of these reasons the Dante audio network, manufactured by Audinate, was chosen as the native network used in the QL and CL series as well as in the corresponding stage boxes.

The simplest configuration in combination with a TF, QL or CL mixer consists of an I/O-box connected to a console by using just a simple network connection and nothing else. In case a second stage box is needed, you can pass the network on to the second stage box by daisy-chaining it.

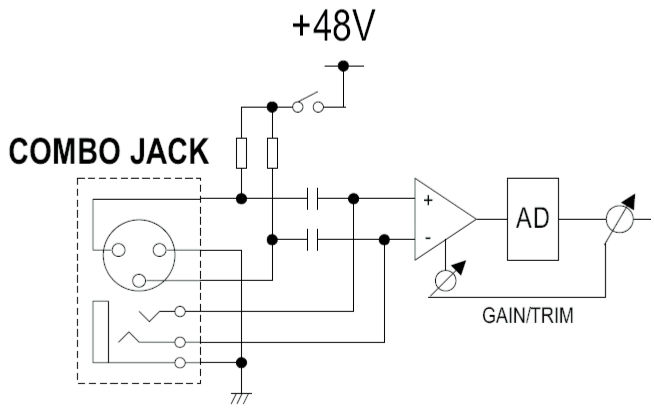
The same goes for additional Dante components. You can, for example, connect a computer as a multitrack recorder directly to the second network port of the console.

Dante, conveniently, provides the possibility to make all audio channels available using a computer as a kind of Dante-powered workstation (utilizing a Dante Virtual Soundcard as network driver). The Dante network, then, will show it as another component. The virtual soundcard will be recognized by the computer as a hardware card with ASIO interface. Simultaneously, the same network can be used to enable the computer to perform configuration and control tasks.

In its simplest form, though, the Daisy-Chain system doesn't offer any redundancy; that's why you need a star topology. It

Sens.	0 dBfs	Noise	
dBu	corresp. dBu	(lin.) dBfs	(A) dBfs
–6	+32	–85	–110
+2	+24	–83	–108
+10	+16	–80	–105
+17	+9	–77	–102
+18	+8	–85	–110
+26	0	–83	–108
+34	–8	–80	–105
+41	–15	–77	–102
+42	–16	–83	–108
+50	–24	–76	–101
+58	–32	–69	–94
+66	–40	–61	–86

Noise *at the output depending on gain. All measurements with 200 ohm load.*
EIN = –126 dBu (lin.) –128 dBu (A) @ max.Gain.



Head amp *with combined gain setting for analog and digital. The analog adjustment happens in 6 dB intervals, the digital part offers 1 dB steps (figure 4)*

allows the use of redundant primary and secondary lines for each device. Should you use more than 2 devices every star needs at least one external switch. All this can be configured simply and cost-effectively by using Standard-Ethernet-GBit switches and Cat.5 respectively Cat.6 cables.

It is really easy to configure a Dante network using Dante "Controller Software". All available devices in the network with their in- and outputs appear in a matrix and can then be routed.

So, three essential requirements have been met: Firstly, the audio network can be set up easily and cost-effectively, utilizing sound and reliable standard network components. Secondly, redundancy – if required – can be achieved using the dual star topology and, thirdly, its configuration will be fast and flexible due to the use of the Dante Controller software. So, even users who aren't that familiar with networks or even rookies: fear not!

Measured Values

How successful the transition from the analog audio reality to the digital level will be, is determined by the capabilities of the A/D converters and those of the upstream preamps. In a digital mixing desk a preamp is of the same importance as in analog ones, as its task is to get everything, from weak microphone signals to high line levels deriving from keyboards or DI boxes to the same overall level. The preamps in TF consoles offer a very wide gain range from –6 dB to +66 dB. Zero dB then stands for the maximum volume of an analog signal with a +26 dBu level. As the preamp is fully remote and recall capable you can't adjust it using a customary potentiometer.

The preamps feature a passive pad and two relay operated amp circuits, arrayed consecutively. The difference between these stages is in each case 24 dB. Switching from +17 to +18 dB gain and from +41 to +42 dB gain is indicated by an audible soft click. Further gain values from 0 to +18 dB can be adjusted by making use of a DCA (Digital Controlled Amplifier). Additional fine-tuning (adjustment in 1 dB intervals) will take place digitally. Figure 4 shows the process of analog gain trim in 6 dB intervals before the A/D converter and the digital gain trim in 1 dB intervals after the A/D converter.

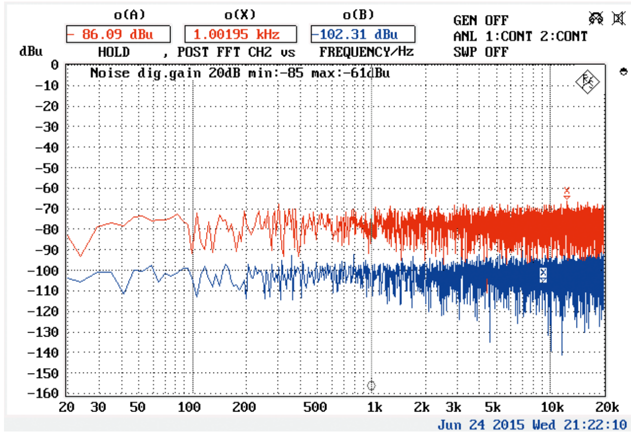
To measure the preamp noise with ADC, a digital gain of 20 dB was set (and subtracted from the measured value later).

So, the input section noise can be measured independently from the output section. At the lowest gain setting the maximum level can be +32 dBu and, accordingly, at the highest gain setting –40 dBu. At the lowest gain an excellent signal-to-noise ratio of 110 dB is achieved by the A/D converter. In case of higher gain values the preamp noise will dominate (but, at the maximum gain of +66 dB, 86 dB can still be achieved).

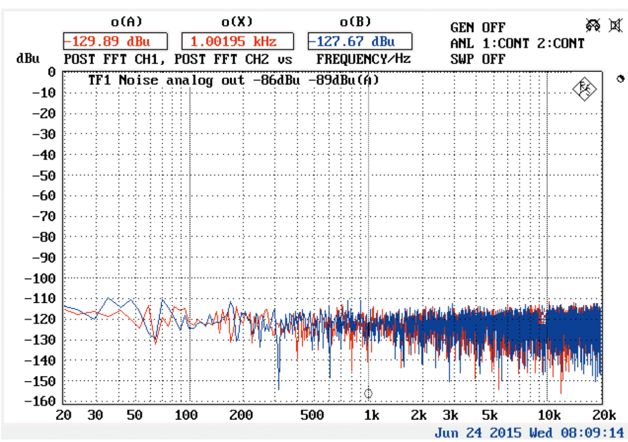
Based on these figures, the equivalent input noise will be –126 dBu. The corresponding interference spectra (figure 6) show a clean white noise. Including the additional 20 dB gain, the overall noise level is –65, respectively –41 dBu. If you then subtract the 20 dB, the corresponding values will be –85 and –61 dBu. The maximum output voltage at the analog outputs amounts to +25 dBu, resulting in dB S/N values of 110, respectively 86.

A separate calculation of the analog output values of the Yamaha TF1 results in an S/N of 111 dB (linear) and of 113 dB (A) noise levels.

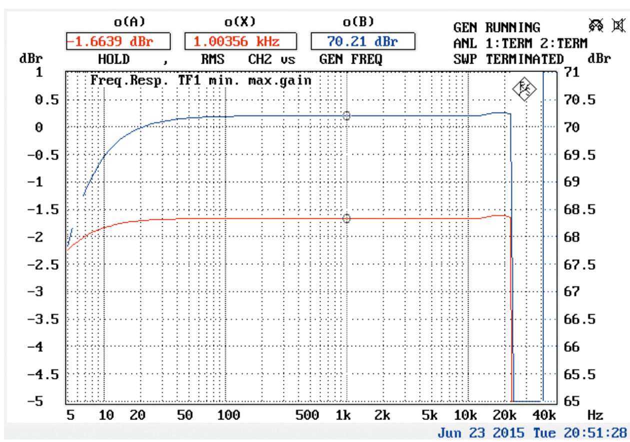
The measuring (as in figure 6) was performed with the master fader down. There is only uncritical white noise in the noise spectrum. Figure 7 shows frequency response measurements for minimum and maximum gain. The curves are quite linear – as to be expected – and only start to drop slightly below 20 Hz. At a sampling rate of 48 kHz the upper end of the curve reaches up to 24 kHz (with all faders in the 0 dB position). This results in a gain of +1.34 dB plus +3 dB due to the Pan setting. The –6 dB preamp setting thus amounts to a value of –1.66 dB.



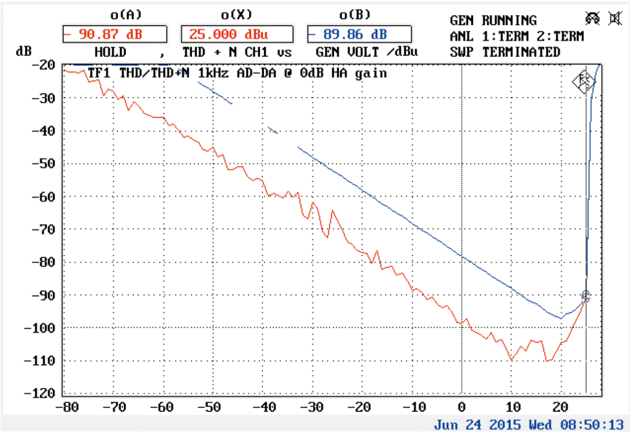
Noise spectrum *at the output in dBu at max. (red) and minimum (blue) preamp gain, measured at the analog output in dBu with additional +20 dB digital gain (figure 5)*



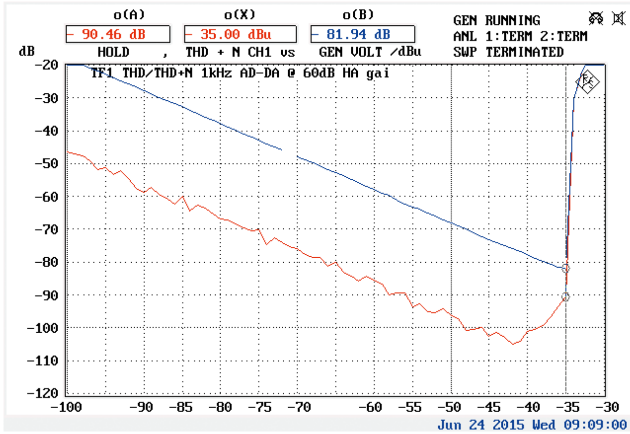
Noise spectrum *at the analog output with fader down at an overall level of –86 dBu (lin) and –89 dBu (A). The maximum output voltage represents +25 dBu (figure 6)*



Frequency response *measured through everything from the analog input to the analog output at a minimum gain of –6 dB (red) and a maximum gain of +66 dB (blue) (figure 7)*



Preamp Gain 0 dB *THD (red) und THD+N (blue), clipping threshold is at +25 dBu input level (figure 8)*



Gain set to +60 dB *THD (red) and THD+N (blue), then clipping threshold is at –35 dBu input level (figure 9)*

Distortion levels and their significance

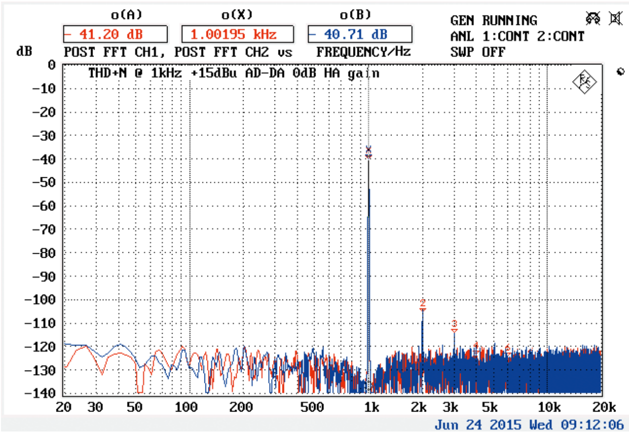
The distortion levels of the TF1 were measured between the analog input and analog output. The measuring conditions correspond to realistic operating situations but there is one slight disadvantage: It's kind of hard to distinguish whether distortions are caused at the in- or output. Measured were THD values, the distortion spectrum and the transient intermodulation distortion, each for preamp gain levels of +60dB and 0 dB. Measurements of the extreme preamp gains (–6 dB and +66 dB) were not taken because these settings do not reflect normal operating conditions and are only required for very special cases. The curves in figures 8 and 9 demonstrate the proportional signal distortion (y-axis) as a function of input level (x-axis). Both curves show the THD (red), the sum of all harmonic distortions and the THD+N (blue), i.e. all signal shares not deriving from the signal source including noise.

When you measure at low test signal levels, the share of noise and distortion will, necessarily, be high. When you increase the test signal levels, THD and THD+N will decrease more and more up to the point, where the clipping threshold is reached. The minimum of the curve is mostly close to or directly at the clipping threshold. In figure 8, the clipping threshold for 0 dB gain is reached at +25 dBu, a point with very low distortion values of –90 dB (= 0.003 %). The distortion minimum is reached at a marginally lower level of 10 dBu and also really great –110 dB (= 0.0003 %). Reaching these good values is a bit more difficult when the preamp is confronted with higher gain settings, here + 60 dB. Still, the figures are, nevertheless, quite good compared to measuring at 0 dB. You'll find the corresponding curves in figure 9.

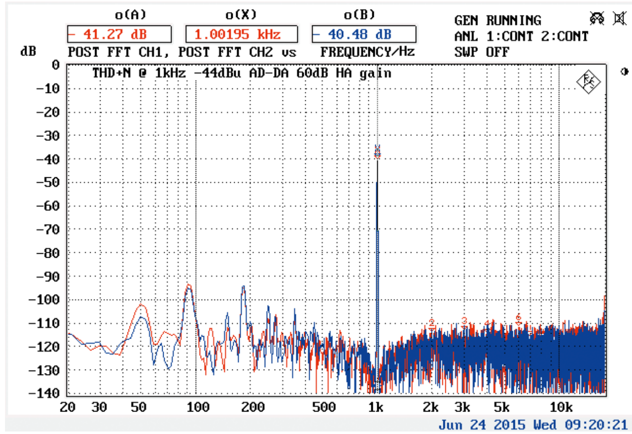
In addition to the absolute distortion values their spectral structure is quite interesting. The desired harmonic distortion spectrum should, of course, have low distortion values but also the lowest possible number of uneven (k_3, k_5, ...) distortion shares descending as quickly as possible towards higher order.

Ideal would be some k_2, significantly less k_3 and, preferably, nothing higher than that. The harmonic distortion spectra for a sine wave at 1 kHz were measured at 0 and at +60 dB gain (figures 10 and 11). The test signal level was always 10 dB below the clipping threshold. Both spectra are quite close to the ideal targeted values and should guarantee a great sound quality of the preamp.

The third of the measurement series deals with transient intermodulation distortion, also known as TIM or DIM. The test procedure is the same as in the THD measuring with only one difference: the test signal is not a sinusoidal one but a mix of sinus signal and rectangular pulse. This kind of test signal with its steep rectangular pulse is far more challenging for the tested circuit than the sinusoidal signal. That is why the DIM test results are said to be more relevant for evaluating sound qualities than just measuring THD. DIM values of –80 dB are



Distortion *of the TF1 at 0 dB gain and +15 dBu input level (figure 10)*

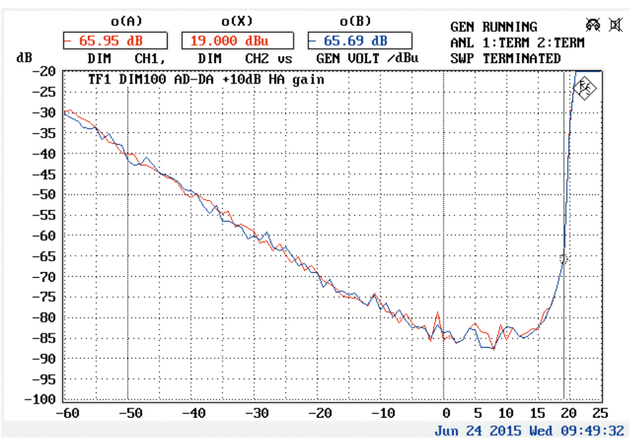


Distortion *at 60 dB gain and –44 dBu input level (figure 11)*

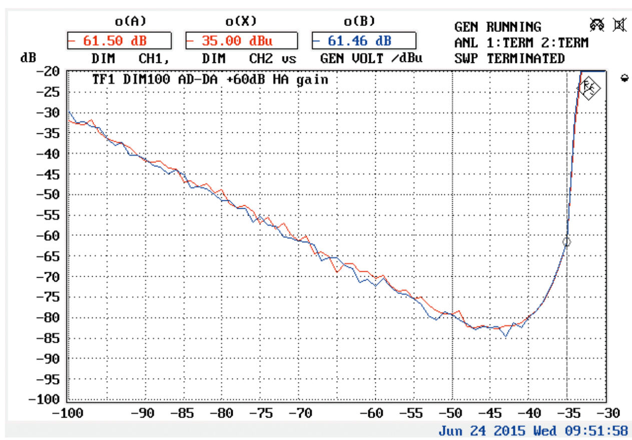
already considered to be really good. DIM curves for 0 and 60 dB preamp gain are shown in figures 12 and 13. The excellent minimum of –85 dB in both cases was measured at approximately 10 dB below the maximum volume. So, again: really outstanding results!

Signal processing and structure

Digital equipment has the advantage that the order of the blocks of signal processing can be chosen quite flexibly. The same goes for the taps in a signal path, e.g. where the signal

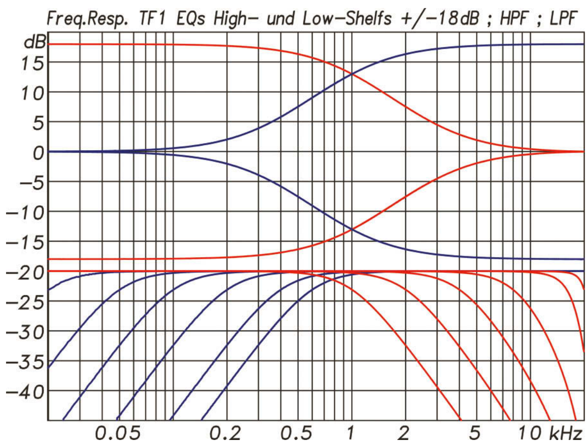


DIM *transient intermodulation distortion at minimum gain (figure 12)*

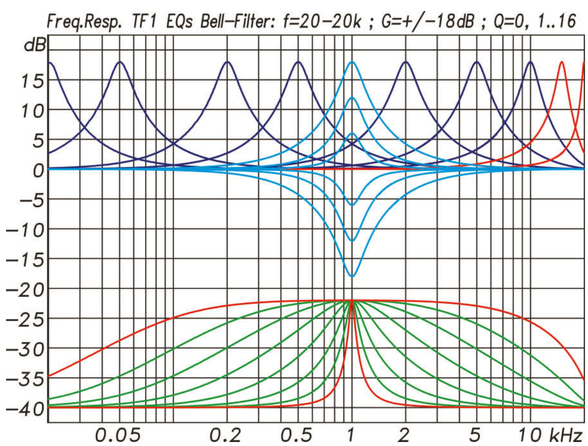


DIM *transient intermodulation distortion at maximum gain (figure 13)*

Parametric EQs *in all inputs fully equipped, alternatively the EQ can be set to "1-knob" mode (figure 14)*



High and low shelving *filters and high pass and low pass filters (below) in all inputs (figure 15)*



Bell-Filter *with settings for frequency and gain (above) and for quality (figure 16)*

for the aux busses is generated. The signal path of a channel strip is structured as follows: input selection, digital gain, high pass filter, 4 band parametric EQ, gate, compressor and DCA (fader). In Input Select the selection "Slot In 1–32" is still greyed out because it will be activated in the desk's firmware along with the release of the NY64 expansion cards.

The tap for recording to a DAW connected via USB can be switched globally. As a rule you'll use pre-processing setup because the settings for a LIVE show are normally not the ones you'll want for a later downmix to a stereo file. Should you like to do the downmix using the console again, you can play back the unprocessed signals to the same position and then perform the mix. To play back a multi-track recording of a concert is sometimes quite convenient because you can use the recording for a sound check on an empty stage. This is advantageous in several ways:

Firstly, the musicians don't have to go through a boring and tiring sound check and, secondly, the sound engineer at the console can kind of get a good LIVE feeling while doing the sound check (we all know that many musicians play louder and heavier during a show than when doing the sound check).

Filters

Each input channel of the TF consoles features an adjustable 20 Hz to 600 Hz high pass filter and a 4-band fully parametric EQ, which can, either, all be defined as Bell filters, or, alternatively, one as high or one as low-shelf filter. The new user interface for the filter section has turned out great: You can enter all data directly, using the physical encoders, or just use the touch panel controls (one or two finger mode). Using one finger you can adjust gain or frequencies, with two fingers you can alter the filter's quality factor. The same can be done using the iPad app. The TF editor transfers all three controls to the mouse wheel; real purists or techno nerds then have the chance to enter all filter parameters directly as numeric values.

Typical for Yamaha: the filters are not compensated with respect to their transfer function. Close to half of the sampling rate, the filter curve will be compressed while sweeping the frequency axis. The reason is that the infinite frequency axis is transformed from the analog to the finite and discrete spectrum of the digital domain.

Is the half sampling rate at 24 kHz, you will already see significant changes in the filter curve shape above 10 kHz. This is clearly depicted by the two red curves in the upper part of figure

16. You can also notice this effect by looking at the filter curves in the display. It is important to realize, that this is not something like distortion but only a slightly changed filter function, which deviates from familiar analog behavior. Other manufacturers compensate this effect mathematically-feasible up to reaching the cut-off frequency. Yamaha, traditionally, refrains from using this compensation method – as usual, it's, more or less, a matter of taste.

At all console outputs you will also find graphic $^1/_3$-octave EQs in addition to the parametric filters. Due to the high number of filters, the graphics EQs have been implemented as so-called 12 out of 31 filters to reduce the crucial computational power requirements: Out of 31 a maximum of 12 frequency bands can be used actively within the range of 20 Hz to 20 kHz. That's usually sufficient in practice. The curve shape altering effect close to half of the sampling rate can also be noticed when looking at the $^1/_3$-octave EQs. Due to the high bandwidth of the respective filters there is some distinct overlapping. Operating adjacent filters will therefore lead to strong cuts or boosts which you have to be aware of.

Dynamic and Effect Functions

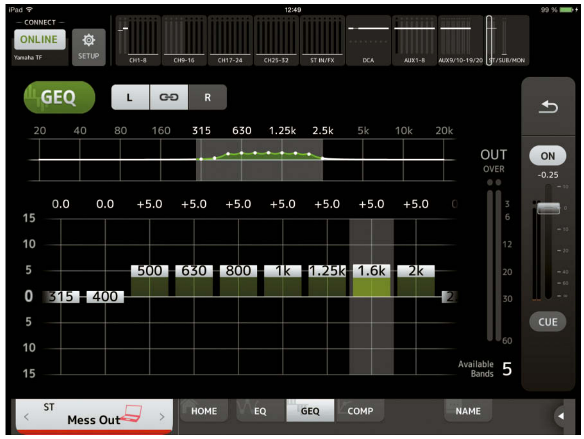
At each input of the TF desks there is a compressor/limiter in combination with a gate; at the outputs you'll only find compresssors/limiters. Again, the design of the user interface is clear; everything is neatly arranged and easy to operate. You can gather from figure 20 what effects different settings will have. The figure shows the compressor reaction to a sine burst with time constants of 50 and 500 ms for attack and release. The threshold level of –20 dB here refers to 0 dBfs in the digital domain.

The two FX buses and – if required – also the six Stereo Aux busses numbers 9 to 20 can be fed from all inputs. All respective masters comprise four band fully parametric EQs and several effects; you will have varied reverb programs, delays, chorus effects and a multiband compressor with four bands at your disposal. If you like to use your own special effects you can feed them in via the FX or Aux outs and return them via two inputs of your choice.

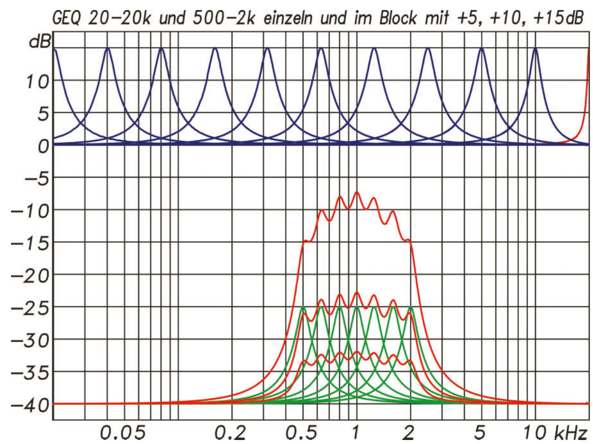
1-knob functions and facilitations

To users who just want to use the desk as soon as possible, the TF series offers a lot of features facilitating operation. For input channels you'll find ample libraries with preset filter and compressor settings to record all kinds of instruments. These presets are combined with mic recommendations made by renowned microphone manufacturers like Audio-Technica, Sennheiser or Shure (including optimized settings for the respective recordings with particular mics). You can also create your own libraries or modify existing ones.

For a desk novice, adjusting gain settings of the preamps often proves to be the greatest challenge. You don't want to set the levels too low and, maybe, cause noise but, on the other hand, you have to avoid clipping at all costs. A great help is offered here by the TF consoles' novel GainFinder function. Finding a



Graphic $^1/_3$ EQs *in the outputs (figure 17)*



Graphic EQ *single filter bands from 20 Hz to 20 kHz (20 kHz Band in red), below filter ranging from 500 Hz to 2 kHz single (green) and en bloc (red) with +5, +10 and +15 dB gain (figure 18)*

Dynamic Processing

In addition to the use of filters, dynamic processing constitutes the most important function of a mixing desk. Especially LIVE music often generates a large dynamic range meaning you would wish a certain compression would take place. Also, there are often problems with noise during quiet music passages due to open, unused mics. By means of compressors loud signal peaks can be reduced and gates can block unused mic channels. Both functions utilize thresholds that determine when the signal processing will start. If the compressor threshold is exceeded the signal will be attenuated in a controlled way. The ratio setting determines the intensity of attenuation. When all values are radically cut off above the threshold, then a limiter as a special type of compressor is in use. The gate will react in a reverse way, i.e. reduce the level or shut off the signal path completely when the values fall below the threshold.
Both functions imply time constants. The attack time determines how fast a reaction to exceeding or falling below the threshold will take place. The hold time defines how long processing will still happen when exceeding or falling below the threshold level has stopped and the release time defines, how fast the level will return to the initial state after the end of the hold phase.

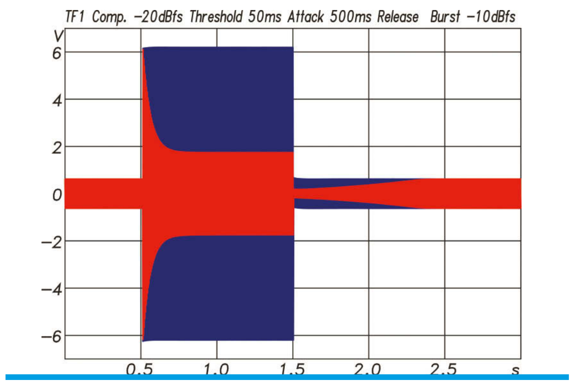
Controlling compressors *YAMAHA TF series (figure 19)*

favorable setting with sufficient headroom for signal peaks is clearly facilitated by just using colored bar indicators.

Another very noteworthy and helpful innovation – not only for beginners – is the 1-knob function for the parametric EQs. You can choose between different settings, e.g. "Intensity" and "Vocals" for inputs, "Intensity" and "Loudness" for outputs. "Intensity" can be applied to all filter functions and shrinks or expands a complete filter curve's gain settings. Let's say you have a given filter setting and you switch into 1-knob mode; the curves original shape will first be preserved with the intensity faders set to 50. At lower values the whole curve will become flatter and flatter, at higher values this process will boost the curve. At the intensity setting of zero, the filter will be deactivated and at intensity setting 100, gain settings will be doubled (i.e. a Bell filter, previously at + 6 dB, will then be at + 12 dB. In figure 22, the red curve represents the starting point. The green curves were measured at intensity settings between 10 and 100.

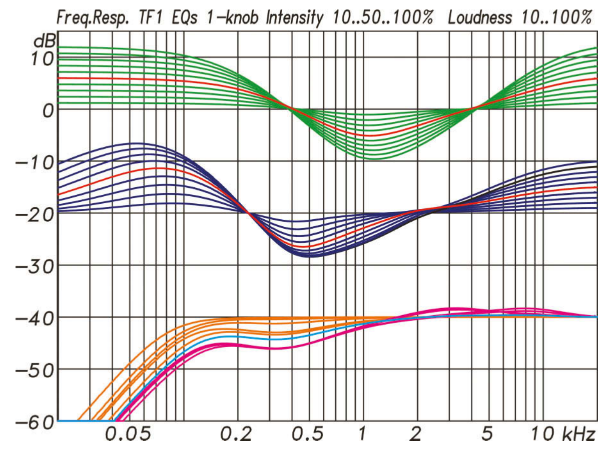
There are pre-configured loudness filter settings, also adjustable between values of 0 to 100. The idea of a loudness filter is to compensate one weakness of our ears: in case of relatively low levels they pick up low and high frequency ranges less than midrange frequencies. So you would increase loudness filtering when you have soft background music, you would decrease filtering or switch "Loudness" off completely when the sound environment becomes louder and louder.

For inputs there is a preset "Vocal" filter with a 140 Hz high pass filter and a slight cut of the low mids and a boost of the higher frequencies.



Compressor *reacting to a sine burst at 50 ms attack and 500 ms release for a threshold set to –20 dBfs (figure 20)*

1-knob mode *adjustments for parametric EQ in the outputs set to "Loudness" (figure 21)*



1-knob filter *set to "Intensity" (upper), "Loudness" (middle) and adjusted for vocals (lower) (figure 22)*

Compressor *1-knob adjustment for quick access without irritating or confusing beginners (figure 23)*

Compressor operation is also simple using the 1-knob function. For fixed time constants and a ratio of 2:1 you can use the 1-knob parameter to set a combination of threshold and output gain. The signal will be moderately compressed and then again the overall level boosted, or, to put it simply, it will seem to be "louder".

Summary

The three TF series consoles expand Yamaha's product range of digital mixing desks in the lower price segment. All 3 models are equipped in a way that enables you to use the consoles' 16, 24 respectively 32 Mic/Line inputs without any further accessories. Users who want to get rid of their old multicore after acquiring one of the new consoles can use the NY64-Dante Expansion card and Dante stage boxes (launch in spring 2016) to create an up-to-date audio network in a simple and secure way. Handling is equally easy and state-of-the-art. The desks feature a big touch panel, an editor software for computers as well as remote apps for iPad and iPhone, thus making them really convenient and very useful for technicians and musicians. Yamaha experts have also deliberated thoroughly about how to facilitate getting started for beginners. That is why they have included ample libraries with predefined channel settings for all kinds of instruments and established microphone models, GainFinder for the inputs and the handy 1-knob mode for parametric EQs and compressors.

All the technical tests have revealed that the TF desks offer state-of-the-art technology and all measured values are awesome.

The very reasonable sales prices of € 2,975 (TF1), € 3,570 (TF2) and € 4,165 (TF3) give customers the chance to get a great, open mixing desk system that can be enormously and flexibly expanded using a simple and adaptable Dante network.

Yamaha Music Europe GmbH
Siemensstr. 22 – 34
D-25462 Rellingen
www.yamahaproaudio.com