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The Top QSC TouchMix Model

QSC TouchMix-30 Pro

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Article and lab testing by Anselm Goertz | Photos by Dieter Stork, Anselm Goertz

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QSC
TouchMix-30 Pro
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The Top QSC TouchMix Model

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Following on from its TouchMix-8 and TouchMix-16 models, September 2016 saw the introduction of QSC's TouchMix-30 Pro, the third and largest mixer in the TouchMix series to date. As suggested by its name, this mixer offers a total of 30 inputs. QSC's official information on the TM-30 includes a long list of other features that suggest a product aimed at the professional user. This is also reflected in the "Pro" suffix in the mixer's name. Despite this, however, the maker has stayed true to the TouchMix control concept. Instead of faders or a channel strip with direct access to the parameters of any particular channel, the primary control interface is the 10-inch touch display. This is complemented by gain knobs for the preamps, a central rotary encoder and a number of buttons that give direct access to certain general functions.

A quick look at the feature list gives the impression of a functionally well-rounded compact live mixing console, just with none of the usual channel faders. Like all the other functions, these show up on the touch display. This means that the TM-30 is in a niche between rack mixers that rely completely on an external mixing app on a tablet and classic console-type mixing desks with traditional control elements. The target group here could be customers who need more hardware than that offered by a tablet and less than you get on a full-on console. The TM-30, of course, like its smaller siblings, offers the option of using an iPad or smartphone as an external controller.

Hardware and Feature Set

Let's begin with a look at the hardware. The TM-30 uses a mini-console format, with its I/O connections all located on the rear panel. The control surface has two rows of knobs for adjusting the gain for each of the 24 preamps. The inputs use the XLR standard, with four offered as XLR/TRS combo connectors. 48-Volt phantom power is available for each preamp individually. Of the 30 inputs, the remaining six are line-level TRS jacks. There's also provision for an additional stereo signal to be streamed from a USB memory device. The two on-board USB-A ports also allow the connection of a hard drive to record and play back all 32 inputs. The USB-B port, meanwhile, lets TM-30 owners utilize their device as a DAW front-end. On the output side, the feature set is equally ample. The TM-30 packs in 14 Aux outputs, a stereo monitor output, a stereo Master output and three headphone jacks.

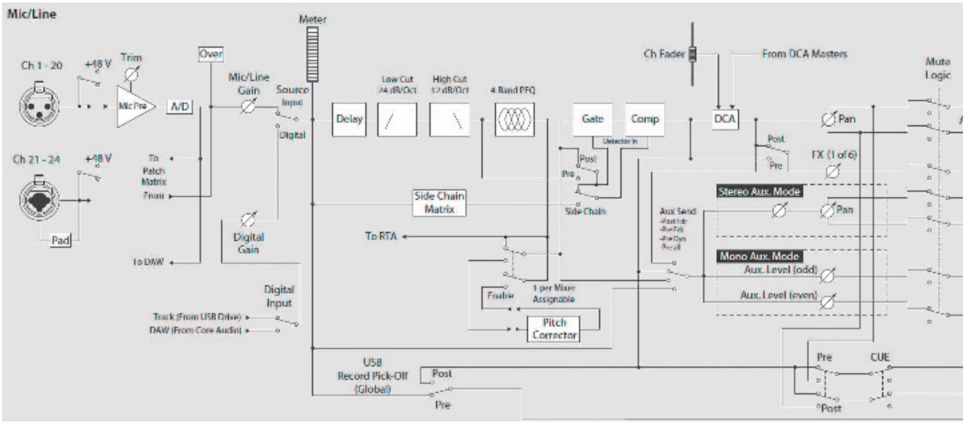
Complex mix architectures made easy

The easiest way to get a sense of how a mixer works internally and what functions are available is to peruse the block circuit diagram. Fig. 1 shows an input channel, the first element of which shows some interesting capabilities. After the signal has passed through the preamp and the A/D converter to arrive in the digital realm, it's routed directly to a matrix that can connect any input to any channel. Also, all inputs can be made available to a DAW for recording or further editing, or saved to a hard drive as a multi-track recording. The signal can be tapped from a point before or after processing has been applied for that channel. In addition, the user can select additional sources for each channel outside the matrix, namely a DAW or multi-track recording saved on a hard drive.

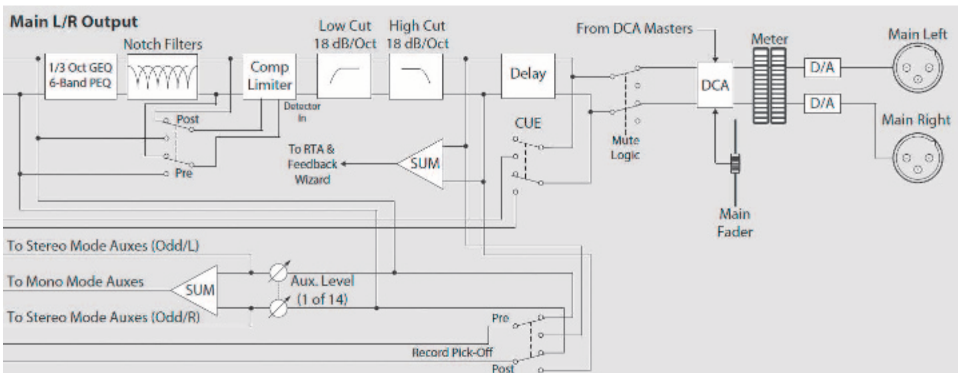
After the inputs have been routed to the desired channel, the signal is processed using a delay, several filters, a gate and a compressor. The filter section comprises a high- and low-pass filter as well as a four-band parametric EQ. The final input channel components are the faders and the bus routing options.

The Master L/R and Aux 1-14 outputs are set up in a similar way. Fig. 2 shows the corresponding block circuit diagram. These outputs also offer filters, delays and a compressor/limiter. The filter section is functionally more comprehensive here than on the inputs. The parametric EQ has an additional two bands for a total of six, and incorporates a graphic 1/3 octave EQ as well as high-pass and low-pass filters. A closer look at Fig. 2 shows that between the EQs and the compressor there's also a filter block with 12 notch filters for tackling feedback problems.

As such, the TM-30 offers a sizable feature set that can compete with considerably larger consoles. But there are a few features we won't find on the TM-30. First and foremost would be remote control and recall abilities for the preamps. Other desirable functions might be a network connection, say, or a separate stage box also with remote-controllable preamps.



Input channel *as a block diagram with a clearly structured and practical structure (fig. 1)*



Output channel *as a block diagram, showing the generous scope of the filtering functions on offer (fig. 2)*

Touchscreen, iPad or Smartphone

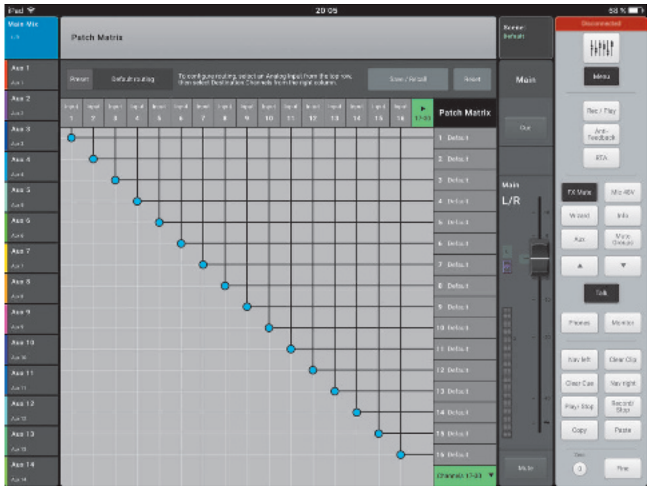
Controlling the TM-30 can be accomplished using either direct access on the mixer itself or one of the tablet and smartphone apps that run on iOS and Android. The smartphone versions are geared towards setting up a personal monitor mix, and restrict the settings available to the volume levels; there's no access to processing or I/O routing. Using a tablet, however, gives you access to all parameters across all supported operating systems. All remote app flavors support assigning rights to each user appropriate for the situation or use. This lets you, say, assign individual band members only the rights they need to set up their own monitor mix. FOH or monitor mixing engineers can utilize a tablet, say, as a remote mixer to adjust a parameter while onstage or elsewhere in the venue. Tablets can also be used to extend the mixing desk's GUI by mounting it on a suitable holder or stand. How easy is the TM-30 to operate? That depends to a great degree on how comfortable you are working without "real" faders. The GUI for each channel strip is designed to make the fader immediately available whenever it's needed, and also to avoid unintentionally changing other settings.

Making smaller, stepped dB level changes is a matter of tapping above or below the virtual fader knob. Alternatively, users can also just tap on the fader and use the controls to make the desired incremental adjustments.

I'm sure it's not a problem quickly to drag down the fader in an emergency. The horizontal faders in the Aux Send view, on the other hand, are smaller. You can, however, switch these to the larger fader view with a single finger tap.

The channel preset and mixer scene functions also speed up your workflow. QSC also provides a live sound preset library of 120 presets for the input channels covering a range of instruments and signal sources. These should make it easy for even the greenest novice to get started. Saving and transferring your settings is performed via a storage device attached to one of the USB ports. One function that proved useful in real-world scenarios was that all channels are given clear and descriptive names, so it's always obvious which channel is assigned to which source. Anyone having

Overview *for an input as displayed on the iPad app. Thanks to the clever interface design, users are hardly ever more than one finger tap away from all important functions (fig. 3)*



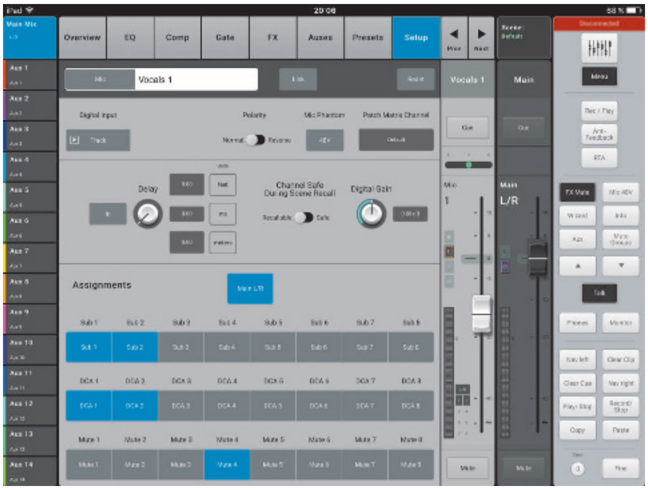
The matrix *connects physical inputs with input channels, a feature normally found only on bigger mixers (fig. 4)*

problems getting set up can fall back on one of the wizards for Gain, Anti-Feedback, Room Tuning or FX. When needed, the full manual for the TM-30 can be accessed at any time using the Info button. Easy to navigate, this feature lets you quickly find the function you're looking for, and you can browse through the information you need in any of six different languages.

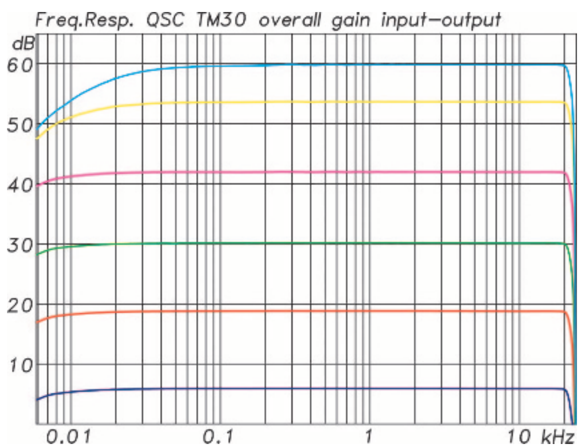
All in all, the clarity of the interface design means that you should already be feeling comfortable using the TM-30 after a only a few minutes. If you want to practice finding your way around the TM-30 or evaluate the GUI concept before buying, the iPad app offers a good way of previewing the mixer's design.

In the lab: Signal-to-Noise Ratio and Gain

Let's move on to the lab test results for the TM-30. These are divided into three categories. First, we'll look at the frequency ranges, gain values and noise levels. The second group com-



Input set up *with routing to subgroups, as well as mute and DCA groups (fig. 5)*



Frequency *and gain measured across the analog input and analog output at preamp gain values from +6 dB to +60 dB (fig. 6)*

prises the measured distortion values, while the third examines the EQs and other filtering functions.

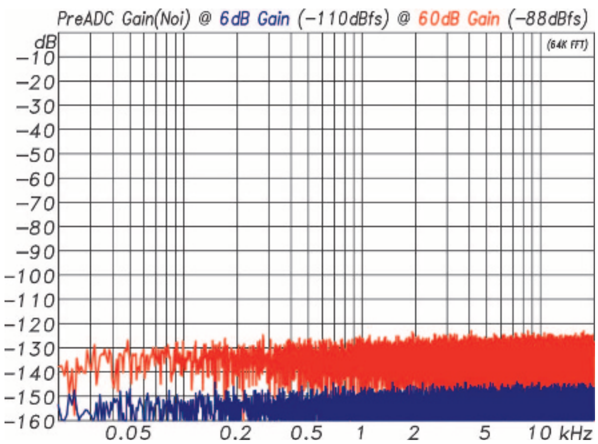
According to the provided technical specifications, the maximum level for the XLR inputs is +16 dBu, while the four inputs with XLR/TRS combo connectors offer +26 dBu. Using a TRS jack on these inputs activates a –10 dB pad. The table shows the values for the possible gain settings from +6 to +60 dB. The Gain value is measured across the analog input and analog output with the channel and master fader settings at unity gain. Figure 6 shows the accompanying frequency measurements at a range of settings between minimum and maximum boost. As expected, the curves are straight as a ruler, and only start to taper off slightly at the low end when the gain is pushed hard. At 48 kHz sample rate, the high end of the curve reaches up to just under 24 kHz. Should you need to record direct to 44.1 kHz, the mixer can also be run at that sample rate; here, the curve dips a bit earlier, however.

The noise measurement including ADC (analog-digital conversion) was made with the mixer set to a digital gain of 20 dB that was subsequently subtracted from the measured value. This gives us a good guide for the noise values generated by the input section only, independent of the outputs. At the lowest gain setting, the maximum input level was measured at +15 dBu. At maximum gain, this value was –39 dBu. The lowest gain settings give a very good signal-to-noise ratio of 110 dB, itself determined by the AD conversion stage. When the gain is pushed, the noise starts to dominate. Even at maximum gain of 60 dB, however, the SNR is still 88 dB. Calculating the equivalent input noise yields a value of –127 dBu.

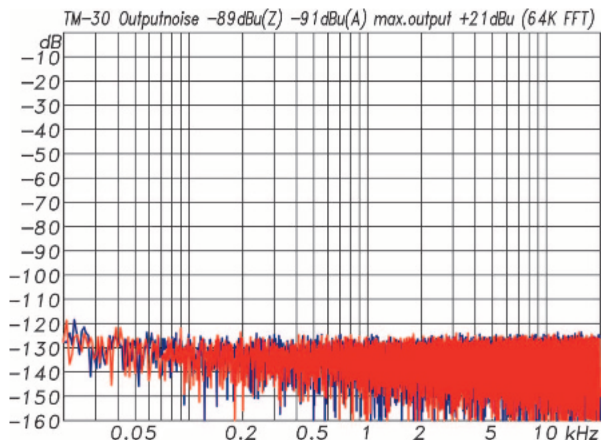
Examining the respective noise spectra for the minimum and maximum gain settings (Fig. 7), we see a clean white noise

Gain dB	0 dBfs entspr. dBu	Noise lin. bew. dBfs	Noise A-bew. dBfs
+6	15	–110	–112
+18	3	–110	–112
+30	–9	–109	–111
+42	–21	–105	–107
+54	–33	–94	–96
+60	–39	–88	–90

Noise measured at the output against the gain level. All measurements made at 200 Ohms input impedance.
IN = –127 dBu (linear) –129 (A) @ max. Gain

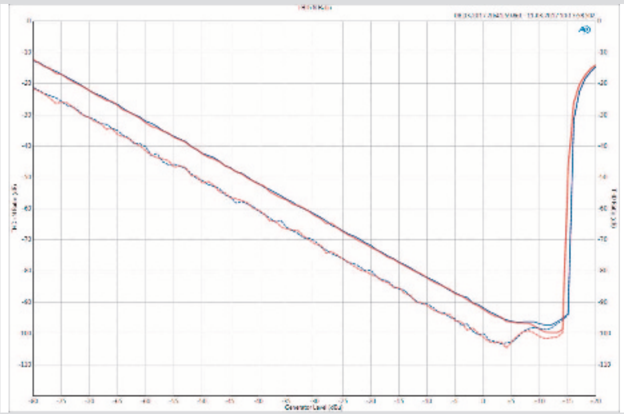


Preamp noise spectrum *including ADC for preamp gain values from +6 dB (blue) to +60 dB (red) (fig. 7). See the article for more on the measurement methods used*

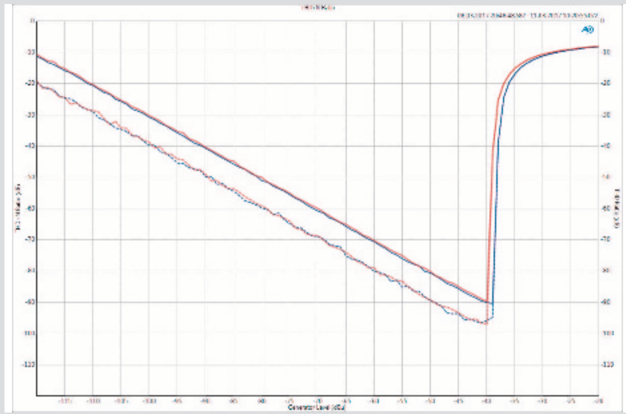


Noise spectrum at output with Master fader set to –∞. The noise level is –89 dBu (linear), maximum output level is +21 dBu (fig. 8)

Distortion



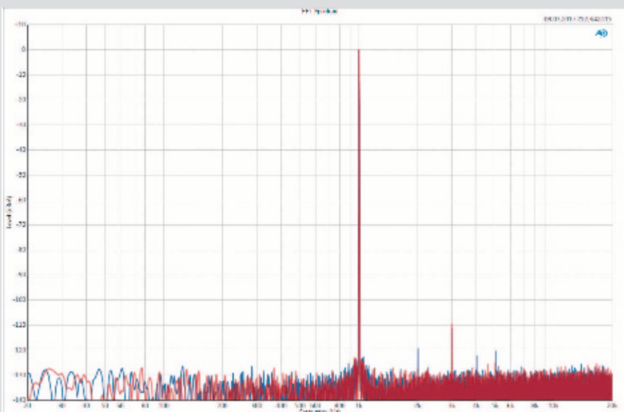
THD and THD+N *measured across analog input and analog output plotted against input level at +6dB preamp gain and 0 dB digital gain. Dotted line = THD, solid line = THD+N, two channels measured (red and blue, fig. 9)*



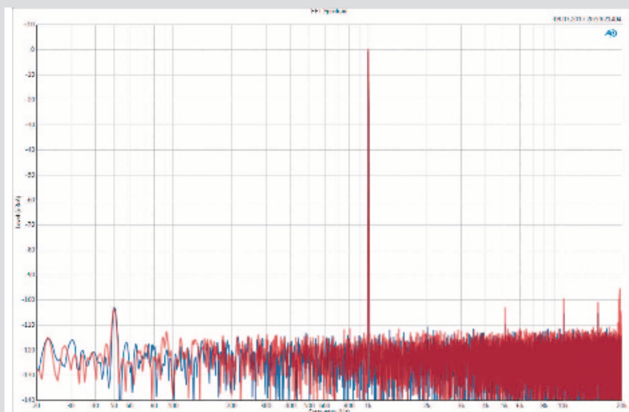
THD and THD+N *measured across analog input and analog output in relation to input level at +60 dB preamp gain and 0 dB digital gain. Dotted line = THD, solid line = THD+N, two channels measured (red and blue, fig. 10)*

The distortion was measured across the analog inputs and the analog outputs. Although this depicts the sound of the mixer as used in the real-world, it doesn't allow us to differentiate between the characteristics of the input and output sections respectively. That would require us to be able to input and pick off the test signals at the required points, which isn't possible on the TM-30. The THD and THD+N values were measured, as were the distortion spectrum and the transient intermodulation distortion, at both maximum and minimum preamp gain settings. The curves in Fig. 9 and 10 show the distortion present on the y-axis in relation to the input level on the x-axis. The two curves depict the THD (k2, k3, k4, k5, …) as the sum of all harmonic distortions and the THD+N as the sum of all parts of the output signal not present in the stimulator signal.

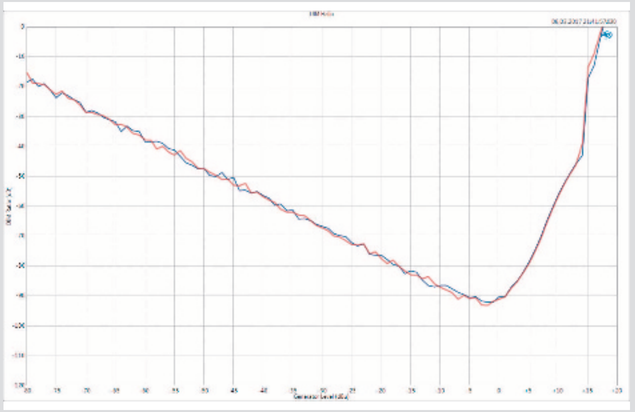
The noise and distortion are, of course, higher for signals at lower input levels. THD and THD+N fall as the test signal



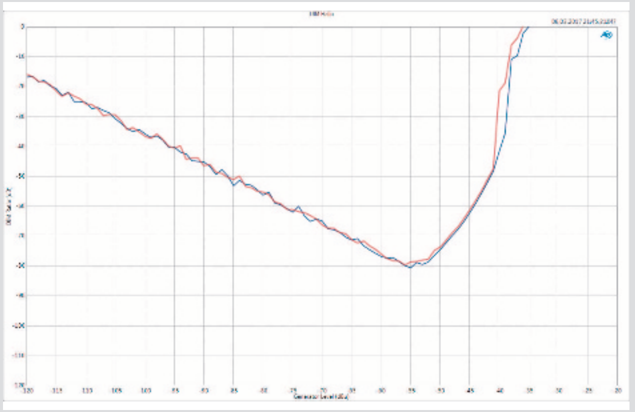
Distortion spectrum *at minimum gain and +4 dBu input voltage (fig. 11)*



Distortion spectrum *at maximum gain and –50 dBu input voltage (fig. 12)*



TID *(transient intermodulation distortion) measured across analog input and analog output in relation to input level at +6 dB preamp gain and 0 dB digital gain (fig. 13)*



TID *(transient intermodulation distortion) measured across analog input and analog output in relation to input level at +60 dB preamp gain and 0 dB digital gain (fig. 14)*

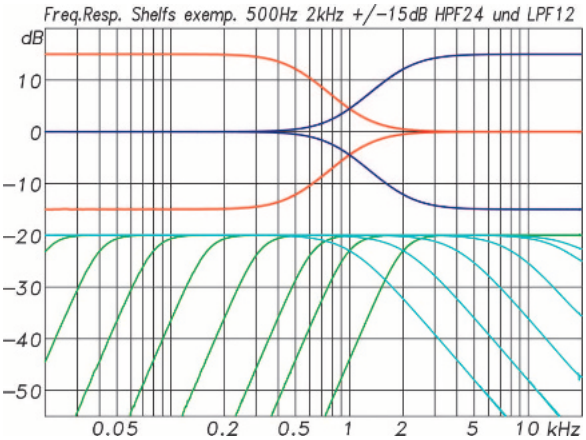
level increases right up to the point where levels are reached that lead to clipping. The lowest points on the curve are mostly just before or at this limit. Fig. 9 shows the results for minimum gain. The clipping limit is reached at around 15 dBu, with very good distortion levels of –100 dB (= 0.001%). The lowest distortion values are attained at a lower gain of +4 dBu, at –103 dB. The preamp has a harder job at high gain levels. Fig. 10 shows the curves for maximum gain settings. Happily, the data shows responses more or less as good as those measured at minimum gain. Overall, the distortion elements in the signal were measured to be several dB higher, while the clipping limit is reached at –39 dBu for the input signal.

In addition to the amount of distortion, its spectral distribution is also of interest. Desirable characteristics include – apart from generally low amounts of distortion – the lowest possible values for odd-numbered harmonics (k3, k5 etc.), as well as harmonics that drop off as sharply as possible the higher up we go. The distortion spectra for a 1 kHz sine wave shown in Fig. 11 and 12 show a harmless k3 line at –110 dB, with nothing else of note to report. At maximum gain, the distortion is not distinguishable from the overall noise floor.

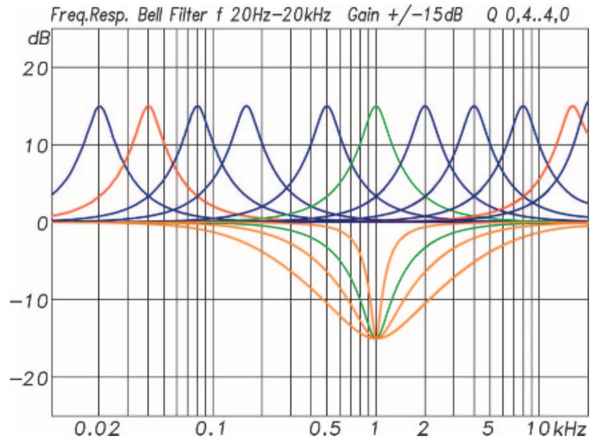
The third series of tests probed the mixer's values for transient intermodulation distortion, or TIM. The process here is identical to the measurements for THD except that instead of using a sine wave, this test uses a mixed square-sine wave. The steep slopes of this test signal put greater demands on the circuit being measured than the smooth shape of a sine curve. That's the reason this test is often seen as being more relevant to the sound characteristics than measuring only the THD. Values for a TIM of –80 dB are considered very good. The TIM curves for minimum and maximum gain are shown in Fig. 13 and 14. The lowest levels of –90 dB and –80 dB respectively are reached when the gain reaches 15 dB below maximum. Then the curve takes a steep turn upwards until it hits the somewhat indistinct clip limit.

Summarizing the results of the distortion lab tests, the TM-30 delivers some very impressive results in the THD category. Even at high gain settings for the preamp, the THD value is –95 dB, 10 dB better than the official data supplied by QSC. Less inspiring are the transient intermodulation distortion characteristics, with the TID rising sharply when the preamp is only at 15 dB below maximum gain.

There isn't much we can say here, however, about what the underlying cause of these results might be.



Parameteric filters *with high and low shelf (above) and high- and low-pass filters (below) on the inputs. The high-pass filters use a slope of 24 dB/octave, the low-pass filters 12 dB/octave. The frequency can be set to between 20 Hz and 2 kHz, or from 2 kHz to 20 kHz respectively. (fig. 15)*



Parametric EQ *band on input and output, settings for the EQ band are frequency (above), Q and gain (below) (fig. 16)*

pattern for both settings. The TouchMix's analog outputs (Fig. 8) deliver an SNR of 110 dB at linear weighting and 112 dB with an A-weighted noise component. The maximum output voltage is +21 dBu. The headline here is that the measured noise values fulfill or exceed QSC's official data for the TM-30 on both the input and output sides.

Filters

Each input channel on the TM-30 has an adjustable high- and low-pass filter as well as four fully parametric EQ sections. These can all be set as bell-shaped filter bands, or one band can each be set to act as a high- or low-shelf filter.

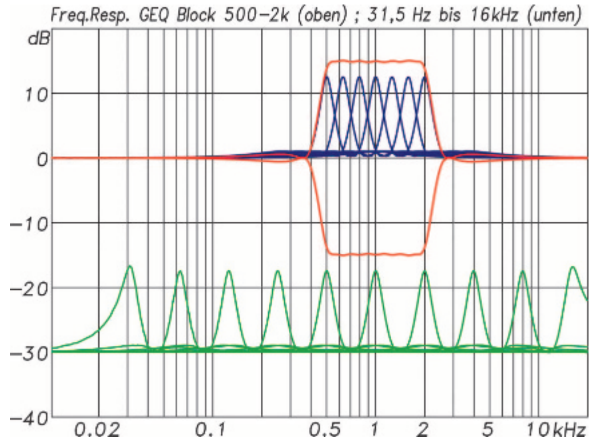
The parametric EQ bands have the following parameters:

1. Low-Shelving:	20 Hz to 20 kHz ±15 dB
2. Bell-shaped mode:	20 Hz to 20 kHz ±15 dB
	Q = 0.4 - 4
3. High-Shelving:	20 Hz to 20 kHz ±15 dB
4. High-pass filter:	20 Hz to 2 kHz 24 dB/Oct.
5. Low-pass filter:	1 kHz to 20 kHz 12 dB/Oct.

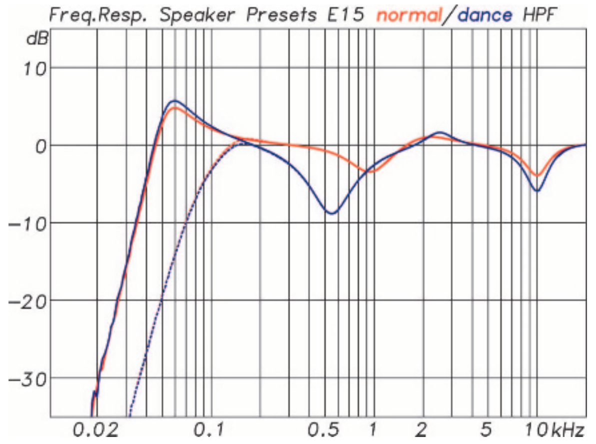
Fig. 15 shows an example measurement for the shelving filter and the settings for high- and low-pass. In Fig. 16, we can see the options offered by the bell-shaped EQ bands. The important thing to take away from Fig. 16 is that the filter curves remain the same across the entire frequency spectrum. This shows that something is acting to compensate for the transition from the analog domain to digital. Without this compensation, the bell curves would become narrower the further we look higher up the frequency range, assuming the Q factor remains the same. If we compare the two red curves for 40 Hz and 16 kHz, we can see that they are identically shaped because of this compensation.

The curves, however, are not shown completely accurately on the display. The measured filter curves for the parametric EQ bands have a larger Q-factor - i.e. are narrower - than shown on the display.

Parametric filters can be activated either together or individually, making it easy to compare signals with and without a particular filter. In "Simple" mode, the only controls available for the parametric filters are Gain and Frequency settings, as well as the on/off button for the high- and low-pass modes. Any changes made in normal mode to the Q-factor or the frequency for the high- and low-pass filters is retained. The output side has more extensive filter options. Apart from high- and low-pass filters, which both have a slope of 18 dB per



Graphic EQ, *above shows filters from 500 Hz to 2 kHz individually (blue) and as a block (red), below is shown filter bands from 20 Hz to 20 kHz (green) (fig. 17)*



A library of filters *specific to many QSC models are also included, here pictured is the QSC E15 with several set ups (fig. 18)*

octave, there are another six filters on offer. There's a graphic 1/3 octave EQ, 12 notch filters as well as special filters for adapting to QSC's K and E series loudspeakers (see also Fig. 18).

The interface for the graphic EQ shows only the controls, meaning that the resulting filter is not, unfortunately, available in graphic form. Fig. 17 shows how the 1/3 octave filters actually behave. The compensation across the filter curve works well right up into the high frequency range. The individual filters are relatively narrow and don't overlap much. Adjusting several filters next to each other results in a summed curve that moves around as a block with only a small "ripple" effect. Thus, the fader settings give a good indication of how the filters are acting.

Conclusion

With its 30 inputs and rich feature set, the QSC TouchMix-30 Pro takes pole position among the TouchMix series of mixers. It remains faithful to the basic concept, with no physical faders or channel strips. Instead, the mixer relies on the 10" touch display or any tablets and smartphones linked into the system. If this makes you apprehensive about approaching this console, you can try out the iPad app. This provides an excellent way of exploring the appealingly designed control surface, which gives fast and direct access to all parameters. This aside, the TM-30 has all the functions it needs as well a lot of well-implemented additional features such as 32-track HD recording or direct connections to your DAW.

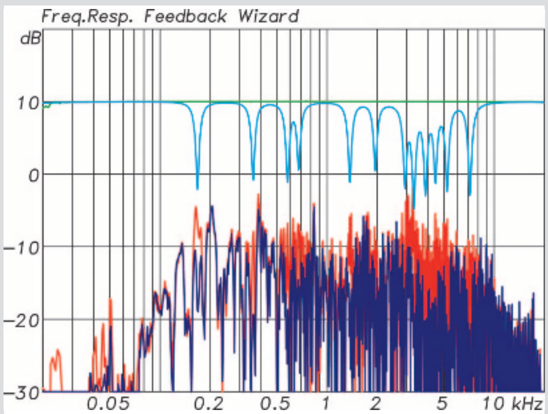
The mixer also includes wizards and presets for its filters and effects that cover a range of real-world situations, making it easier for live sound novices to find their way around and develop a feeling of security using it.

The results of the audio lab tests are very good across the aboard and compare well with larger and more expensive consoles. If the TM-30 represents the implementation of TouchMix users who found that the other models to date were lacking in channel count or feature set, then we can look forward to further developments in this product range. Let's finish off by having a look at the pricing. The TM-30 is priced at €2,379 MSRP incl. VAT, with an optional transport bag, dust cover and tablet stand (shown) available for a bit more.

Anti-Feedback Wizard

Each of the 16 output channels of the TM-30 has a filter bank with twelve notch filters. In addition to the frequency, several controls are available for each notch. The Q factor can be set to between 10 and 30; gain settings from 0 dB to –20 dB are also available.

The Anti-Feedback Wizard *automatically looks for frequencies causing feedback and activates filters to counteract them (fig. 19)*



Open loop gain *and notch filters activated by the Anti-Feedback Wizard (fig. 20)*

The filters can be used to reduce feedback in a live sound scenario, or employed in a control room to suppress annoying room modes in the low frequencies.

QSC have included a software assistant to help the user get feedback under control. The assistant monitors the signal in real-time for potential feedback problems and alerts the user to these frequencies. In Manual mode, the user has two options: set the notch filter by hand or simply press the Manual Kill button, which activates a filter to act on the frequency for the feedback.

As an alternative, users can also launch the Feedback Wizard, which gradually raises the gain on that channel and inserts a filter on any feedback as it arises. The only filters available to the Wizard are the ones still set to 0 dB. This lets you adjust some filters yourself and then leave some filters to be activated by the wizard, or the other way around.

Fig. 20 shows a typical filter curve as set by the Feedback Wizard. The red/blue curves show open gain loops for a microphone/speaker set-up. As soon as the loop gain reaches 0 dB at any single point and the phase relationships are in the correct configuration, feedback is generated. The loop latches on to the frequency with the highest gain and the typical whistling sound is heard. At lower volumes, we can already hear a resonance that presages full-blown feedback. For spoken word signals, the loop gain should never exceed –5 dB at any frequency. For music, that "never exceed" value is –12 dB. As we can see in the measured response in Fig. 20, it's often peaks at certain frequencies that cause the problems. A narrow notch filter is used to reduce gain at the corresponding points. To avoid removing more of the original signal than needed, the notch filter should be narrow and not used too aggressively. Normal 1/3 octave EQs or bell-shaped filters aren't very useful here, as they remove too much of the original signal. Although we might be able to combat some nascent feedback and raise the gain somewhat, the portion of the signal we actually need has taken a hit in terms of volume. The light blue filter curve in Fig. 20 was created using the Feedback Wizard, which reduces the volume at critical frequencies, cutting off those dangerous peaks. As we can see, not all peaks are attenuated. This is due to the signal's phase, as not every amplitude spike automatically causes audible feedback. All in all, this curve looks pretty healthy. Don't forget, though, that if the situation changes, such as a movement in the mic position, the critical frequencies can change quickly, and the notch filters no longer do their job. The process as described works well with fixed microphone positions, such as lecterns or podiums. The correct filter adjustments can yield a few extra dB.

Before we all get too euphoric, however, let's remember that the most we can expect to achieve in real terms by using special filters is around 2 to 3 dB of gain. But it's this small increase that can sometimes make the difference between clear and incomprehensible speech.

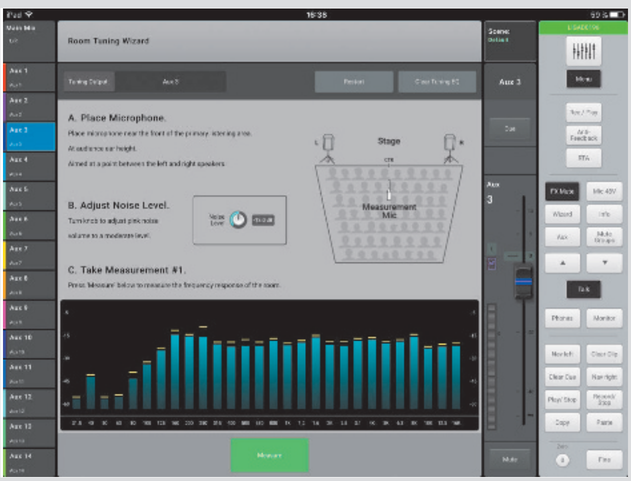
RTA Functions and Room Tuning Wizard

The TM-30 can run two real-time analyzers (RTA) simultaneously, which can then be displayed on the console or a tablet. The smartphone apps don't support RTA data display. RTAs are activated by the pressing the corresponding button. Users then select a signal source, choosing from the Main outputs, any Aux outs, the Talkback input or the current Cue signal (Follow Cue). Independently of this function, an RTA can also be called up together with any EQ type at any point in the signal flow, letting you monitor directly the effect any filter settings have on a signal.

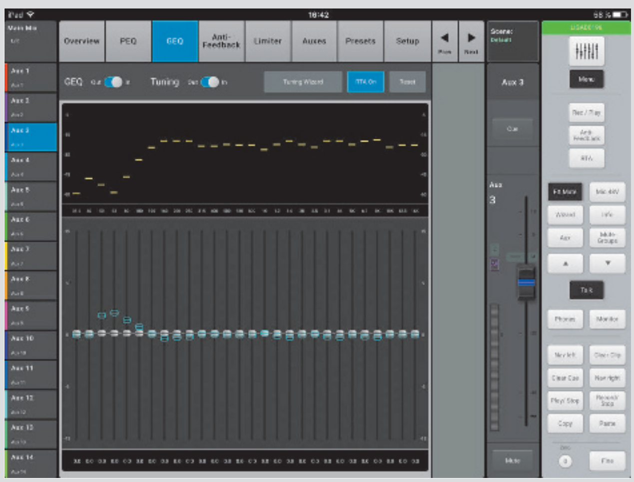
The Wizard button accesses a range of wizard functions for FX settings, Input gain and the Room Tuning Wizard. The latter filters the PA signal, employing an RTA as a measurement and calibration tool. This requires a good measurement microphone connected by cable to the mixer's Talkback input. Next comes the selection of the output signal to be analyzed and corrected, as well as a measurement method. The selected method determines how many measurement positions are used; the results are then averaged. The calibration signal used here is pink noise. Its volume is set using the Noise Level control.

When the data has been captured for the required number of positions, the Wizard sets a graphic EQ to achieve a linear signal. In addition to the Tuning function, a second EQ is also available, the User EQ. The User EQ can be set to Flat, Live (using a predefined set up for live sound reinforcement systems) or Keep, which flattens the system response but retains the User EQ settings untouched.

How well these kinds of wizards work also depends on external factors. It's a good idea to use as many measurement positions as possible within the space covered by the speaker(s) in question and then average out between them. Any noise that can disrupt measurements should be avoided. We recommend not to use this function for the first time just before a live show, but rather to try it out on some sample speakers first in a store room or other location to get a sense of how the filtering on the TM-30 works and what effect it's having on the sound. Having gathered some experience with these functions, users can then store their own EQ presets for certain scenarios on the mixer.



The Room Tuning Wizard *automatically measures and adapts to the sound coming from the PA (fig. 21)*



Settings *for the graphic EQ as suggested by the Room Tuning Wizard (fig. 22)*