WHITE PAPER

Aircaster M12 Review

An independent engineering assessment of a hybrid broadcast mixer

About Geoff Archbold

Geoff is one of the country's most experienced radio studio engineers. He has worked in broadcast audio for over 30 years, working for Soundcraft, Philip Drake, Clyde Electronics and latterly Clyde Broadcast.

He has worked extensively with mixers from Lawo, Studer, Yamaha, Klotz Digital and of course Clyde, as well as playout systems from RCS, Netia, P-Squared, Dalet, Clyde and many others.

As a systems engineer he has worked with all of the UK's leading commercial radio stations, running both large and small projects, in the UK and overseas.

He is currently retired, undertaking the occasional consultancy project.

Geoff has worked with the KMS founders in the past; this review reflects his own testing and opinions.

Introduction

The Aircaster M12 from Klotz Media Solutions (KMS) is billed as a 'Hybrid Broadcast Mixer'. I have worked with the people behind KMS for many years, notably Thomas Klotz and Phil Collins, and I must confess I was a little surprised at first to see such a product from them.

The M12 is a modular mixer, which KMS claim is suitable for both live assist and production applications, in situations where there is a DJ in control, or where there is a technician operating the mixer with talent in an adjoining room.

The brochure claims 'Digital Flexibility: Analog Simplicity'. We'll see.

Construction

The M12 can accommodate up to 12 input modules, of which there are three types to choose from. To the right of the mixer are Monitor and Meter modules, which cannot be re-positioned, but input modules are interchangeable and can be fitted in any combination.

At the top and bottom of each module are two small screws; the black ones secure the modules into the frame, the silver screws can be undone to about 20mm, and can be used as handles to withdraw modules, removing the need to pull on the pots.

Modules plug into a rigid backplane pcb, and locate very positively – no need to fish around for ribbon cable harnesses.

At the back of the mixer is a cover which simply lifts off to allow access to input and output connections, which are directly into the modules themselves.

To keep the height to a minimum, to maximise sight-lines to playout screens, KMS supply low profile ¼" jack plugs and XLRs with each module, or as a cost option can provide break-out cables to standard-sizes in-line connectors.

Power supply connections, guest headphone outputs, external meter outputs and red-light outputs appear on the rear of the mixer.

There is provision for two external power supplies, offering a high degree of resilience as PSUs are notorious for failing.

A legend strip at the bottom of the modules accepts labels and I understand that KMS offer a Word template for customisation.

Modules themselves have pretty standard U-shaped top panels, silk-screen printed.

Input modules use long throw Alps faders, mounted on pcbs with plug-in cable assemblies, for ease of replacement.

As far as I could see, Neutrik connectors are used throughout, together with sealed push-button switches. All switches have led indication. Pots are typical of lower cost mixers, and are possibly the weakest link.

The mixer is designed to sit on a table top, and has quite a low profile – approximately 2", with the rear connector cover taking it up to 4.5". The width of the connector cover is suitable for mounting a playout monitor or keyboard.

Input Modules

The three types of module, mono, stereo and phone, each have two user selectable inputs, a gain control, routing to individual PROGRAM and RECORD mix buses, a PFL button, fader and 'Peak' led indicator.

The mono module has selectable mic or line level inputs and includes a 3-band equaliser, with in/out switch. I prefer this arrangement to centre-detented pots as it allows instant comparison between in and out.

Under the connector cover the mono module has two further switches, which can only be accessed using a paper-clip or small screwdriver.

One of these switches allows selection between dynamic and condenser microphones – reducing the fixed gain and enabling phantom power for condenser mics, allowing mic types to be changed without having remove modules to access jumper links or navigate through configuration menus in more sophisticated products.

The second 'hidden' switch selects whether the associated mic is located locally, in the control room, or in a separate studio, controlling appropriate loudspeaker muting under live mic conditions. (I think these are very useful features, particularly for broadcasters with limited in-house technical support).

Listening on headphones to the equaliser confirms a useful selection of turn-over frequencies allowing a good range of tonal adjustment.

Over and above the preset gain for condenser or dynamic mics, the front panel gain control allows user adjustment over a 20dB range, sufficient to cater for a reasonably wide range of vocal outputs.

The Peak led illuminates pre-fader and glows progressively brighter as signal level increases.

Finally there is a MIC ON led, which illuminates when the fader is open, but goes out if the external Cough button is pressed.

The stereo module has two inputs, one for balanced line level analog inputs, the other for USB (digital) inputs – negating the need for external soundcards – saving a lot of potential configuration time and cabling.

These modules don't have equalisers, but with so much content originating from PCs, I don't see this as a major disadvantage.

There is a momentary action START button and led indicator, which sends a contact closure to an external replay machine.

There is no fader start.

Initially I thought this to be a bit of an oversight – particularly as so many 'on air' mixers traditionally incorporate large illuminating START/STOP ON/OFF buttons on inputs.

However I am aware that increasingly content from PCs is played from either a mouse action, touchscreen or dedicated button box, so maybe it's not so important.

A reasonable criticism is the small size of the buttons, but in fairness they have a positive action and making them larger would have considerably increased the mixer's footprint. At least they all have associated led indicators.

The phone module also has two inputs, one for connection from an external TBU, the other a USB input. Each input has an associated cleanfeed output, with a momentary action Talkback button, allowing communication with callers before putting them to air.

My Android mobile phone connected via the USB connector without any issues – removing the need for an external hybrid. I also tried a connection to/from a PC, easily connecting to WhatsApp.

I understand that iPhones don't offer such USB connectivity, but KMS offer a simple cable interface which allows an iPhone to be connected via the analog inputs.

Direct mobile connection via USB saves the cost and inconvenience of an external hybrid.

The phone module has a two-band equaliser, with in/out switch, also allowing reasonably effective tonal control of callers.

One interesting and in my opinion very useful feature on the phone module is a switch which

allows the cleanfeed to be derived from either the PROGRAM or RECORD mix buses, supporting both live phone-ins and pre-recordings.

The M12 can support two phone modules, allowing two callers to go to air simultaneously.

All input module analog connections are electronically balanced, as are the cleanfeed outputs.

I would have liked to see a DIVERT button for remotely controlling an external TBU, but if a mobile phone or PC is being used this is not a problem.

Monitor Module

This is located on the extreme right of the mixer.

There are separate selectors for control room and guest/studio, each offering a choice of PROGRAM or RECORD mixes, or an EXTERNAL input. The External input has a level control, allowing the output from an off-air receiver to be adjusted.

I would have preferred this input to be balanced, but if the off-air receiver is local it should not be a problem.

Each set of selectors feed separate speaker and headphone volume controls, with control room outputs being stereo and studio/guest outputs being mono.

The control room speaker outputs are unbalanced – again no issue with local speaker amps, and the studio speaker feed is electronically balanced.

There is a momentary action talkback button for communication with guests and another for comms with a second studio (which also has a return talkback feed), and loudspeaker outputs automatically mute whenever mics are live, depending on their location.

The monitor module incorporates a talkback microphone with an automatic gain control amplifier. This is useful in tech-op situations where there may not be any mics in the control room. The default setting is to use the internal mic, but any mic channel may be set to be the source of talkback according to internal pcb jumper links.

Finally this module contains two multiway RJ45 connectors, one for connection to an associated Talks Studio, the other to a second On Air studio.

Meter Module

For some reason this is marked as the 'Mixer' module. This contains the balanced PROGRAM output connectors (XLRs) and analog RECORD outputs (unbalanced).

The module also contains a USB soundcard, with a user selectable output from either the PROGRAM or RECORD mixes, and an input which can automatically inject audio into the control room monitoring.

The PROGRAM output can be used to directly feed a streaming PC, or the RECORD output can be used to feed a recording app, like Audacity or similar.

If the studio playout system provides a dedicated CUE/LIBRARY stream, this can provide a very user friendly way of previewing playout content without any mixer actions – when a library item is played the monitor system automatically switches to this feed (which has a user adjustable level control).

Metering is by stereo bar-graphs (24 leds per side), with a 2dB resolution over the more useful range. Four different colour leds are used, the idea being to keep within the green led band to maximise level without risk of distortion.

Buttons allow selection of either VU or Peak meter ballistics, the latter having a faster attack and slower release compared to the VU characteristics.

A peak hold facility is also provided.

This arrangement is quite sensible, allowing users to select whichever type of metering is most suited to their program format and content.

There are also unbalanced buffered meter outputs available via a TRS mini jack on the rear face of the mixer, for connection to external meters.

At the bottom of the module is a bank of STATUS Indicator leds. Aside from the PSU indicators, I'm not really sure how useful the other leds may be, but I suppose for novice users an indication of what is going on internally could be useful – local or remote speaker muting, control room DIM (under outgoing talkback conditions), incoming (reverse) talkback audio and Cue audio presence.

GPIO

Mono inputs have TRS mini-jack REMOTE connectors. These provide outgoing MIC LIVE commands and incoming TALKBACK/COUGH inputs, for use with external Guest Cough panels.

These panels have a single momentary action red illuminating pushbutton.

With the corresponding fader fully down, pressing the red button activates the channel pre-fade mode, allowing the guest to speak to the mixer operator, off air.

When the fader opens, the button illuminates – indicating a live mic condition. If the button is pressed with the fader open, the audio in that channel mutes, acting as a COUGH button – with the MIC ON button on the channel flashing.

This is a very simple interface but one which offers very useful functionality – particularly in Talks programs, and is augmented by the optional Mic Arm indicator ring, which can be attached to any mic stand.

The stereo module has an outgoing start command, which can be set to maintained or pulse action, using internal pcb jumper links.

As stated previously, the lack of fader start and large buttons may be considered a limitation.

On the rear face of the mixer are two red light outputs for Control Room and Studio. These also use TRS mini jacks and provide both contact closures AND power for external indicators, activated by any control room or studio mic live condition, so no need for extra external PSUs.

KMS offer a wall mounted mic live indicator, with different display options, powered directly from the Aircaster, so no need for external supplies and relay interfaces.

Finally the multiway Talks Studio connector contains dry contacts to activate a remotely located display for interface to an existing indicator.

Performance

The Aircaster brochure provides an unusually detailed set of 'typical' performance figures, and in my opinion more importantly explains the measurement conditions.

I could see no point in repeating all measurements on all channels, but did make a number of readings on what I consider key parameters on randomly selected modules.

I used my old faithful Lindos LA101/102 test set, without which I would be so lost. My evaluation mixer had 10 input modules, four mono, four stereo and two phone.

Noise

(Measured using rms characteristics, with reference to 0dBm).

With all faders down, without any routing to the PROGRAM bus, the output (mix) noise was -90dB (20–20kHz) and -83dB (2–100kHz).

With four stereo faders fully up, set to unity gain, routed to the PROGRAM bus, I measured -89dB (2–20k).

Mic noise from a 200ohm source resistor at the maximum gain of 70dB setting gave an Ein of -127dB, quite respectable.

Line input noise from a mono or stereo input set to unity gain (EQ out) was -87dB from a single channel.

Crosstalk

(I consider this to be one of the most important measurements, from a practical usage point of view).

Bus Routing 'offness' (bus de-routed) – measured below 90dB at 10kHz, on a mono channel.

Cough switch 'offness' below 93dB at 10kHz.

Fader offness – measured on a stereo channel, 88dB at 10kHz.

Left/right crosstalk on a stereo module – measured -88dB at 10kHz.

Cleanfeed rejection (phone channel) – measured -79dB from Program mix, -85dB from Record Mix, both at 10kHz.

Monitor module control room muting – measured -84 and -85 at left and right outputs at 10kHz.

Below 10kHz crosstalk was significantly lower, in many cases down in the noise.

Frequency Response

Measured from mic input to PROGRAM output at all gains, worst case -0.5dB at 20Hz and 20kHz.

Line inputs the same.

Control Room LS outputs measured -0.3dB at 20Hz and 20kHz.

Distortion

Measured from a mic input at a PROGRAM output, at a level of +20dB at the output – less than 0.1% THD at any gain setting, 100Hz–10kHz.

Line inputs less than 0.03% THD at any gain setting.

These sample measurements met or closely matched the published specs, and in most cases exceeded them.

In Practice

Being an engineer, as opposed to a presenter, there were only limited 'user tests' that I could carry out.

As mentioned previously, tone control adjustment seemed more than adequate, and headphone levels (using 32 ohm cans) could be painfully loud.

I did not notice any clicks when activating switches or remote functions. LS muting was silent.

Automatic reverse talkback and Cue audio switching was seamless, with sensible sounding time constants.

My Android phone connected via USB and worked exactly as expected, on and off-air.

I connected four stereo channels to a PC via a powered USB hub. Three channels were

connected to different playback streams from Zara Studio Playout software, the fourth from Audacity; the outgoing record stream was connected to Audacity's record input.

Everything connected easily, and remained stable.

It was easy to send Zara streams to the Program output and Record mixes to Audacity whilst playing out from Zara; the auto-cue function was particularly useful for playing back recordings without the need for any mixer switching.

In Conclusion

There is much to like about this product:

- Ease of interface via USB and analog connections
- Modular construction easing maintenance
- Familiar user controls – 'menu-free'
- Easy plug and play operation
- Excellent audio performance
- Free from software updates and instant start after loss of power – no boot time
- Built-in soundcards
- Suitable for both on air live assist and production
- Suitable for tech-op or self op styles of operation

On the down side:

- Buttons are on the small side
- No fader starts
- No large dedicated ON/OFF or START/STOP buttons
- No PAN or BALANCE controls on input modules – though I suspect that in practice these are very rarely used
- Connectors directly into modules – some may find this annoying
- An analog based console may be considered a backwards step

> *Some will see that as a backwards step. Having tested it, I don't. The connectivity is thoroughly modern, it installed and ran without any configuration, and it measured as well as many digital consoles I have worked with. For stations without in-house engineers, a mixer that needs no software updates, recovers instantly from a power cut and can be repaired by swapping a module may be exactly what's needed.*

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by Geoff Archbold, independent engineer