Saturday, 26 December 2015

The Engineer's Bench Podcast; Audio top tips

Introduction

We wanted to have three sessions of the podcast with some top-tips for the new year; useful tricks that allow the jobbing engineer to solve problems that we've come across over the years; this list is by no means exhaustive but all these tips are things that have saved our bacon!

This time it's audio, but coming up we've got video and data/power - that last one seems like a strange but generally if you're doing a lot of work in data centres or comms rooms the thing you have to deal with alongside fibre and copper data is the mains.

1. AJA HD10AM/AMA – SDi AES problem solver

https://www.aja.com/en/products/mini-converters/hd10ama

The HD10 is an audio de-embedder/re-embedder that includes an HD/SDi DA. For £500 it's worth having a spare in the workshop as it is a real problem solver for selectively changing channels within an HD/SDi signal or shuffle channels etc. It is only a single channel device - you can't de-embed from one SDi signal and then re-embed into another which is why we refer to as a "re-embedder" but for swapping channels 1 & 2 with 3 & 4 (for example) it is just the thing. I've used them in football coverage where four channels of audio arrive (stereo mix on one & two with clean M&E; stadium atmos really on 3 & 4) and four have to leave with an alternative commentary on 1 & 2 with the same stadium clean atmos on 3 & 4. So - take out 3 & 4 into a local mixer (which also makes headphone mixes) and then the new audio mix with the new commentary gets re-embedded into 1 & 2.

They also do an AES version which has the benefit of eight channels of audio (four AES streams; same XLR i/o cable from the D25 multi-way audio connector on the HD10). I've used this one to "wrap around" a Yamaha 03D mixer (which has AES i/o on a D25 connector) which essentially turns the mixer into an SDi i/o mixer; eight channels in and out from the HD/SDi router.

Show & tell on camera.

Audio MIDI utility in OS-X – audio device management on Macs

Often you have to get audio in and out of OS-X using either the built-in unbalanced stereo 3.5mm jacks or an external USB audio interface. What if you need to feed the same source to two devices? There is a little know utility on the Mac called “Audio MIDI setup” which allows you to combine devices at the OS level.

So if you want to monitor what you’re sending out via a high-quality audio interface using your earbuds and the built in 3.5mm jack you can create a multi-output device (call it something useful like “M-Audio & earbuds”) or what if you need to provide i/o to an application via dissimilar routes but the app only support the default audio device type? Create an aggregate device - in fact that’s how we record Hugh’s audio in this podcast; Skype sends to an aggregate device that goes to Phil’s earbuds AND goes to an external mixer for mixing with Phil’s mic for recording in Quicktime.

Although I’m not religious about computers; I love & hate Windows, Mac & Linux in equal measure! You can’t do this kind of stuff outside of OS-X.

Formula for passive audio pads ACCURATE!

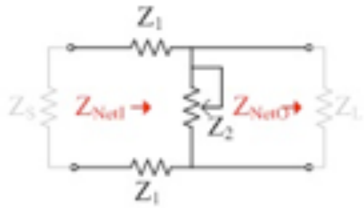
http://philtechnicalblog.blogspot.co.uk/2013/02/passive-low-value-audio-pads.html

I've often had to knock up audio attenuators to make music gear (which is typically +4dBu for zero level against 0dBu for broadcast) and my usual m.o. is to approximate everything around a 10k potentiometer; typically 100ohm sending impedance, 10k ohm terminating impedance and an H-network for balanced lines and a T-network for unbalanced. You can find numerous examples online.

So, for a variable 1.5 -> infinite pad you need Z1 and Z2 at 390 ohms and a 5K potentiometer. For the last few that I made the 5k pots arrive with their wipers at the centre position and all six that I've made so far have been bang on 2dBs at that centre point. It pays to be precise.

Pad Calculator

Simple Pad		**For Modern Audio Lines**			
Zs	150	Max 300	Min 30		
ZL	20000	Max 50000	Min 10000		
Simple Gain	0.92764				
Z1	390	R1ZoutR1//R3	784.148		
Z2	5000	R3//Zin	4000		
Zo (approx)	754.564			**ZnetO**	**784.148**
Zi (approx)	4880	Vin=	1	**ZnetI**	**4780**
		Vout=	0.80943		
Gain (approx)	0.80943				
Analysed Check	**0.81136**				
dB	**-1.81574**				



$$G = \frac{Z_L}{\left(\frac{Z_L Z_2}{Z_L + Z_2} + Z_S + 2Z_1\right)\left(1 + \frac{Z_L}{Z_2}\right)}$$

$$Z_{NetI} = \frac{Z_L . Z_2}{Z_L + Z_2} + 2Z_1 > 10Z_s$$

$$Z_{NetO} = \frac{(Z_S + 2Z_1). Z_2}{Z_S + 2Z_1 + Z_2} < 10Z_L$$

Why did 70s rock singers tape two mics together?!

I've been enjoying "Guitar Heroes at the BBC" on BBC4 where they compile clips from Whistle Test, Rock goes to College, TOTP etc. I've always wondered why rock singers from a period of only a few years would have two mics taped together. By the time I was paying attention in the late 70's the practice seemed to have stopped so I suppose it was a technical development that made the change.

I asked the question on Twitter and Facebook and got great rock'n'roll answers; "...so they could take it to 11", "early form of stereo recording" etc. In fact when I went back over my old BBC notes I had been told why they did it but only a few weeks out of university I don't think I understood common mode rejection!

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So - having re-read my notes and had a trawl around the web (my word, there is some awful rot spoken by people who know very little!) here are the two reasons (and I'll list them based on the technology that fixed the problem), they both rely on the fact that the two mics are wired anti-phase to each other and the assumption is the singer sings predominately into only one of them (doesn't matter which).

1. pre-compressor/limiters you needed a way of loosing some of the induced stage and line noise - this does it.

2. pre-parametric eq - you needed a way to reject howl-round and this does it.

So - you mix the anti-phase feeds in two channels on the desk and all noise/feedback etc gets canceled and the voice (predominantly coming down one feed) remains. Interestingly another technique to gate a mic is to have either an optical detector on the mic stand or a pressure mat in front of the mic which mutes the channel when nobody is near the microphone.

When is it appropriate to bodge-unbalance an audio signal?

We all know the difference between balanced and unbalanced signals - the principle of common-mode rejection of induced noise on a line doesn't just apply to audio, but it's where we bump into it the most in broadcast. If you're not familiar we'd direct you back to our podcasts on audio principles from a couple of years ago.

Suffice to say there are times when you have a balanced audio signal and you need to feed it into an unbalanced input of some piece of equipment. You can get balanced to unbalanced converters which may be "rep" coils (1:1 audio transformers) which "lift" the difference WRT to earth and give you an output where you can ground one side of the coil and hey-presto, an unbalanced version of the same signal; in fact, in reverse that's how you can balance an unbalanced signal. This image shows just that; a wallbox with balanced lines from the MCR and two audio rep-coils hidden in the box providing phono outputs on the wallbox so that both the PPMs (balanced) and a DVD recorder (unbalanced) can be fed with the same output of the Avid (or whatever).

However - what if the equipment has rep-coils on it's output driving the line? Good quality broadcast gear where (internally) the signal is unbalanced are often set up that way and so you're entirely at liberty to ground the cold pin of the XLR connector and hey-presto the entirety of the signal is now developed across the hot & earth of the connector so you have an unbalanced signal with no loss of level. However - what if the output of the equipment is driving the balanced line via a high-quality (expensive!) audio transformer? Lots of modern gear uses op-amps in unity-gain configurations to drive the +ve and -ve going lines (non-inverting & inverting inputs on the two op-amps). What happens then if you "bodge unbalance" the line? Well, you've shorted one of the op amp outputs to ground and that little bit of silicon sits there getting warm and six months later it burns out. I had just that happen at a big facility where we were driving every audio circuit with Avitel balanced DAs that drove the line with silicon rather than transformers.

Within the year half of my op amps had failed output and it wasn't until I looked at the list of things they were feeding; VHS machines, PC inputs, etc (this was the nineties!) that I realised. So, my habit now is I just use the hot and ground connection to bodge unbalance. You loose 6dBs of level, but for me that's the better way of doing it. Old-school wiremen still have to be reminded though! "Cut-back the black in the phono, don't take it to the sleeve of the connector" is my cry!

Using a spare noise gate for talkback

All small digital audio consoles have a plethora of internal effects and often quite good dynamic effects; compressor/limiters. Inside the Yamaha 01V / 02R / 03D you can assign parametric eq and compressors to each channel and so occasionally I've had to deal with a talkback mic in a small studio control room that is too noisy to be useful; they often get fed to a presenter earpiece or up an ISDN or Skype connection and so a noise gate to keep the line silent whilst the director isn't talking is very useful.

Passive volume control using LogB potentiometers

Often you have to provide an audio volume control where you might not have a piece of equipment nearby that is usable as a monitoring level adjuster. There are lots of sub-£500 “Big Knob” type gadgets but what if you need just a volume adjuster?

A 10k LogB potentiometer provides an excellent solution; you wire the track across the unbalanced audio and then the wiper provides the fraction of the audio signal that is required. The 10k track doesn’t load anything (all audio gear drives from a low impedance source and assumes a high impedance termination) and the LogB character gives a nice smooth change to the volume.

Encoder delay through RF cameras and how to avoid trouble

In a live environment like a football game or concert there is sometimes the requirement for a roving camera to interview people in the stadium or venue sticking the camera up on the big screens and having the mic(s) up on the stadium PA mixer. RF cameras (where there is no trial or optical connection taking the pictures and mics back to the OB truck) put in a GOP of delay into the signal path - typically 12 frames (which is half a second). When you put this up on the PA there is half a second of delay between the pics and the speakers and not even professional presenters can talk against their own voice coming back half a second later! Members of the public clam up and it’s an unusable state of affairs.

I've had audio supervisors accuse me of putting in the delay with excessively long cables behind the PA desk! The only way around this is to run a cable from the back of the camera to the PA mixing desk.

In all areas it's worth remembering that cables can't do some things - delaying a signal more than the speed of light would suggest is one of them!