Austin AudioWorks

The Design Considerations - 1

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The design philosophy driving the Austin AudioWorks CCT3 Headphone Amplifier centers on the elimination of the dynamic audio coloration that occurs in the Amplifier - Headphone portion of the full audio listening chain from storage all the way to the ear.

We applied Current Carrier Technology to create low-distortion gain without feedback while operating in the current domain. For the last year we have been working on a better way to get voltage gain without the aural artifacts resulting from the use of either feed-back of feed-forward as noted in 1909 by Noble Laureate Karl Ferdinand Braun and detailed in Harold Stephan Black's classic paper in 1934[1]. Since then all forms of gain in amplifiers have been subject to these design rules and thus the cloaking, constriction, and veiling of the resulting audio signals when driving a dynamic transducer in either loudspeaker or headphone form.

In its simplest form feedback and all its variations creates new and additional audio energy in the power amplifier when connected to a dynamic interactive load. This new energy works to cloak and veil complex audio information leaving it lifeless. The passion of the sound itself is stifled and transmuted into mediocrity. The passion of sounds the artist laid down is lost, it becomes sterile, and the soul of the music is diminished. Pretty harsh words but when you consider the immense value and satisfaction that good music reproduction can give us they are perhaps appropriate.

The Austin AudioWorks design philosophy has been simple, connect the dots of 40 years of amplifier design experience to employ today's better performing and more consistent semiconductors and focus on the relational aspects of the Audio Reproduction System. As a disciple audio design has good ground rules and has advanced the individual elements (storage, conversion, amplification, processing, and transducers) to very good levels. But the key word just used was 'individual'. We engineer a product in a vacuum that is by itself. We get great measurements that fit our design predictions based on extremely powerful computer simulations founded in over 100 years of rigorous physics.

We listen to a product as part of a collective, that's how it was designed or tested. The sound we hear is a 'Gestalt', a whole experience based on as many accurate clues as possible. The more clues you get to add to the perception, the more credible that perception and thus the resulting desired mind-state. The simple fact of the matter is that reproduced music is an illusion. If we do it right we can 'hallucinate' into another space, in a positive and safe space to share one of mankind's deepest and most profound sensual experiences. It all happens in our heads – and to pull it off we need one heck of a lot of information coming into our ears in the right way.

And the information we seek to do this MUST NOT generate other information[2].
New additional information, especially when it is based on the information we want, confuses the finer calculations the brain makes and disturbs the perception, the illusion, going to that space. Consider an echo, the same information but at the wrong time makes a mess of what you are trying to hear.

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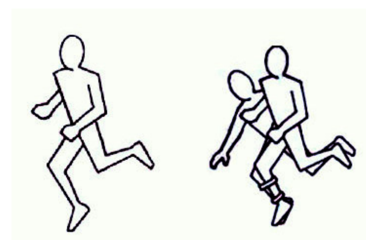
1 H.S. Black, "Stabilized feed-back amplifiers", Electrical Engineering, vol. 53, pp. 114-120, Jan. 1934.

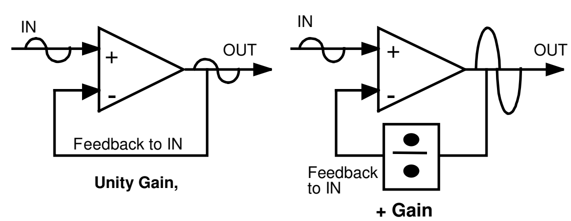
2 As an example consider heterodyning, the mixing of two signals (A and B) together in a non-linear medium and get 4 signals back (A, B, A+B, A-B), very useful in radio design, bad news for audio. By the way, all medium is non-linear to some degree.

In the audio chain the most non-linear spot is the connection between the amplifier and the transducer. It's not the cable, it's what's going on across the cable. It's what is going on in the transducer. All speaker and headphone elements are microphones. They convert energy two ways, from electricity to sound (air displacement), and from sound to electricity. In the vast majority of music storage and reproduction processes, exactly the same physics are used in the recording microphone and the play-back loud speaker. Open up a standard dynamic microphone and you will see a magnet with a gap, a voice coil and diaphragm, and the smallest of wires. Feed some electrical audio energy into a microphone, put it to your ear, and listen to the music.

Electrical processes are practically instantaneous, they happen really fast. Not so in the mechanical world, everything has mass and mass stores mechanical energy and storage means time lost. It is the nature of mass to have inertia; the resistance to a change in motion, and nothing in the Universe is perfectly still, everything is moving all the time. According to Einstein and Mach all mass effects all other masses, so it's a pretty crazy place. And in the middle of this we are trying to get the air around us to carry exact replicas of the air displacements that hit the microphone some time in the past. When you really think about this it is so cool. This information is immensely complex, covers 11 octaves of bandwidth, must be linear phase across that bandwidth, have a dynamic range of 100+ dB, etc, and is enough information to fool our minds even when we know the truth.



When you connect the amplifier to the speaker (headphone) and reproduce music, both their worlds kind of change. It's like the other three-legged race done at office picnics for therapy and team building. Two perfectly good people, each with one leg tied to the other, the leg is free. Two highly functional bipeds become a dysfunctional tri-ped, we introduce a non-linear element in their dynamics, what fun, let the games begin. Seriously though, mixing many things in a non-linear medium creates noise in information terms[3].



Lets follow the path for a minute. An electrical signal (voltage) arrives at the output of the final amplifier in the audio chain. In a Unity Gain stage all the output is fed back so the output equals the input. In a +Gain Stage the output is reduced (divided) so the feedback is less, the amplifier puts out more so that the feedback matches the input. In a power amplifier it is typically about 20 times the voltage of the input. The power gain is different, it is a variable based on the load.

The job of the amplifier is to offer up as much current as it takes to create the right voltage drop in the load. The lower the load impedance, the greater the current to get the voltage, good old Ohms law in action. When the voltage drop in the load is equal to amplifier output then the current stabilizes. Prior to that, the amplifier will do whatever is necessary with the current to get the load's voltage right. The

[3] - The equation for mixing two tones in a non-linear medium and two more tones, the sum and the difference. For example, mix 500 and 600 Hertz and you get 500, 600, 100 (600-500), and 1100 (500+600) Hz. And yes you can get –100 Hz (500-600) but that requires headphones or speakers that work in negative space, I have no clue as to how to get there to listen them.

ends justify the means in this case, all because Feedback influences the gain of the amplification.

The Power gain is easy to calculate, let's assume 50 ohm headphones that put out 100 dBSPL for 1-volt RMS drive. Ohm Law says Volts divided by Ohms equal Current

1 Volt/50ohms =0.02 Amps (20 milliAmps or mA)

Power is equal to Volts times Current so 1V x 20 mA = 20 milliWatts

But if we are going to hold the voltage constant then it is the current that changes when the impedance of the headphone changes with frequency.

The Power is then Current Squared times the load, in this case

(0.02 x 0.02) x 50 = .02 (20 mW)

We know that an amplifier typically has an input resistance of 10k Ohms with a voltage gain of times 10, so it needs 0.1 VRMS drive to get a 1 Volt out. The same math as above gives us an input power

0.1 Volt / 10,000 ohms = 0.00001 W (10 microWatts or uW)

To get 20 mW output we need 10 uW in for a power gain of 2000. As the load's impedance changes with frequency the Power Gain goes up and down as needed.

Notice that Power equation with the voltage squared function – that's a non-linear function. So let's look downstream of the Amplifier.

Once we have stimulated the transducer with energy from the amplifier, the diaphragm moves taking air with it. The diaphragm stores some of the energy and then a little later feeds it back to the amplifier. You can call this energy Back EMF, or variance in the impedance of the load, it's all the same in the end.

The amplifier reacts to this return signal by feeding it back into the amplifier's input, but all this took time, and the amplifier is busy changing its gain with what is coming in now. Chaos reigns supreme!

And that folks is the dynamic distortion responsible for the degradation of the audio you hear. It gets really very bad when you have hundreds of signals all at the same time. A simple strumming of a guitar cord creates well over 60 signals. Each signal is some energy at a specific frequency and phase, usually harmonics of fundamentals. Any more or less and you have a new sound, not the guitar cord but something else.

It is the result of Feedback in the amplifier and the non-linearity of the transducer interacting with each other in a very real-time only dance. This problem doesn't happen in a non-reactive load (a resistor), only in a reactive one such as a speaker or earphone.

The cacophony of signals and images is totally confusing to the mind. It's what the music becomes and not even fun to listen to, all that survives is the beat so it all depends on what you want out of your music, its soul, or a mere shadow of it.

That is what stifles the Eargasm and you won't have one until you fix the problem. And to solve that problem we have created the CCT3.