Understanding Distortion in Analog Amplifiers
A White Paper by Roger Paul

Whether you are open-minded or skeptical, I hope to show how the technology needed to enable you to listen to music (or any sound event) **in a new way** is not only possible but exists today. Designed strictly from theory, it has already been past the "proof of concept" stage. I was not going to release this white paper until I could confirm 100% that this amplifying method works. It does. I have had my critics in the past for using non-standard terms to describe how my designs work. It is the only way I think which is outside the box. The secret for how to do this will be described in a way that might seem foreign to you. It might seem like science fiction – but it is science fact. By the end of this article I hope to have most engineers and those critics back on the same page.

The Quest

I have searched for years for a way to eliminate distortion in analog amplifiers. Years ago the great Ray Dolby found a way to reduce noise. It has taken decades more to find a way to reduce distortion. Why? Mostly because it is more difficult to remove distortion than to remove noise. No real attempt has been made to properly troubleshoot the literal amplifying process as it relates to audio. Some would say this is a moot point since modern technology has long since conquered the bulk of distortion – reducing it to levels where it no longer matters and is seen as "good enough" or "you couldn't tell the difference" if you were to reduce it more. How naive of mankind to think we know everything there is to know about distortion. If we did – why didn't we eliminate it?

Where we are today - State of the Art?

Almost all present day high-end amplifiers are all playing in the same ballpark when it comes to really great sounding systems. (I will not make the argument for such things as "tubes vs transistors" etc. I have worked in both worlds). The amazing fact is that the acceptable "ballpark" distortion levels are actually very high. I won't bother to give you a value because it is irrelevant. Only the best very expensive gear has had the closest approach to "perfection" and still falls short. There is one thing to blame for this. There is a type of distortion that was present all along but remained hidden… until now. I hope to shed new light on the subject of amplifier distortion and a concept I developed to deal with it. The problem was extremely difficult to pin down because of its size. After much work in essentially a new area of circuit behavior analysis, a **true breakthrough occurred**, allowing me to solve a decades old puzzle.

A word about Negative Feedback

Been there, done that, doesn't work. Negative Feedback (NFB) is handy but unfortunately a crude method of trying to make the output of a circuit "look like" the input only bigger. Classic NFB is really only good for very slow (low) frequency signals. Locking onto a sine wave at bass frequencies is easy to do because it acts like a servo. However, as you move up the spectrum, NFB becomes more difficult for a number of reasons. If the signal is music instead of a sine wave, the level of difficulty sky rockets. I know there are many who might believe that NFB used sparingly is a good compromise yielding a circuit with good bandwidth, and low harmonic distortion. The problem is that it only has the effect of suppressing (not eliminating) distortion. I was not looking for compromise – I was looking for removal.

I will **not** be showing you charts and graphs of even order or odd order harmonics etc. Most people reading this paper have seen an endless plethora of articles by engineers and other designers describing distortion types by circuits made from different components and topologies. Tubes, transistors, Class A, push pull, single ended etc. We have seen it all for years. (Those graphs are readily available in the old Motorola tube books from the 50's) There are even different camps of audiophiles that have a real preference for one type or another.

Stop and think about how we have settled for this. "How do you prefer your distortion?" That is more insane than anything I'm doing here. So what do you do about what's left? How do you **stop** it from distorting? Is it even possible to stop it? The answer is yes. Forget anything you think you know about amplification and distortion. It does not apply here.

Thinking outside the box

Besides designing audio equipment for years I am also an experienced network administrator, computer programmer and was the top programmer for not only highly successful private sector companies but programmed for the government as well.

Software Modeling

If you follow my logic through this article you will see my use of programming concepts to describe the action of a circuit as a **procedure** like a **subroutine** or **function**. Software programs often have **error traps** built into them to detect if something has gone wrong during the running of a program. Some errors are just "trapped" and reported back as an error number followed by the program "crashing". On the other hand an **Error Handler** is capable of trapping an error and providing a solution to deal with it. Often this is done "on the fly" in the background and unseen by the user. I have made use of this programming "viewpoint" as a way of thinking about electronic amplifiers. When observing circuit interactions, I see certain aspects of a circuit as an **object** with **properties** the way object oriented programs are written. (Note: These "objects" are not referring to circuit components or semiconductor devices).

Mechanical Modeling

There were times I have studied circuit actions in a different medium such as a purely mechanical function having nothing to do with electronic parts. After discovering problems with a (virtual) mechanical model of my circuit, I was able to convert the “mechanical fix” into an electrical equivalent and apply it to the circuit successfully.

Optical Modeling

To address other issues I turned to optical modeling. I was trying to find a way to “magnify” a signal as if it can be **seen**. (except without the distortion and impurities of a real world glass lens). This is why anyone who has followed anything I’ve done recently often hears the terms “focus” and resolution” because my amplifier designs **literally** have focus and resolution.

The Hearing Process

Since we are discussing amplifiers for the intended use of listening to music at home, it is important to know exactly how we hear. Whether you are in the concert hall itself or in your own living room – the way we hear applies to **both** places. The **phenomenon** is the same. On the surface you can say a series of things “happen” and we perceive the **result** as our ear-brain system **recognizing the original event** that triggered the **initial** flow of “sound waves”. All wave types can be seen as **the movement of energy through a medium**. This might seem like a no-brainer but it is all relevant.

Observing the sonic event from the beginning

I will use a live orchestra performance as an example here because it is rich with many instruments and is among the most difficult sound events to **reproduce** properly.

We see the live performance as a sound event in which there are natural “sound waves” traveling away from the stage. These waves flow freely because they are traveling in medium of air. The music enjoyed by a live audience member seated in the 10th row center comes to him or her by a **delivery system** which includes the air present in the auditorium. The **air** serves as the perfect **medium** allowing the sound vibrations generated by the various instruments on the stage to “travel” as waves **uninterrupted.** The speed at which it flows is known as Mach One. These waves ultimately arrive and enter the listener’s ear canals (**still** at Mach One). What **really** happened just then? Everything that was vibrating on the stage simply increased or decreased the **instantaneous** air pressure as a result of squeezing (compression) or stretching (rarefaction) the **local** air pressure surrounding the instruments. This is nothing new. However, what was **propelled toward the listener** was only the result or **influence** impressed upon, and carried by the medium (a wave of sound). When we say the “sound” has arrived we are really verifying that the result or **influence** has arrived. We call it a “sound wave” because it is in a medium that allows sound waves to exist and flow freely.

For the local listener in the audience, it is “mission accomplished”. Music from the stage was delivered or “streamed” directly to the listener’s ear canals using sound waves to communicate the event through the medium of air. The local air molecules surrounding the (violin string) are still there. They never left the stage. Only the exact disturbances in pressure have arrived, to be **decoded** by the ear-brain system and **recognized** as a **real** live event.

Picture a microphone (Transducer-1) placed where the listener sits. As the continuous waves of sound strike the diaphragm, they are converted into an electrical **“equivalent” of sound waves**. As long as there are no problems handling the **electrical version of a sound wave**, we should be able to reverse the process later on at the speaker (Transducer-2) and turn the “**equivalent” sound waves** back into **“real” sound waves**.

Once we place our viewpoint from within the electrical world, we now refer to the “transducer converted sound wave” (that came in from the outside) as an electrical signal. This “signal” is a **representation** of the (outside) sound wave. The electrical world **begins** on the back side of the microphone **at its output terminals**. This still sounds very basic but I am going through great lengths to impress the significance of this event.

Here is where we go off in another direction…

If we go back to the air medium and **view the sound wave as an object** we then can say that this **object** is a **representation** of a sound **wave**. The same way the “signal” in the electrical world is a **representation** of a sound **wave**. The only thing we’ve done is create a common **wave object** that allows us to study the same **phenomenon** in two different mediums. This allows us to refer to an existing object that has the same function in one medium, from within another medium. It essentially acts as a translator so the two mediums are on the same page.

Here is how to refer to the new object –

It’s not a sound wave (that only lives in air)
It’s not a signal (that only lives in the electrical world)
It is a Wave Object (with specific properties representing the wave phenomena)

A wave object can exist in both mediums. It is an intangible “notion” of **influence** that can still be treated as an **object with properties**. The wave object is an **influence class object**.

When the medium used to convey information is air, we can treat it as a programming object as well and refer to specific properties. Important properties of air (as they relate to this project) are its astronomically high impedance and ultra low mass. If you carefully trace the events in chronological order (like an event log) you will see that because adjacent air molecules all have the same property, any disturbance in pressure simply flows to the next segment of air in a **continuous** motion and without any corruption. The free flowing **wave object** does not have to morph or change shape for the entire trip because (when you are at the venue) there is nothing but pure air between the orchestra and the ear canal. Air is the connection or conduit that allows this information “streaming” to take place. It is a **linear transfer** and does not distort.

Mankind’s attempt at Sound Reproduction

What I am describing may seem very rudimentary like “Sound 101” but the underlying events go much deeper and are the findings of my life’s work so bear with me. It is important to follow these events no matter how basic you might think they are.

The “Streaming” method

When you listen to music on your “stereo system” at home, you are forced to listen to the original venue as it is delivered to you through more than one medium. There could be significant delays or a break in time (like the recording was done a year ago). However, when you rejoin the “streaming” method – your impression should be that it is happening now.

This body of work assumes a desired “stereophonic” view (from 2 mics) I only talk about 1 mic for the sake of simplicity and may refer to a violin that represents the entire orchestra. Picture the path between the performance (on stage) and your ear canals (in your living room). This path can be shown in 3 basic segments. The first segment represents the medium of air in the auditorium the night of the performance. The second segment represents what happened once the (violin) vibrations strike the microphone diaphragm (transducer 1). The microphone allows the sound waves from the venue to be **transformed** or **converted** immediately into a form of energy to be handled by the **second medium** which is the **electrical** world. (I am of course making a blanket statement about all the electronics. Other “non air” events take place as well when you consider storage and retrieval from disks / CD’s etc.) To reach your ear canals, the third segment uses the medium of air like the first segment but involves the **opposite** conversion of electrical signals. These signals are transformed **back into sound** vibrations by your loudspeakers (Transducer 2). Nonetheless the first and third segments use the same medium. Here are the 3 segments…

1. Air - allowing the transfer of **wave objects** (sound waves) from Violin to microphone.
2. Electrical- the embodiment of all electronics situated between the microphone at one end and speakers at the other.
3. Air again – allowing the flow of **wave objects** from speakers to your ear canal in your living room.

We can reduce these 3 segments down to only two medium types (Air and Electrical).

The Problem

Allow me to state the obvious… The middle **electrical** section does not act like the other two **air** sections. This is because **in the electrical world there are obstacles** that impact the smooth flow of information carried by the **wave object**. Remember the wave object exists in both medium types. **The AIR medium has no obstacles**. When an obstacle is encountered in the electrical medium, it has the **reactionary** effect of **changing or morphing** the **shape** of the wave object**.** The object itself still functions but 2 of its properties have been altered. This damages the **integrity** and **stability** of its payload (the ongoing, streaming information).

To be more precise, only **one** of its properties has been altered by the obstacle, the second property changes because of its relationship to the first property. Like a see-saw, when you pull down on one end the other end goes up. This is not to imply that their relationship is "opposite" from each other, just that if you tamper with one property – your tampering with the other.

Wave Object Morphing

The wave object that represents the sound wave in the air medium **never encounters obstacles** in its path (ideally – assuming line of sight path to microphones) and therefore no damaging alterations take place. Wave Object Morphing only happens in the electrical world and only when it encounters an obstacle. Obstacles can be any **physical** object that is found in the circuit like the base of a transistor or any circuit configuration that appears in the path of the wave object. Depending upon what it is exactly, the wave object must change its shape in order to successfully communicate the wave information to that spot in the circuit. The changes may be in the form of an increase or decrease in impedance, voltage or current levels. The interface between the flowing wave object and the receiving device's input configuration gets translated on the fly and the wave object continues to flow on to the next device in the chain making its way to the output.

Critical Damage to the Acoustic Relativity

The constant sequence of morphing its way through obstacles manages to modify critical information carried by the wave object. This alters some embedded data that may represent an actual location of a sound object (the violin's place on the stage) from getting through. (You can still hear the violin but you are not sure where it is). Because of the **acoustic relativity** the apparent location not only of the violin but sound objects near the violin or perhaps the entire left side of the stage has been artificially "moved". The venue is "seen" through a non-linear lens. The ultimate "projection" of this event now carries astigmatism.

Morphing comes with a price!

When a wave object morphs to accommodate a different interface, its velocity becomes unstable. Note: "hitting" an object does not automatically slow it down. Sometimes the shape needed to complete the transfer causes the velocity of the wave object to speed up. The only thing that is certain it that **morphing throws the speed off**. These tiny changes in speed are in fact a form of **Micro-Doppler Distortion**. The apparent location of objects is diffused or out of focus. A warp in the time domain now exists do to a non-linear transfer event.

Instead of trying to make the output “look” like the input (like classic NFB), we need only to concentrate on the **purity** of its **velocity**. If you can perfectly stabilize the velocity – everything else will fix itself.

The shift in velocity due to morphing is a result of the change in energy required to morph. In other words, if it didn’t have to morph – its speed would be constant. Sound familiar? Yes the wave object back in the air medium does not run into obstacles and so not having to morph means its velocity (speed) is constant. The speed of sound is referred to as Mach One. It is important to note that this speed is not the speed of the medium. The velocity of the medium is zero. Mach One is the speed property of the Wave Object.

The Solution… The Quantum Reference Thread

Amazingly, the solution for correcting the problem is just as tiny but with a level of sophistication never attempted. I have had to develop new circuits with new functions and subroutines helping the main amplifier. I use a new technique that enables me to take advantage of “current fragmentation”. It uses Quantum Physics to produce a tiny “piece” or “offshoot” of current from a larger sample. This tiny reference thread is extremely delicate and hyper sensitive to its environment. Its impedance is well above 90 Gig Ohms and it has very unique property. The tensile strength can be used as a high output, low noise **velocity monitor** to sense relative motion including the easy detection of **Micro-Doppler changes along the Time Domain Axis**. It is not physically possible to place these “Doppler detectors” on the wave object itself, so I place the detectors on all of the physical obstacles. This way when a flowing wave object passes through an interface point, its relative speed is monitored and if necessary, trimmed by a control circuit which can quickly dump red shift or blue shift energy as a countermeasure producing a **tiny time warp.** The time warp is then transferred to the wave object by inheritance. As a result, its velocity is held to a single value and remains pure. No other issues matter like the % of linearity. **If the velocity is pure – the transfer is linear**.

As a result of actively adjusting the velocity property – we cause the other interactive property to react in a way that invokes intentional auto-morphing. Any non-linear (receptor) obstacle is met with a pre distorted time domain. It automatically secures a perfect match at the interface point while the velocity is held dead constant. The delivery system locks on to the velocity and allows an amplifier to become “Doppler Free” with a metered output speed of exactly Mach One.

Because this whole transaction speed is regulated by the Quantum Reference Thread, the control circuit subroutine acting as an **Error Handler** operates in a hyper sensitive, “near digital”, analog mode. The speed at which it can correct the tiniest variations in velocity is so fast, in either direction, it can be seen optically. Velocity correction happens virtually in real time. The Wave Object will still run into physical objects but they have become velocity transparent and have no effect on the steady flow of information. **This forces the electrical Wave Objects to flow through a physical circuit board at one speed - Mach One.** Their speed has become **synchronized** with the speed of OUTSIDE wave objects by the Quantum Reference Thread which acts as an analog velocity reference clock. Remember the maximum circuit speed is way higher. The Wave Objects **payload** (containing "converted sound waves") determines the delivery speed.

Auto Focus

As I found more ways to increase the sensitivity of the Doppler Detectors – the increase in resolution has been proportional. An increase in detection of 1000:1 produces a projected sound image where the **location** of instruments on the stage is 1000 times more accurate. This gives you absolute separation between adjacent sound objects with **unlimited depth of field**.

What this means

This means that for the first time, an electrical medium has been successfully programmed to "act" like air. Without the "Quantum Clock" there is **no regulation in delivery speed**. (That is how standard amplifiers operate). By adding the reference clock – all obstacles become transparent clearing the way for a **single** "line of sight" transmission through (**apparently**) one medium. It retains ultra low mass and matches the **stable velocity** of AIR. It is now possible to "stream" the actual venue directly into your listening room because the electronics have been "cloaked" and taken out of the equation. A clue for how to make a "distortionless" amplifier was built into the evidence of how its payload was damaged. The "cure" for the damage was for the circuitry to have a new property of invisibility to Wave Objects.

Acceleration and de-acceleration can be detected so early on that the **Error Handler** can catch and stop harmonic distortion **before it happens**. Deviations in velocity are not allowed by the **Error Handler** effectively removing the ability of the amplifier to distort. When the input and output handlers are matched to air on both ends, your ear-brain system will think that what it hears was delivered by air alone. You end up hearing **the same thing** the live audience member heard that night. All forms of distortion in analog amplifiers begin as Micro-Doppler Shift. Prevention and removal at this low level yields an amplifier with virtually ZERO distortion.

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