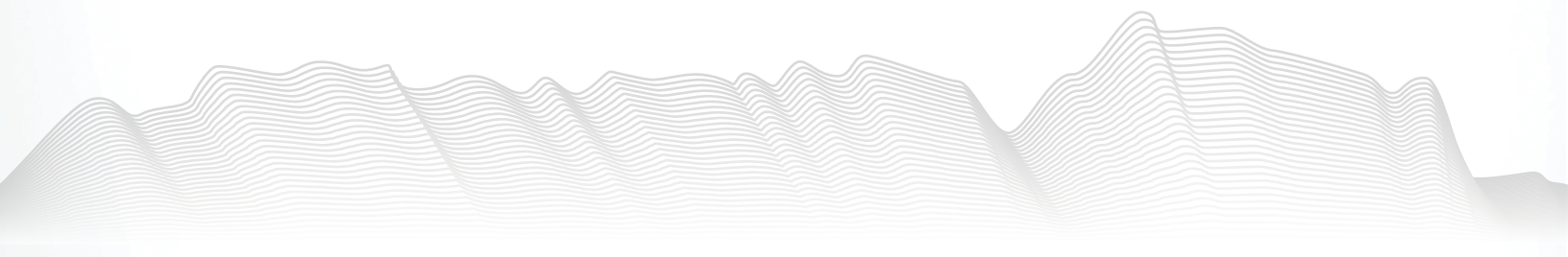




Multipoint Iterative Matching & Impedance Correction Technology (MIMIC™)

A Fractal Audio Systems White Paper | April 2013



INTRODUCTION

We at Fractal Audio Systems are proud to announce a significant advancement in the field of amplifier simulation technology: Multi-Point Iterative Matching and Impedance Correction (MIMIC™). MIMIC offers a level of realism unmatched by any other product at any price-point.

In this white-paper we present the interested user with a simplified explanation of the technology and its inherent benefits.

Guitar Amplifier Evolution and the Paradigm of Electric Guitar Tone

The origin of the electric guitar and the electric guitar amplifier stems from the classic “necessity is the mother of invention” motivation. As musical genres evolved, guitar players found themselves unable to compete with the volume of brass instruments and acoustic drum sets. The electric guitar and amplifier were devised as a means for guitarists to achieve enough volume so as to be sufficiently audible in a live context.

The earliest guitar amplifiers were nothing more than, well, amplifiers. They were designed to simply make the signal louder. The devices available at the time most suitable to this task were the vacuum tube and direct radiation loudspeaker. These early amplifiers were crude and distorted easily when operated beyond their linear range, which was quite small. Unintentionally, and serendipitously, however, the sound of a distorted guitar amp into a limited frequency response speaker became desirable and eventually ushered an entire new genre of musical styles.

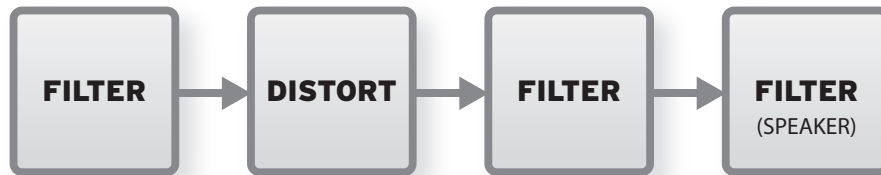
Designers soon began to add features to guitar amps to allow tone shaping and other forms of signal manipulation. These very first amplifiers serve as a foundation for the “fundamental paradigm of electric guitar tone” which is a nonlinearity (aka distortion) preceded and followed by filtering.

The figure below illustrates the fundamental paradigm of electric guitar tone:

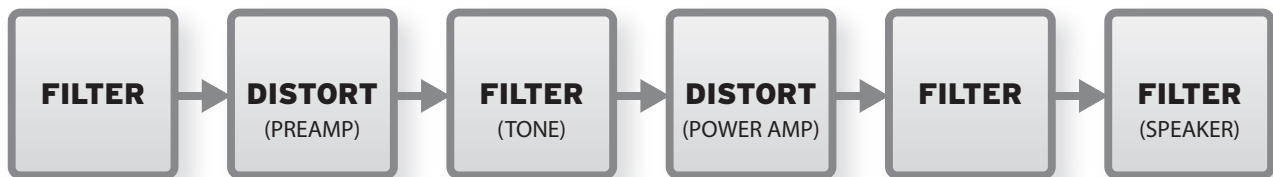


FIGURE 1: THE FUNDAMENTAL PARADIGM OF ELECTRIC GUITAR TONE

We can further elaborate on the paradigm by breaking it into its constituent components, as desired. For example, the output filtering is comprised of the output filtering of the guitar amp and the filtering of the speaker. So a more detailed model might be as follows:



As guitar amps evolved designers added more gain, more tone shaping and more features. A modern guitar amp can have multiple stages of gain and various filtering between the stages. Therefore an equivalent block diagram of a modern tube amp might be:



Amplifier Simulation – Theory and Practice

TUBE AMPS

Before we can get into modeling we need to talk more about tube amplifiers.

Tube amplifiers (amps) are the most popular form of electric guitar amplification. They are cherished for their musical distortion characteristics and dynamic response. While tube amps were never originally designed for these specific attributes (they were originally designed to be as neutral as possible given the technology of the day) musicians were quick to recognize the desirable sonic qualities of an amp being operated beyond its intended operational boundaries. Over the years tube amps have evolved with the express intent of being distortion generators, exploiting the particular nonlinear characteristics of vacuum tubes.

The sound of a tube amp is primarily attributable to the equalization applied to the input signal and the unique way in which vacuum tubes distort when overdriven. Despite their simple circuits, the sound generated by a tube amp is a complex relationship between the various parts of the amplifier.

SIMULATION BASICS

The method used by all amplifier simulators, despite what moniker they elect to use, be it “modeler”, “simulator”, “profiler”, “digital amplifier”, etc., is based on the fundamental paradigm we set forth in Figure 1 above. The most popular amplifier simulators use digital signal processing (DSP) to provide the various signal processing necessary but it is entirely possible to use analog signal processing, and indeed there are several products that do.

There are two basic building blocks used in amplifier simulation: filtering and waveshaping. Filtering, or equalization (EQ), manipulates a signal to change its frequency response. Filters can be implemented in a variety of ways. In digital processing filters are typically one of two types: finite impulse response (FIR) and infinite impulse response (IIR). FIR filters are complex filters suitable for a wide range of filtering tasks and most products use FIR filters (commonly called IRs) for their speaker simulations, sometimes in conjunction with IIR filters. IIR filters are simple but efficient filters that are typically used to implement basic filters such as lowpass, highpass, shelving, etc. filters as well as graphic and parametric EQ.

“Waveshaper” is a fancy name for a nonlinear transfer function. Waveshapers are used to introduce distortion. All amplifier simulators use a specific class of waveshaper known as a sigmoid function. A sigmoid function is a particular class of nonlinearity that is mostly linear near the origin and gets progressively more nonlinear. The name is derived from the shape since the shape of a sigmoid is a “lazy ‘S’”. Sigmoid waveshapers introduce distortion by clipping a signal and are used in amplifier simulators as they most closely replicate the type of distortion that occurs in a tube amplifier when overdriven.

The earliest amplifier simulators used little more than IIR input filters, static waveshapers and IIR output filters. Sound designers crafted the input and output filters in an attempt to replicate the voicings of popular amplifiers and speakers and the waveshapers attempted to duplicate the distortion. Needless to say these early simulators were crude sounding by today’s standards. Much of this was due to the poor speaker simulation. The frequency response of a speaker is extremely complex. There are multiple dips and peaks in the response that are impossible to accurately reproduce using IIR filters.

IMPULSE RESPONSES (IRs)

One of the biggest technological advances in amplifier simulation was the use of FIR filters to replicate the filtering of the loudspeaker. While early products relied on what was basically a graphic equalizer to replicate the complex response of a loudspeaker, FIR filters (or IRs) allowed designers to capture the actual response of a loudspeaker (plus microphone). This afforded a huge improvement in overall realism.

IRs are captured by essentially sampling the loudspeaker. A sample of the speaker's impulse response (hence the name) is recorded and digitized and the data stored for subsequent use. The sample then becomes the coefficients of the FIR filter directly.

All modern digital simulators use IRs in some fashion for their speaker simulation. Many products use IRs and then augment the response with IIR filters to compensate for the lack of low-frequency resolution. To adequately replicate the response of a loudspeaker requires at least 20 milliseconds of response time (typically around one thousand samples). However this requires copious signal processing power and many products are incapable of processing IRs of this length. Truncating an IR to 256 samples (about 5 ms) is common but this severely reduces the low-frequency resolution. Therefore the IR is often augmented with several, or more, bands of low-frequency equalization to correct for the lack of resolution at those frequencies. (Note that the Axe-Fx II can process IRs of up to 43 ms in length and does not suffer from this limitation).

FIR filtering is achieved through a process known as convolution. Convolution can be performed via frequency-domain methods using the Fast Fourier Transform (FFT) or time-domain methods (referred to as direct convolution). Frequency-domain convolution is far more efficient but introduces significant latency and is unsuitable for amplifier simulation. Direct convolution has zero latency but requires significant processing power for even the shortest IRs. Therefore commercial products try to keep the IR length as short as possible in order to keep the requisite processing costs as low as possible.

IRs are now ubiquitous and many products even allow the user to install aftermarket IRs for more tonal choices. The advanced user can even create his or her own IRs using widely available "deconvolution" tools. The Axe-Fx II allows the user to capture IRs using on-board signal generation and deconvolution further simplifying the process. One of the newer simulators even allows the user to capture IRs using test tones injected into an actual tube amp and then recording the response of the amplifier. Unfortunately this approach doesn't allow accurate separation of the amplifier response from the speaker response which limits overall device flexibility and is also subject to significant very-low-frequency inaccuracy due to the absence of system response at those frequencies.

An advanced type of IR processing known as dynamic convolution holds promise for future products but the processing power and storage memory required make this technique prohibitive in consumer products. For example a typical loudspeaker would require many megabytes of data storage whereas a conventional IR can be stored in several kilobytes. These techniques are also extremely input level sensitive and must be carefully calibrated to work properly. Furthermore listening tests have not clearly established any audible improvement over classic static IRs.

DYNAMIC WAVESHAPERS

While IRs achieved a much-needed improvement in frequency response realism, amplifier simulators still suffered from poor dynamic response and users complained they didn't "feel" the same as a real tube amp.

This unnatural response has been addressed in various ways but the most common is the "dynamic waveshaper". In its simplest form the dynamic waveshaper is nothing more than a static waveshaper preceded and followed by time-varying amplifiers. This technique is widely used and is able to realistically simulate the sag and compression of an overdriven tube amp.

Most products now use a single dynamic waveshaper to simulate the entire nonlinear response of the amplifier. The Axe-Fx II uses a highly sophisticated approach that involves multiple triode simulators and a dedicated power amp simulator. It is our firm belief that multiple distortion stages can only be accurately simulated using multiple digital stages of distortion, especially separate preamp and power amp simulations as the distortion contributed by each is unique and the best tones are achieved through a combination of preamp and power amp distortion. Products that use a single distortion stage exhibit "glare" or "smear" in the midrange as the same waveshaper is used to distort the entire spectrum whereas with an actual tube amp operating in its "sweet spot" some parts of the spectrum are distorted by the preamp and some parts by the power amp. Preamp distortion provides greater string separation but can be harsh. Power amp distortion tends to be warmer but can be muddy. A well-designed tube amp balances out these two distortions to achieve clarity without harshness.

PRACTICAL IMPLEMENTATION

All modern digital amplifier simulators use an underlying generic amplifier model (or several base models) that can be adjusted to replicate a desired tube amplifier. These adjustments include various programmable filters, waveshapers and dozens, or even hundreds, of parameters that control the underlying model. These parameters can, for example, control the shape of the waveshaper(s), the amount of compression, the shape of the input filtering, etc.

In operation the signal from the input source (typically an electric guitar) is digitized and fed to the amplifier simulator. The simulator applies input equalization, distortion and then output equalization. The user can adjust various parameters that control the equalization and amount of distortion as well as other pertinent aspects of the simulation. Note that the Axe-Fx II uses multiple stages of distortion with filtering between each stage to accurately replicate the frequency selective nature of a real amp's distortion.

The processed signal from the amplifier simulation is then input to a speaker simulation which is simulated using IRs (and IIRs when necessary). In some cases the user may wish to defeat the speaker simulation if he is using a traditional speaker cabinet and therefore does not desire the speaker simulation.

USER CONTROL

No amplifier simulator would be complete without allowing the user to control the simulation. All modern simulators include gain, tone and various other controls that allow the user to shape the sound.

Most tube amplifiers are unique in that they provide tone control via a passive network known as a tone stack. A tone stack is actually somewhat crude but imparts a particular sonic quality to a tube amp that cannot be replicated using conventional active tone controls (like you would find on hi-fi equipment). Most modern simulators replicate this tone stack to varying degrees of success whereas some make no attempt to replicate the behavior and therefore suffer from lack of authentic control.

The Axe-Fx II has taken tone stack replication to the next level and in Version 10.xx firmware includes 52 meticulously modeled tone stacks. These tone stacks not only replicate the unique sonic signature of their real-world counterparts but even replicate the taper of the controls and their interaction.

LIMITATIONS

Despite all the improvements in amplifier simulation, all modern simulators suffer from various degrees of inaccuracy in comparison to the actual amplifier being simulated. The inaccuracies can be categorized as follows:

1. Deviations in control response, i.e. tone controls don't work the same.
2. Deviations in frequency response, i.e. the output doesn't sound quite the same.
3. Deviations in nonlinear response, i.e. the distortion quality is different.
4. Deviations in dynamic response, i.e. the simulation doesn't feel the same.

For an amplifier simulator to be truly perfect it must address all of these limitations. Fortunately we now have the technology to address these concerns: Multipoint Iterative Matching and Impedance Correction (MIMIC™).

Introducing MIMIC™ Technology

MIMIC is a breakthrough in guitar amplifier simulation technology. MIMIC addresses the four fundamental inaccuracies in simulation through a combination of techniques and renders the resulting simulation indistinguishable from an actual tube amplifier. In the following sections we will cover the four categories and MIMIC's technology in correcting them.

1. CONTROL RESPONSE DEVIATION

The ultimate amplifier simulation is one that behaves just like the amplifier being simulated over all possible settings of the various controls. A simulation that only represents the amp at one particular set of control positions is a poor simulation since the ideal settings for one user aren't usually the same for other users. Furthermore most amplifier controls are highly interactive and more than just simple gain and tone controls. For example, the drive control on many amps changes the amp's input frequency response as it is rotated, typically getting brighter as the drive is decreased. This is desirable as guitars with powerful pickups are usually darker and sound better with more input treble boost. Since the pickups are more powerful the user will typically set the drive lower which then results in more treble boost. Conversely guitars with weaker pickups, i.e. single coil guitars, are brighter and sound better with less treble boost. As the drive control is increased the amount of treble boost decreases.

Deviations in control response are due to only one reason: incorrect mathematical representations. Simulating drive and tone controls is very difficult and laborious. The equations are onerous and complicated, especially for tone stacks. While it is tempting to throw one's hands in the air and use generic hi-fi style tone controls and a generic gain control, we believe this is contrary to the ultimate goal of exactly replicating a tube amp.

1.1 FIXING IT

Unfortunately (for us, not you) there's only one way to fix it: hard work. We exhaustively worked out the equations for all the tone stacks by evaluating the mesh and node equations. Fortunately we have computers that handled the polynomial reduction but it was still a Herculean effort.

We also worked out all the equations for presence and depth control calculations so these controls work like the actual amp (except for the taper in some cases).

Finally we measured the taper of the controls on the actual amps and entered that data. This ensures that the controls on the model behave identically to the controls on the actual amp. Note however that the taper of the presence (and depth) control can deviate from the actual amp. In our tests we found that the presence control on many amps did nothing for the first 80% of its rotation and all the action occurred in the last 20%. We feel that this design anomaly is undesirable and therefore did not model that aspect.

2. FREQUENCY RESPONSE DEVIATION

In Figure 1 we introduced the fundamental paradigm of guitar tone. All guitar simulators are based on this fundamental paradigm and, indeed, all real tube amps are implicitly based on this same paradigm. However, an amplifier simulation will often sound different than the real amp. This seems implausible given the apparent simplicity of the block diagram. There are a variety of reasons why this can happen.

One reason why this can happen is simply poor implementation. The simulation simply doesn't replicate the real amplifier's input and output filters accurately. This can be due to flawed data (i.e. inaccurate schematics) or mistakes in translation (i.e. human error). Either of these will cause frequency response deviations that are detectable to the human ear.

Another cause for frequency response deviation is due to what are known as "parasitics". A real tube amp doesn't necessarily behave exactly like its schematic predicts. This is caused by parasitic elements that are unintentional but, nonetheless, impact the frequency response. Older amps that use point-to-point wiring are more susceptible to this whereas PC-board amps are more immune, although not totally.

Deviation can also occur due to amplifier-to-loudspeaker interaction differing from the predicted interaction. The output impedance of a tube amplifier is somewhat high and this causes the frequency response of the amplifier to follow the impedance of the speaker. Most amplifiers impart some amount of negative feedback to reduce this but, nonetheless, some degree of interaction is always present. The Axe-Fx predicts this interaction very accurately but there are still cases where there is deviation due to this interaction. Another possibility is component tolerance and drift. While the design may specify certain values for components, in practice the actual components will deviate from these values. Over time the deviation can increase due to component aging. In some cases these deviations improve the sound of the amp. In other cases, however, this can be detrimental.

2.1 FIXING IT

MIMIC addresses frequency response deviation using a patent-pending measurement technique that identifies and corrects these deviations. A complex series of tones is injected into the amplifier being simulated and the output is measured. These measurements then form data sets that are stored with each model and correct the various filters. These data sets are fairly large but, fortunately, the Axe-Fx II was designed to be future-proof and has copious non-volatile memory. You will notice that the size of Version 10.xx firmware is quite a bit larger than previous versions. This is due to the inclusion of this correction data.

MIMIC also detects when there are significant deviations between the measured and predicted responses and issues a warning. This warning is an indication to reevaluate the model and identify possible data entry errors.

Note that MIMIC is more than just EQ matching. It matches all the various filters in the simulation to their real-world counterparts by carefully applying test signals of various amplitudes to isolate the desired filters.

3. NONLINEAR RESPONSE DEVIATION

While frequency response deviations are the most noticeable type of deviation, differences in nonlinear response are also detectable by humans. Tube amplifiers impart two distinct types of nonlinearity: clipping and crossover distortion. Clipping distortion is the primary source of distortion in an overdriven tube amp and is intentionally created by overdriving the preamp and/or power amp. Crossover distortion occurs in push-pull power amps and imparts distortion on low-level signals. While large amounts of crossover distortion can be objectionable, small amounts give an amp character and are especially useful in aggressive, modern tones. Some modern simulators do not include crossover distortion simulation and we feel this is a glaring omission.

Deviation from these nonlinearities arises due to the transfer function of the simulated distortion being different from the transfer function of the distortion generated in the tube amp. All amp simulators rely on waveshapers to generate distortion. One common method is the “ $(k + x) / (k - x)$ ” technique. This method implements a waveshaper that is somewhat programmable in terms of how “hard” the clipping is. Other techniques involve polynomials or other mathematical functions. The Axe-Fx II is unique in that it implements actual triode models for the preamp simulation. These models behave nearly identically to an actual tube triode (and are far more accurate than the aforementioned techniques), complete with frequency and amplitude dependence.

Another primary source of nonlinearity deviation is bias point deviation. The ratio of odd-to-even harmonics generated by a nonlinearity is a function of the quiescent operating point. In other words, how much the signal is shifted to one side of the transfer function. This is known as the bias point. While we can predict with excellent certainty the bias point of a tube, the real world doesn’t always agree and human error can also be a factor.

3.1 FIXING IT

MIMIC addresses deviations in nonlinear response by applying pseudorandom noise of varying amplitudes to the amplifier being simulated and comparing the response to the response of the simulated amplifier. Histograms of the responses are collected and analyzed using statistical methods and any deviations in distortion transfer function and bias point are then identified and corrected.

4. DYNAMIC RESPONSE DEVIATION

The “feel” of an amplifier is mainly attributed to its dynamic response. Tube amplifiers compress when overdriven due to sagging of the power supply. The amount of sag depends on a variety of factors including the power transformer, type of rectifier used, etc.

Deviations in dynamic response can make an amplifier simulation feel too stiff or too spongy in comparison to the real amp.

4.1 FIXING IT

MIMIC addresses dynamic response deviations by applying tonal pulses to the amplifier being simulated and to the simulation simultaneously. The pertinent parameters of the simulation are then varied until the dynamic characteristics of the simulation match those of the amp being simulated. These corrected parameters are then stored with the model.

Summary

MIMIC™ processing is a major advancement in the field of guitar tube amplifier simulation. This technology identifies deviations in the response of the simulated amplifier to the actual amplifier and generates corrective data bringing a level of accuracy that has been heretofore unachievable. An inherent advantage of this technique is the exact separability of the amplifier simulation from the cabinet simulation. This allows those that desire “amp-in-the-room sound” to achieve that with no compromise as compared to competing technologies that cannot fully separate the amplifier simulation from the cabinet response. Simply bypass the cabinet simulation and connect the Axe-Fx II to a quality power amp and guitar cabinet of your choice. For those comfortable with Full-Range Flat-Response monitoring (FRFR), MIMIC achieves polished, studio-quality tones with ease.

A tremendous amount of work went into this technology and we are pleased to offer this upgrade free-of-charge to all owners of the Axe-Fx II Digital Guitar Processor.