Rockman: the Concept

Play guitar at a professional level without tubes and without amps, and make it available to anyone

www.rockman.fr

Foreword

❑ I have truly hesitated before publishing this memo. After all, the only person who could write about the concept he had in mind is Tom Scholz himself, and I'm not entitled to say "that's the way the Rockman concept was born".

Moreover, it is a very long essay about music, sound, and guitar gear: not the kind of thing that someone who needs straight information about his Rockmodule or his Rockman headphone amp will read easily.

❑ Yet, I've been studying in detail the Rockman product line for over two years now. The first time I opened a Rockmodule, my reaction was something like "hey, that's how I would have designed gear myself!". Well, some people like me just "would", while others "do"…

Two years later, I managed to imagine what the rationale behind the Rockman concept was. I based my analysis on the following assumptions:

- Tom Scholz was an engineer before being a professional musician. Let's think as an engineer addressing musicians' problems, and we'll get a chance to be close to the truth.
- Tom Scholz is someone who gives more value to control than to randomness. In other terms, each detail in the Rockman line is here on purpose, never for the sake of it.
- Tom Scholz is the kind of man who considers that every problem has a solution and that when you can't find a solution, that's basically because you're not addressing the right problem. If something is missing in the Rockman products, then the feature is certainly not truly interesting.

❑ The various considerations about music, sound and gear design you will read here are thus personal opinions, based on 30 years of experience in music and electronics, and oriented towards the understanding of the Rockman line.

Reading them will be helpful to anyone who wants to make his own opinion, even if he doesn't agree with what he reads, and even if he doesn't use Rockman gear... yet!

❑ All in all, this memo makes sense, and is probably close to the "truth". As a disclaimer, I have to say that:

- The points of view expressed here are personal opinions – other people will certainly have other approaches.
- Though I think that Tom Scholz has followed most of the principles described here, there are other ways to arrive to the same gear concept.
- If you do not agree with what is written here, and have serious reasons to say I was wrong, please contact me!

Jark, November 2007

About Music

❑ Music is “the art of combining sounds”. Great! What is an art? What is a sound? An art is not The Art: an art is a domain where people work to improve the quality of their creation, and/or the way they create their achievements. Yes, music is a domain of creation, and, for the lucky ones, achievement.

The Art is something different, which has several definitions: a quest for beauty, a free-expression area, a way to generate emotions… This debate would take us away from what we are here for: Rockman gear.

So let's assume that creating and playing music is a structured activity, that has some rules and some objectives – even if these objectives are not explicit, and if the rules are, very often, made to be broken.

❑ Sound… when this classic definition (combining sounds) was proposed, a sound was clearly something consistent: a sound has a timbre (i.e. a frequency spectrum), a dynamic structure (we call that an envelope now), and a duration. In other terms, a sound was what we would call today... a note!

In these ancient times, the rest was simply “noise”: every aural stimulus that has no complete structure.

Of course, there is no law prohibiting the use of “noises” within music! But if you don’t have a structure based on notes, can we still call that music? Do we call photography or computer graphics “painting”? No, we don’t: we have other names for these alternate artistic domains.

Well, it would have been fair to create new names for these domains where people create sonic opuses without instruments and without notes… For example, when you listen to "Let it be", then to “Revolution 9”, a new term, something like an “aural collage”, is certainly be more relevant.

❑ There are several definitions of music, and musical gear can accordingly be designed with hundreds of different methods. It’s always created to help musicians in their daily concern: combine sounds, no matter what you call music or sound.

Rockman gear was created in conformance with the classic definition of music, as it is exposed above. Rockman gear is not here to generate random tones or explore technical possibilities. It was created to provide a better control of their tone to the guitar players. Nothing else. Tom Scholz has one device that can generate weird noises and sounds: the Hyperspace Pedal. But it’s the only effect that was not included in the Rockman line…

About Rock Music

"Let there be sound", and there was sound
"Let there be light", and there was light
"Let there be drums", and there was drums
"Let there be guitars", and there was guitars
"Let there be rock!"

Yes, rock is a show. A show based on guitars and drums.

And rock requires Sound. Sound with a capital “S”…

❑ You can write the best songs, play guitar like a living god: if your band has no “Sound”, no one will be interested. On the other hand, it’s perfectly possible to produce a record with a Sound and nothing else: just turn on your radio if you need some examples…

Getting a good sound is easy when there is one instrument only: turn a few knobs with the help of you ear and you’re done. If you are a top rank player, the way you play will anyway be more interesting than the sound itself: you’re a lucky guy…

But rock music is played by bands, not by people sat alone on a tool. Mixing the various instruments is a complex and tricky operation, which goes much beyond adjusting volume pots.

❑ Let’s take a classic rock band: bass, drums, keyboards, guitar and lead vocal, playing live in a club. If these people lack experience, the result is usually the following:

- The drummer strikes louder and louder, cause he’s a drummer.
- The guitarist turns his volume up, cause he’s a guitarist and is very proud of his own sound.
- Everyone forgets the bassist, cause his round sound will be drowned in the global mess. Bassists being nice people, they don’t turn their volume up, and the bass part is reduced to a global hum in the background.
- The keyboards are too loud if played in the high frequencies, and too soft in the lower range. Keyboard players being full of courtesy, they never complain about it.
- The singer shouts as loud as he/she can, but never loud enough to be heard correctly. And if the singer was singing loud enough, the guitarist would play louder...

All in all, the audience gets a muddy and messy mixture of kick drum, screaming guitar and striking snare drum. A few people know there’s a bass playing, cause they have seen a big guitar with four strings on the stage. And if the singer is good looking, they will remember the gig, especially if the singer is a cute girl.

❑ Yet, the sound of a record is different. You will say: a record is made in a studio, with expensive gear and a complete team around the musicians. Well, it's hopefully perfectly possible to get a correct sound on stage, but it's more complex cause it must be done in real time.

A friend of mine used to gig a lot in the eighties, in a local band. He's a professional player, who has owned tenths of amps and guitars, and has played in many different contexts. In this local band, the drummer had an e-drums kit. The bassist had a Rockman Bass, my friend played with a Rockman X100, and the other guitarist had a Rockman Soloist. Nothing expensive, nothing impressive. But the result was "people always told us: you sound just like a record!".

❑ Sound is not a matter of gear, and is not a matter of cost. Sound is a matter of know-how. What you get when you buy a Rockman is a tiny bit of the know-how that was developed for Boston. And this know-how works both on stage and in a studio: you may want another approach of sound, but this know-how was made available to everyone, while the other bands and sound engineers usually keep their secrets for them…

About Sound

Sound is of course a question of taste, but it has also some basic rules.

❑ Mixing excellent individual instrument sounds will probably give bad results, just like mixing several fruit-juices will probably give just a thick, sweet drink without any precise taste: the individual qualities of each fruit-juice will be lost in the mix.

Take a lettuce, oil, vinegar, salt and pepper. None of them tastes really good one by one: would you drink a glass of vinegar, or eat salt with a spoon?

Mix them carefully, and keep the balance between each of their characteristics: you will have a tasty salad, and will enjoy eating it.

Fruit-juices are very similar in terms of taste: they are all sweet. Mixing them will just kill their individual qualities, cause the individual tastes "overlap".

Lettuce, salt, pepper, oil and vinegar are all different.

Their tastes will not overlap, and if their individual qualities are not obvious, mixing them will give something complete and balanced: acid and sweet, soft and fresh.

❑ Conclusion:

- A correct band sound is not made of excellent sounds stuffed into a mix.
- The individual sounds of a correct mix usually sound weird, taken one by one.
- A correct mix is made of individual sounds designed to sound good within a mix, not to sound good individually.
- Mixing individual sounds that sound good, one by one, requires severe cut-offs before getting something correct in the end.

Rockman gear was designed with one thing in mind: since the final objective is to make records, or to get a correct global band sound on stage, let's directly design guitar gear that will sound good in a mix. Focusing on a good individual sound is not the best way to design gear.

About mixing

❑ Just like cooking a meal requires a precise knowledge of each ingredient taste and flavor, mixing instruments sounds requires a precise knowledge of their "taste" and "flavor". Taste and flavor, in the sound domain, are timbre and dynamics.

A musical sound (a note), has a fundamental frequency, and some harmonics. A musical instrument thus generates frequencies in a given interval that go much beyond the fundamental frequencies, and these additional frequencies give some personality to each instrument.

❑ The problem with a band is that some instruments generate frequencies over the same ranges. If the harmonics of the bass are in the range of the singer's fundamentals, and if the guitar plays in the frequency range of the singer too, the singer's voice is drowned in the usual bad live gig mess… That's the frequencies overlapping.

But frequencies without dynamics are nothing. The way all these frequencies raise and decay is at least as important as the frequencies distribution. In other terms, if you find a way to cut some frequencies in a guitar sound, you will still recognize it's a guitar. But if you find a way to alter the dynamics, let's say with a very slow attack, it will sound more like a violin than a guitar.

This fact is the main key to a good band sound:

- You are allowed to cut some frequencies, cause the human ear can recognize an instrument even if some frequencies are cut-off.

- You must do your best to respect the dynamics of each instrument. Each instrument will be clearly heard if its dynamics can express freely.

❑ Frequency overlapping kills the individual dynamics: the singer's voice, lost in a frequency range where several instruments overlap, loses its dynamics cause this range is too busy, and becomes difficult to hear.

About guitarists

❑ Drawing a caricature of the basic guitar player is easy…

- Gear is everything, playing is natural
- In a band, the key member is the guitarist
- In a song, the key part is the guitar solo
- The best way to prove it is to turn the volume up
- The objective of life is to find the perfect sound
- Getting the right sound implies having more gear
- Turning the volume up is necessary to show how good your sound is
- A big cab is necessary to prove that you can play louder
- Etc… etc…

Great: it's nice to confirm that guitar players have a huge ego, and that their own personal sound is more important than anything else.

But the aim of the game is to have a good band sound. Not to duplicate the NAMM show on stage!

❑ The addiction of guitar players for gear is a complex phenomenon, and I'm no psychiatrist… But there are a few facts that anyone can see:

- Tube amps are still alive, more than ever, though everyone knows they have serious drawbacks
- People are still fond of cheap stompboxes, though everyone knows they are a technical heresy
- Complex rack systems and sound processors can get good, and sometimes impressive, individual sounds, but it's much easier to sort out the sounds that can be used than to get the right sound out of them
- The sterile debates about tube vs solid-state, digital vs analog are always on, though no one can tell which is which when he hears it on a record…

❑ Well, it would be a little naive to claim that Rockman is the perfect answer to all these issues, but all in all, Rockman is a solution that has one quality: it addresses these four items, and provides a reliable, efficient and professional answer to the four of them…

About tubes

❑ I have written, two years ago, a short French website, called "Les lampes pour les nuls" – Tubes for dummies.

I was just getting into Rockman gear, which was then only a great addition to my favorite tube amp – an old Boogie.

The first purpose of this website was to explain in words that every one can understand what a guitar tube amp is. The more I was writing, the more I was realizing that tubes had, from far, more drawbacks than advantages. When I finished writing these few pages, the conclusion was that I wasn't fond of tube amps: I was fond of my Boogie, which is totally different!

❑ The difference between solid-state and tubes doesn't rely in technique. The only real difference is that, in 2007, it's possible to build a correct solid-state for a very low cost, while it's impossible to build something correct with tubes below a certain price. The matching marketing strategies are thus different.

Full point.

The rest is strictly a topic for intellectual and sterile discussions.

❑ If you like intellectual and sterile discussions, here are a few starters. You are authorized to use them whenever you're looking for a silly discussion subject…

- Tubes amps are louder! No. A Watt is a Watt, but a tube amp can be used beyond its RMS zone without collapsing. Beyond the RMS zone is the power-amp saturation zone, which is difficult, not to say impossible, to use with solid-state with a musical result. A tube amp can therefore generate more electric power than its rated RMS power – with a 1.5 to 1.7 typical ratio, and a theoretical limit of 2.0. That's all...
- Tube amps sound better! No. Tube amps are usually expensive devices, resulting of years of careful R&D, and are therefore rather high-end amps. A real cheap amp is always a solid-state device. High-end gear sounds "better" than low-end gear: what a surprise! Moreover, high-end amps are equipped with speakers having a better sensitivity: a 25W amp with an expensive 103dB/W/m speaker sounds as loud as a 100W amp with a cheap 97dB/W/m speaker…

- The big stars play with tube amps! Yes, but no. Until the seventies, the only professional amps were tube amps – solid-state was too young to allow serious developments. So, of course, people like Hendrix, Page or Clapton have made their reputation with tube amps – the only amps they could find. The cultural habit is still here, but a lot of famous records were made with solid-state gear – everyone believing they were made with tubes. This trend has increased when digital gear has appeared.
- We have a different feeling with a tube amp. Yes. That's true, provided you don't play ultra-clean or ultra-saturated. Tubes have a very specific way to shift gently from clean to distortion, and this dynamics provides a special feeling to the guitar player playing in the crunchy range. This feeling is, unfortunately, something that the audience will not share: it has no impact on the sound!

❑ All in all, this tubes-transistors technical opposition is over since at least 25 years now. From a commercial point of view, the discussion is still open, but that's another story…

It doesn't mean that tubes became obsolete, nor that solid-state sounds better than tubes, or that tubes will always sound better. It just means that it became possible to get a professional sound from solid state gear, and that people who do not appreciate the capricious behavior of tube amps had an alternate solution. As a matter of fact, solid-state has a quality that tubes don't have: the sound never changes from take to take.

❑ The first Rockman, created in 1982, was the proof that a solid-state device could easily replace a 100W tubes stack with two 4x12' cabs. Replace doesn't mean copy: the Rockman has a similar but, of course, different sound. But no one said that the objective of a musician was to copy what was made before…

Do we spend hours trying to duplicate the sound of Robert Johnson playing "Walking blues" in 1936? No. So...

Walk on!

About pedals

❑ The pros and cons of stompboxes and rack units are purely technical and financial. Music has nothing to do here. As a matter of fact, any circuit can be adapted from one to another, even if it's not always that obvious.

The stompboxes are a "bad habit" inherited from the past. In the sixties, the first guitar-oriented effects were very small circuits that didn't need a large enclosure: fuzz-boxes and wah's. The other effects (reverbs and tape echoes) were too big to be mounted in a stompbox format.

Then came the phaser (an electronic imitation of the huge Leslie cabins). The Univibe, certainly the first commercial phaser, was built in a floor-based format: it was, after all, logical to place it side by side with, let's say, a Vox wah and a Fuzz Face…

❑ As a matter of fact, the question of rack vs stompbox didn't come a long time ago: the amps had no "effects loop", so an effect could be placed either between the guitar and the amp, either after the amp, i.e. at the mixer level, after miking the amp. The pre-amp, guitarist-oriented effects were therefore all developed in a stompbox format, while the post-amps effects – flangers, echoes, etc… were more sound-engineer tools than musician devices.

Things have changed in the eighties, when people got better information about their gear, and moreover, when stompboxes became common and available. A flanger or a chorus pedal placed before a saturated amp sounds horrible, for example, and people have learnt this kind of things.

So the principle of an effect loop was created. This loop allows connecting effects after the saturation of… the preamp. Fine when you play a solid-state amp, cause the power amp is linear and is not used for saturation, but it doesn't really help if you play an old-school tube combo and use the saturation of its power amp!

Anyway, that's when the question of the stompbox format arrived: if you want to connect your favorite flanger in the loop of your amp, you need one cord from the amp send to your pedal, then another from the pedal to the amp return. Feet and feet of tone-killling cables…

❑ OK. Let's stuff all the electronics in rack units, once for all, and let's include a remote footswitching for, at least, the bypass function. That's logical, and it makes sense from a technical point of view. There's one drawback: cost.

The reason why stompboxes are still alive is that they can be designed – and sold – at an attractive price. It doesn't mean a stompbox will sound bad: it is of course perfectly possible to build big pedals with high-quality components and a built-in power supply instead of batteries.

❑ Oh yes, batteries… Why are stompboxes used with batteries? Manufacturing cost, one more time! The constraint for a manufacturer today is to create circuits that can be operated with a 9V battery… with all the associated drawbacks.

❑ The following technical details are minor points compared to the long cords issue, but the consequences of battery operation become critical in specific conditions:

- Less available voltage (9V in a stompbox, up to 24V or more in rack units) means more noise, the signal-to-noise ratio being lower.

- Less available voltage means less headroom: most of the stompbox units cannot handle the line-level signals of an effects loop.
- One voltage only, instead of a balanced two voltages power-supply in a rack unit, has consequences on the design of the effect. The biasing technique can, in some specific cases, make the circuit really weird and cumbersome, with degraded performances.

❑ Rockman gear was clearly high-end equipment, so the cost was not an objective. That's how the decision of making only rack units was made. SR&D actually made two stompboxes before ceasing its activities - the UDG and the AGP – but these two low-cost items were market oriented, and were not designed in the spirit of the original Rockman gear.

About digital processors

If you are reading this page, you certainly know that Rockman gear is purely analog, and that Tom Scholz doesn't like digital processors.

❑ By coincidence, my initial specialty as an engineer was… digital signal processing - DSP. I will always remember this teacher warning us, when we had to choose, for our last year in our engineer school, between the "analog electronics" and "digital processing" classes: "You will learn everything with analog electronics. Digital may seem fancy 'cause it's new, but you won't learn as much." That was in… 1984.

Our teacher was right. Digital processing is computer science, while we were in fact interested in signal processing. Programming lines of code versus creating circuits and real devices… Hopefully, my father had taught me the basis of electronics, and this basis is still useful today. I graduated in 1985, and what I have learnt about computers 22 years ago is completely outdated...

❑ Anyway, while DSP was still a young domain in the eighties, it is now mature, and we can now find convincing music-oriented digital processors. Like the tubes vs solid-state debate, the question "which one is better" will soon be over. But it is not over yet, strange as it may be. The "perfection" of digital processing has revealed several severe drawbacks, amongst many qualities. These drawbacks can be cured by increasing the programs and algorithm sizes, and the technical capacity of the processors: more bits, more bytes, more RAM, etc…

❑ But digital processing is, unlike analog processing, based on something that will always keep it apart, and therefore make a difference: quantization.

As long as digital processors are asked to act in a linear manner, they can do an excellent job if the gear is not too cheap, especially with impulse response and convolution approaches. For some non linear processes, it is possible to get excellent results, provided that the problem can be deciphered into algorithms or formulas. It's only a matter of money and programming labour.

But that's where the problem is: musical gear is based on hundreds of non-linear behaviors, all these defaults of the components that make analog gear sound "natural" (one can wonder what can be natural in a chain of guitar amps and effects, but why not…).

You can build a mathematical model for each non-linearity: clipping, compression, etc… But you cannot create a mathematical model of a device which has a different behavior each time the signal level changes, especially if these behaviors evolve continuously. You can of course imagine an algorithm with 4, 6, 8 different behaviors depending on the signal strength. You can dream of a processor capable of 1024, 2048 or more different behaviors. But you will always be limited to a certain number of cases, because the numbers handled by the computer are quantized and not continuous. It's not a problem if you can find a mathematical law linking these 1024 behaviors – a computer can do that. But in the real life, there is no law to describe the individual changes of all the components of a complex analog circuit.

Is it really critical? Well, as long as digital gear will only aim at copying analog stuff, it will certainly be critical. The real problem, and the only question, is "why can't we quit copying, and why don't we create new guitar sounds based on digital gear?".

❑ I don't have the answer. All I can say is that digital signal processing has brought two things only to the guitar players: pitch-shifting (harmonizers and whammy pedals) and reverbs. You can read all the webpages you want, the rest is purely analog gear duplication...

If you consider that pitch-shifting existed much before the digital era, by the means of tape-speed control, and that reverb is probably the only effect that doesn't need electronics (it will give you a good reason to enter a church if you don't see what I mean…), what did digital processing bring us? Nothing, or nothing really new.

Tom Scholz has a sentence that summarizes perfectly the digital processing subject: there's nothing digital stuff can do that analog gear cannot do, and analog is anyway easier to control and design. And that was exactly what my teacher tried to tell us when I was a student!

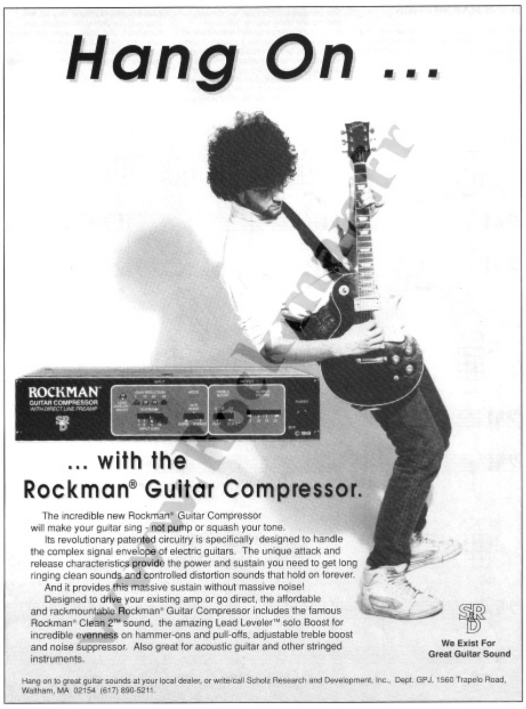
❑ OK. Back in the eighties, when SR&D was creating and building Rockman gear. Remember that Tom Scholz is not totally opposed to digital technologies: Tom has created samples-based drum machines before they were actually available as commercial items, and if you read carefully the Boston records sleeves, you will find that Lexicon reverbs are used for the vocal parts…

But after reading what I write in 2007, why would have he wasted time in the eighties with digital gear for the guitar players?

About compression

❑ We have said above that dynamics is a critical characteristic of a sound that allows the human ear to recognize an instrument. Dynamics allows hearing an instrument within a mix, or a voice amongst instruments.

The other characteristic is the timbre: the proportions of harmonics added to the fundamental frequency of a note. But altering the timbre (spectrum) of sound is possible, and the human ear will still recognize it's a guitar, a piano or a cymbal.



What about altering the dynamics of a sound? If one suppresses the envelope of a sound, or changes it in a manner it cannot be recognized, it will, of course, be impossible to tell what the original instrument is. The best example is certainly the "violin effect" that a guitarist can achieve by raising his volume pot from zero at the beginning of every note (listen to "Survival" by Yes for an excellent example).

Respecting the dynamics of a sound is therefore critical. But respecting this dynamics doesn't mean it's impossible to control it. Most of the modern instruments – drums, electric guitar, electric bass – have huge dynamics: the amplitude variations can go much beyond what the human ear likes to receive.

❑ Dynamics control is all about time constants:

- Tweaking the volume over the duration of a song is just a subtle effect, used to add life to the music
- Change the volume of an instrument within a few bars will be heard as a technical problem
- Change the strength of notes within a bar will be heard as a instrumental problem
- Alter the volume within the duration of a note will radically change the sound of the instrument

❑ There are three basic dynamics processors:

- Compressors
- Expanders
- Gates

Since the major issue with modern music is to tame the huge dynamics of instruments, expanders are rarely used. Gates are very specific tools that can be used to modify the sound of a drum or a cymbal, for example, and are not used with guitars.

Joe Meek is certainly the man who has explored and settled the foundations of this process, and his circuits are still in use today. Compression is widely used in modern music, for the best and the worst.

❑ Compression for the worst. A compressor can kill the life of an instrument track, cause playing soft or strong is a critical means of expression for a musician. It can also kill the musical message of a track, or of an album. The radios that broadcast music must compress the music, for technical constraints. The DJ's have to compress too, to get this high volumes that people want in a disco. But when you really want to listen to music the way it was meant to be, you don't want any compression.

❑ Compression for the best. Compare drum tracks recorded in the sixties or the early seventies with what we do today. First of all, the miking techniques are much more sophisticated than before, and almost each instrument of a drum kit is now recorded separately. This allows applying compression, but also gating and expansion on a case by case basis (and reverb, echo, EQ…). Drum tracks have gained expression, musicality, and most of all, clarity.

Of course, top-rank drummers don't really need that: people like Manu Katche, Steve Gadd, or basically, every experienced jazz drummer have a sufficient control over their hands and feet to sound good even with two mikes only in front of them. But in the real life, where drummers are human beings and not aliens, dynamics processing and compressors are almost mandatory when a clean and strong drum track is desired.

❑ Compression for guitars. A compressor can work as a limiter (put the peaks under control), as a sustainer (increase the notes decay time), or both. With a short response time, a compressor allows leveling all the notes in a guitar part: this is extremely interesting for arpeggios, for example, or during a very fast lead part, when the right hand can "miss" the pick of a note.

❑ Tom Scholz knew all that, of course, when he started thinking about building new gear for Boston. He has registered a patent which is, in my opinion, one of his most interesting publications: US 4,627,094

Plug a guitar directly in the soundcard of your PC, and record what gets out of it. You won't like it… the sound will be thin, even if you boost it. Boost it again, and it will clip and sound even more horrible.

Take a compressor, and do it again: you will lose in expressivity if the compression is too strong, of course. But all in all, it will sound much better. More "natural". The sound is almost acceptable. Add some EQ, some reverb, and you're done: you have a real guitar sound. No amp, no saturation, no speaker, no mike: just a little box with a few components…

❑ The explanation is fairly simple, and is explicit in Tom Scholz's patent:

> "On a guitar, the first sound or pulse that comes out can be a huge peak which is almost always much stronger than the signal which follows within a few milliseconds. A guitar amplifier tends to smooth out these sounds because it cannot respond to them fast enough, because it clips (distorts) large signals, and because the speakers have a slow response"
>
> Scholz – US 4,4627,094 – col.7 – l.25 to 30

This paragraph is probably the most important for someone who wants to understand the Rockman line. Tom Scholz has created the first realistic solid-state amp simulation, because all the Rockman preamps, from the first Rockman headphone amp to the last SR&D product – the Ultimatum Distortion Generator, start with a built-in compressor.

This compressor acts both as a limiter and a sustainer, but the limiter aspect is what gives this natural feeling and this loudness: since the peaks are smoothed out, the rest of the notes can be boosted without clipping. The sustainer function is only additional comfort if you have weak pick-ups, but it doesn't have the same criticity: getting pick-ups with a high output level will provide better results.

❑ The rest – distortion, filters, BBD's – was pretty straightforward in comparison: that's what all the other manufacturers actually do, with different approaches and results. SR&D has "only" developed high-quality devices. But the compressor inside every Rockman preamp is the key of SR&D's specificity.

Compression was the basis of the Rockman DI approach, allowing playing without amps nor cabs.

About saturation

❑ Saturation is a key effect in rock music. People started using it in the early sixties, by means that had very little chance to give an interesting musical result: push the amps beyond their nominal limits. Well, it worked, so let's forget history, and focus on facts.

Saturation is only one of the many distortions that can occur in an audio set-up. As a matter of fact, distortion applies to any process which is not strictly faithful to the original signal. With this audiophile definition, every instrumental piece of gear is a source of distortion.

❑ Saturation refers to pushing a device beyond its normal limits: a speaker can be saturated (mechanical saturation - it doesn't last long), the output transformer of a tube amp can saturate (that's magnetic saturation), and of course, all the active components of an electronic circuit can saturate (the passive components can burn out, but do not saturate).

Saturating active components causes signal clipping. These components have, by definition, a power supply that can provide a limited amount of energy. Ask more energy, and it won't respond: the signal is clipped. That's what happens in an overdriven tube amp.

An overdriven solid-state circuit (transistors, FET's, OpAmps, etc…) can give similar results, and can be used to obtain the same sound as a tube amp – provided you can control the result with the proper filters.

Clipping can also be obtained without active saturation: you can use diods to achieve clipping, because their natural behavior is to clip one half of the signal. Diods are actually in-between components based on the same technology as transistors, but they are used as passive components.

❑ That's a brief recall about saturation and its objective: clipping. We want to clip the signal to obtain a sound that has some musical qualities.

Oh, I was forgetting something critical about saturation and clipping: they don't only alter the dynamics of the original signal. They just suppress it! In other terms, a saturated signal has the same dynamics than a hyper-compressed signal – though the principle is totally different.

So what? Losing the dynamics makes a guitar sound absolutely different, as if it was another instrument. And that's what makes it attractive for rock music: this new instrument – saturated guitar – has its own identity, its own culture, and even allowed creating new styles of music: hard rock, metal rock, heavy metal, arena rock, wouldn't exist without saturation, or should we say, clipping.

❑ So, what does clipping, exactly? Clipping does two things:

- Saturation and clipping suppress the dynamics of the signal: the amplitude variations (the envelope) are erased, gently first, then strongly if you increase the input signal level (that's the "gain" pot).
- Clipping adds artificial harmonics. The more you saturate, the more harmonics you get.

Great, but we already had natural harmonics, hadn't we? Yes, we had, and they are still here. So a saturated guitar signal is made both of natural harmonics and artificial harmonics… Merging the two families (artificial and natural) of harmonics (they all have the same frequencies) will cause complex inter-modulation effects. Some interactions are pleasant to the ear, and some are not.

Moreover, the possible number of combinations is simply huge – and this explains why there are so many types of saturations, distortions, fuzzboxes, overdrives, etc…

❑ All these effects are based on the same structure. All. They just differ by the filters placed before the clipping stage, that control the input signal harmonics distribution, and the filters placed after the clipping stage, that control the result – natural and artificial harmonics. You can of course, make things more complex by placing several clipping stages in the circuit, but the final result will not really change.

❑ As for the clipping technique, each manufacturer has his favorite approach: FET's, OpAmps, regular diods, tubes, bipolar transistors, MOS-FET, etc…

Tom Scholz has selected the LED clipping technique, probably after a long series of tests and prototypes, and, why not, because LED's light up when they clip!

"I don't use tubes or transistors as overdrive elements; I use LED's […] It doesn't matter if it's tubes, transistors or LED's, if you do the right thing with them. On Third Stage, for instance, I started off using tube amps for several of the songs. I went back later and changed some of the parts so that some songs had pieces of tubes and Rockman. Nobody can tell which is which: I can only tell you because I was here (Guitar World, Feb. 90)."

The clipping element is a basis, the component that will create these artificial harmonics. As a standalone circuit, a clipping diod, transistor or tube sounds horrible. The art of distortion doesn't stand in the choice of this clipping basis: it stands in what you connect before and after it to get something musical.

And what you place before and after are filters.

About filters

❑ An EQ being only a series of adjustable filters, it seems logical to address filters and EQ's as one topic, though their usage is of course different. The message is that they have exactly the same technical role, and are actually based on the same techniques and circuits.

❑ Filters belong to the family of linear processes. It may seem contradictory with the fact their frequency response is all but linear: let's recall what a linear process is.

- When you add two input signals, the output signal of a linear process is the sum of the two processed signals.
- When you multiply the input signal by a given coefficient, the output signal is multiplied by the same coefficient.

The following effects are therefore linear: EQ's, filters, boosters, but also tremolo, chorus, flanger, phaser, echo, etc…

The following effects are non-linear: distortion, overdrive, fuzz, compressors, expanders, gates, etc…

Let's note that the wah is nothing but a filter, with a specific response and a foot-control device. In other terms, a fixed wah can be simulated by any good EQ…

❑ For a sound engineer, EQ's and filters are used:

- to avoid the frequency overlaps within a mix
- to correct a possible weakness in an individual sound by boosting slightly a frequency range
- to control the global balance of all the frequencies at the band's sound level (this can include the venue frequency response during a concert)

❑ For most of the musicians, EQ's and filters are used for one thing only: modify the harmonic distribution of the sound, in the hope of getting a better sonic result. This is the classical use of a post-distortion EQ.

❑ For a guitarist who uses distortion, EQ's and filters have a critical role that is, unfortunately, ignored by most of us: an EQ placed before the clipping stage will control the mix between natural and artificial harmonics that now compose his sound. And that's a complex operation…

- The input signal has a fundamental frequency its natural harmonics
- Each of these frequencies will generate artificial harmonics during clipping: in other terms, clipping generates harmonics for each natural harmonic
- The pre-clipping filters control the gain for each frequency of the input signal, and therefore have an impact on the post-clipping artificial harmonics
- The final result must be filtered to get something pleasant to hear…
- The filters have a strong impact on the noise generated by the circuit

All in all, the pre-distortion EQ acts like the gain pot of an amp, but can do it for each frequency band. The post-distortion EQ will act like a master volume, but here again, frequency by frequency.

❑ What manufacturers do is to determine pre-clipping and post-clipping filters that give the global voicing of their effect.

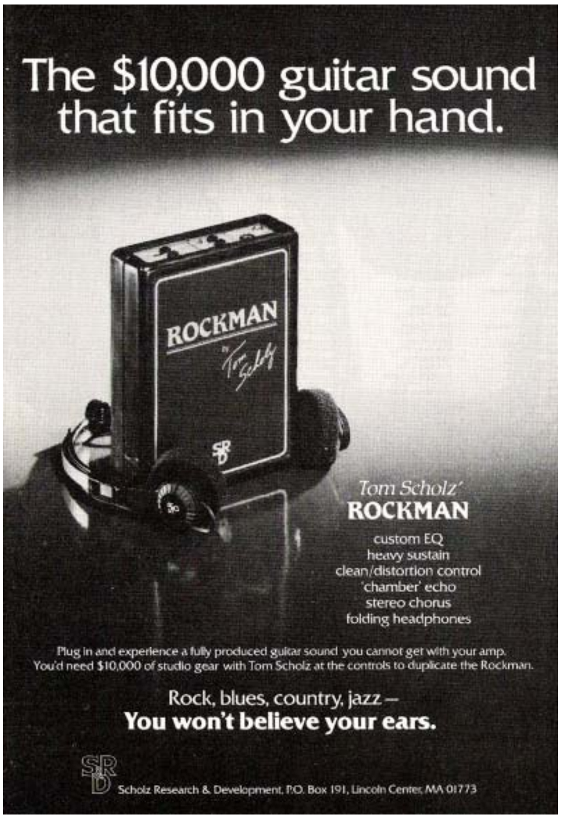
Since there are hundreds of possibilities, there are hundreds of possible distortion pedals and hundreds of possible amp's voicings. Not surprising to see so many commercial products that all do the same thing! They are all built the same way, and differ, basically, by the few components of the filters. Take a good distortion, place a first EQ before it and a second one after, and you can virtually get any sound you want...

❑ We didn't speak about the filters, as seen by the gear designers. Filters belong to the family of linear processes, but one can also say that filters can be used to simulate any linear process… And that's where they become extremely interesting for a gear designer!

❑ The sound of a guitar, as we hear it on a record, is the result of a complex chain of processes. What we call a correct sound can be achieved by connecting amps, cabs, mikes, and EQ's. That's the classical approach. But from a technical point of view, this devices chain is a series of linear and non-linear processes.

There are two basic non linear processes: compression and clipping. In the classical approach, these processes are performed by the amp-cab-mike chain.

There is one basic linear process: frequency response alteration. All the elements of a set-up alter the frequency response. In other terms, they behave exactly like filters…



❑ Fine. We have all the elements now. The mission is to create a correct guitar sound and to record it on a tape (the mission is not to create a sound that only the guitarist will hear!). This mission will be successful if we can:

- Compress gently the signal: this compression is rather electro-mechanical with an amp, but we can do it with simple electronic circuits.
- Control the final harmonics distribution (the timbre) by filters and/or EQ's. You can do that in the cumbersome, classical manner: electronics, a cab and a mike. But you can do it with electronics only, and call one of your filters "cab sim" if you think it's important for the marketing department.
- Add artificial harmonics on request via a clipping technique or another. The component itself is not critical, provided you know which filter is adapted: some pre-clipping filters will probably be necessary, and some post-clipping filters will certainly help.

❑ Yes, you can also do that with several pedals plugged into a big amp, several mikes around a huge cab and hours of sound engineering in the hope of getting something nice. It's up to you: you probably love spending thousands of dollars for every song you record.

But you can also do it with:

- A compressor – one OpAmp, one J-FET, a few caps and resistors
- A clipping stage – one OpAmp and two LED's can do that. Note that the LED's will light up when then clip (tubes don't!).
- Some filters. Complex filters, but only filters.

❑ That's called Rockman: the $10,000 sound you can hold in your hand...

About Delays

❑ As for now, we have addressed the basis tone of the guitars, as expected on a record. We have seen how compression, saturation and filters allow creating a guitar sound which is compatible with what our ear is used to get. But this sound must be located in the 3D space: that's where delays, echoes, reverbs and stereo chorus have a role to play.

❑ As a matter of fact, the guitar is only a part of the band, and all the instruments cannot be located at the same place:

- Point 1: Some instruments are far away, others are close to you.
- Point 2: Some instruments are on the left, some instruments are on the right.
- Point 3: We assume that all the instruments are placed on a horizontal plan...

This physical distribution was natural when music was always played live, with acoustic instruments. But rock music is played by electric instruments, with two main situations: on stage, or via a record. In both situations, the sound flows from two speakers, not from the individual location of each instrument.

❑ Let's address first the far/close control issue. The stereo question will be addressed in the next section.

Remember that the speed of sound is slow: around 300m/s. A drummer located 6 meters behind the other instruments will be heard 20ms after the rest of the band…Imagine a test configuration where the drum track would go through a 100ms delay (one shot, no repeat, only the delayed sound). It would be equivalent to a drummer located 30m behind the band. That's a beginning. Just a silly test, that allows finding the right solution. In the real life, a plain one shot delay is not sufficient to give the feeling of distance: it helps, but it's not enough. In fact, we would just think that the drummer is late on the beat…

Instead of a delay, let's add some reverb to the drum sound. A reverb doesn't repeat the sound: it adds a sort of decay after the original end of the sound. During this decay, we get a complex combination of what we've just heard. This combination is a complex sum of micro-delays that can have an average value of, let's say 30ms. 30ms correspond to a distance of 10m. Got it? A sound with reverb will be heard with a feeling of distance! Adding reverb to a sound is one of the tricks that can provide a distance feeling. Echo chambers (or delays now) can be used in a similar manner, usually with repeated echoes.

❑ Reverb and echo are very old effects now: they can be achieved via electro-mechanic means, and reverb is originally a natural effect that any large hall can provide. They belong to the “time-based” effects family, a term which covers now other effects such as flanging and chorus.

❑ With Boston, Tom Scholz has used almost every possible delay, echo and reverb technique. In the Rockman line, entirely based on solid-state analog technology, all these echo& reverb processors are based on Bucket-Brigade Delay (BBD) chips. These components, available since 1975, allowed developing fine delay units and acceptable reverbs, but also allowed creating the modern flangers and chorus effects.

About Chorus

❑ We have seen in the previous section how to render, with delays and reverbs, the distance of an instrument on a virtual stage.

The other space dimension is the left-right position: that's what stereo is made for.

❑ Let's review the basic rules of stereo:

- Rule 1: When two identical signals are sent to the speakers, these two audio sources combine perfectly and the sound comes from the center: we get mono from a stereo system.
- Rule 2: When two identical signals are sent to the speakers, with a phase inversion, the sound is cancelled at the center, and the sound seems to come from two separate sources. That's what we get when the connection of one speaker is inverted on a hi-fi set.
- Rule 3: When two identical signals are sent to the speakers, one of them being stronger, the sound comes from the strongest side: that's panning.

That's all about stereo, as long as we deal with plain signals: no echo, no delay, no reverb. We can control the sound along the horizontal, left/right axis. We can get a fixed position by playing with the balance (panning control), split the sound in two with a fixed phase inversion, or modulate the panning to obtain a rotating autopan effect.

❑ But we can go further, and use short delays to get more: create a wall of sound effect, as if the instrument was located not at a given point, but all along the left-right axis: that's the stereo chorus.

Ask two guys to climb on stage, with a few meters between them. Let them play exactly the same guitar part. It will sound somehow like a 12-strings guitar, and it will actually sound like two guitars, because they will not play strictly the same thing.

Instead of two guys, just keep one, and connect his guitar to two amps, a few meters between each other. It won't sound like two guitars: it will sound like one guitar, and you will hear it at the center: remember stereo, Rule 1.

❑ Let's set a 20ms delay between the two amps, which corresponds to a distance of 6m or so. We'll get a weird stereo effect, with a first feeling of space. But it's not sufficient to provide the feeling that two guitarists are playing along. It will mostly sound like someone playing between the walls of his bathroom, not what we call a wall of sound...

But if you 1) set a 20ms delay between the two amps, 2) modulate slowly this delay between, let's say 17ms and 23ms, it will sound almost like two guitar players! That's what a stereo chorus does. The sound doesn't come from the center of the stage: it comes from the complete stage width, as if two guitars were playing along.

❑ The principle of a chorus (doubler) is to apply a short delay, between 20 and 50ms, and to modulate slightly this delay, to avoid hearing two clearly distinct sounds. The listener has thus the feeling of having two instruments merging their sound, but one cannot say where the sound comes from. The final result is a wide stereo image, as if the instrument was not located at a given point, but all along the stage width.

The 20ms to 50ms choice is not an innocent choice. Each time period can be associated to a frequency, by the simple formula F = 1/T. The human perception ranges from, roughly, 20Hz to 20kHz, and the really audible sounds start around 40Hz.

The period of a 20Hz sound is 1/20=0.05s, i.e. 50ms. The period of a 50Hz sound (approximately the lowest note of a bass guitar) is 20ms. In other terms:

- A longer delay would make the chorus sound like a very short echo.
- A shorter delay would cause sound interferences that would alter the tone of the instrument (that's the comb filter effect used in the flangers).

❑ I will not address flanging, phasing and the comb-filter principle here. It would be very interesting, but there is actually no effect of that kind in the Rockman line.

❑ Stereo chorus, sometimes called doubling, has played a significant role in the history of SR&D and Rockman.

The Rockman Stereo Chorus, developed on the basis of Tom Scholz original doubler, is the main device used to convert a mono signal into a wide stereo sound, without altering the original instrument tone. The “long chorus” switch available in the Rockman chorus rack units changes the average delay from 20 to 50ms, thus allowing different sonic atmosphere and a deeper stereo effect.

About switching

❑ In 2007, every guitar player who reads a little about gear and effects is sincerely convinced that the best switching technique is what we call now true-bypass: a hardware switch that completely disconnects the effect from the audio path.

As a matter of fact, true-bypass does what it’s paid for: a bypassed effect doesn’t alter the sound at all. In other terms, the true-bypass quality can be appreciated only when you actually do not use the effect!

❑ Let’s come back to the origin of effects switching, and see why the true-bypass technique is far from being a perfect solution.

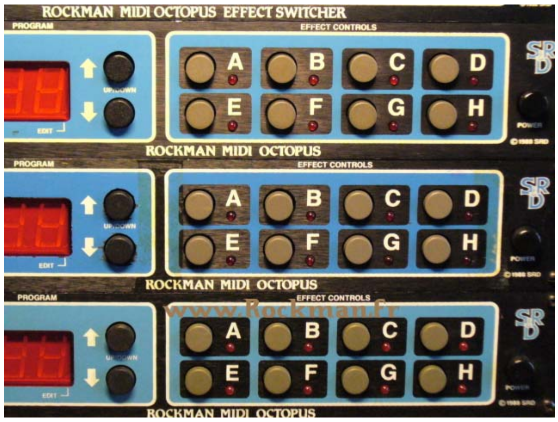
A switching system must provide several functions:

- Transmit the signal without loss nor alteration when the effect is off.
- Transmit the input signal to the effect and retransmit the processed output signal to the rest of the chain, without loss nor alteration.
- Allow switching the effect on and off without signal level difference and without unwanted noise.
- If the effect in not at the player's feet, in order to minimize the audio cords length, allow remote switching.

True-bypass is a good answer only to the function 1. If the effect has poor characteristics (e.g. low-cost buffers), it's the worst answer to functions 2 and 3.

As for function 4 (remote switching), true-bypass can be used, provided the designer has included a costly relay system in the device.

❑ All in all - and that's my own experience speaking, rather than a technical approach - true-bypass is truly bad, except in one very specific context:



When you want to merge an excellent tube amp with other devices, especially solid-state gear, the slightest addition to the all-tubes circuits will of course alter the sound.

Alter doesn't mean degrade: the sound is just different. If you really want to keep intact the pure sound of a stand-alone quality tube-amp, and use in the same rig other devices, the only solution is to install remote-controled relay true-bypass systems in your rig.

It's costly, and usually requires custom design if you take into account the fact that each set-up is unique.

❑ Tom Scholz has made the choice of rack-format devices: the only solution to have only short audio cords. He thus needed something reliable to remotely control his gear, and had suppressed the tube amps from his set-up. So what? No true-bypass. Active switching with professional-grade buffers were installed in every Rockman piece of gear, and that was the only way to provide the four functions of a quality switching system.

❑ The Midi Octopus was created to add Midi capabilities to the Rockman rigs: the Octopus controls the footswitches of each module and memorizes the configuration of each preset. For effects without remote footswitch, the Octopus can be completed with Remote Loops: this allows bypassing a stompbox, or basically any non midi device connected to the rig.

Synthesis

❑ The huge work accomplished by Tom Scholz is an answer to all the issues that are exposed in this essay. There are hopefully several ways to reach a sonic objective, and the Rockman approach is not the universal key to all the musical approaches. But the concept has proved its numerous qualities, and has reached the objectives that Tom Scholz had defined when he got involved in this adventure:

- Create gear that would provide the easiest and most reliable way to produce the records of Boston, and to get the same sound on stage,
- Make this gear flexible enough to allow using it in other musical and instrumental styles,
- And... satisfy his engineer needs to design and create innovative devices!

Had the corresponding gear already existed, SR&D wouldn't have been created. The initial spark was that Tom Scholz needed gear that wasn't available off-the-shelf, which is different from saying "I want this sound and I can't achieve it", or "I don't like the sound of the gear I have".

❑ The sound did exist before the Rockman line was created. With this sound in mind, Tom Scholz needed tools that would male obsolete the cumbersome collection of Marshall stacks, custom electronic boxes and mechanical inventions that had been Boston's daily environment for the two first albums.

Designing tools is an engineer approach, while creating music is a musician's job and controlling the sound of a band belongs to the sound engineers and to the producer. Tom Scholz plays the four roles, and that is the key factor that made the Rockman possible.

As a sound engineer and a producer, Tom introduced two strong requirements that are not always taken into account when people create gear:

- The sound generated by the Rockman line is directly usable by the sound engineers, thus bypassing several sound processing steps while making a record.
- The sound of a Rockman rig will stay the same, day after day, week after week, year after year, unlike vintage gear based on older technologies.

❑ You have certainly noticed that we didn't speak about amps nor cabs here: that's normal, since the Rockman concept is to get rid of them!

Yet, SR&D has sold some huge 2x250W amps with their matching 3-way cabs: they must be considered as a monitoring system, a sort of personal PA system for the guitarist, and have no role in the sound.

SR&D has also sold regular guitar amps, the most famous being the amazing programmable XP100 and the A12-50 combo: but these items were produced in limited quantities, and were not the core of SR&D's offer.

❑ This revolutionary approach - playing guitar without an amp - was in fact logical if you remember that Tom Scholz was first a design engineer. Though he was now a musician and a producer, he has applied engineering methodologies that have proved for decades their efficiency:

- Top-down approach: never select solutions before defining objectives and requirements and identifying contraints.
- Functional analysis: the requirements that are used during the design are defined in terms of functions, never in terms of possibilities.
- Value analysis: a piece of gear aims at providing strictly the required functions, at the lowest possible cost. Over-design costs money for nothing in return and is prohibited.

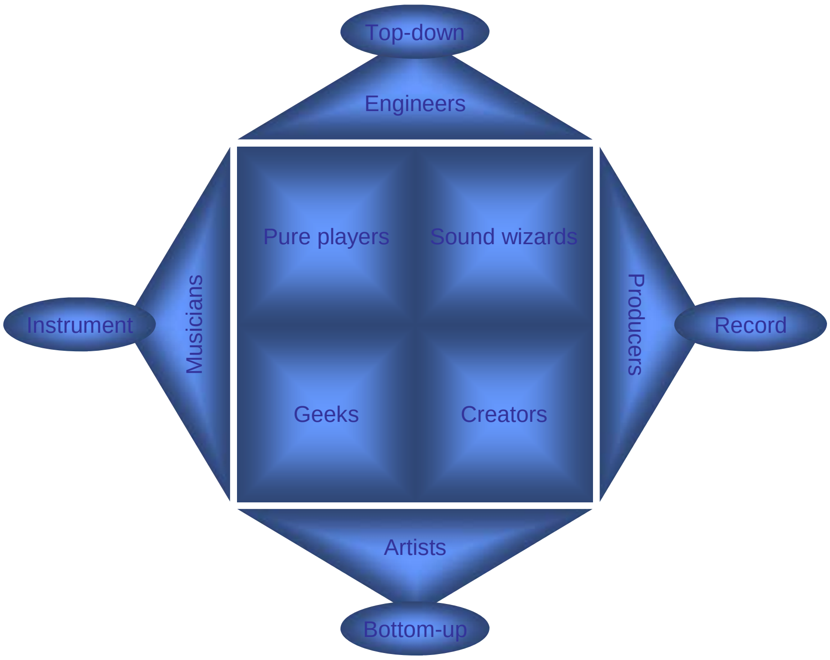
You follow a top down approach when you pack your bags for a week-end and say "I can live during two days without a second pair of jeans, so I take only one". You follow a bottom-up approach when you say "I have six dresses and eight pairs of shoes, and I bring them, just in case".

You are making a functional analysis when you say "I need to go to Paris in less than three hours", instead of saying "I will drive to Paris". In the first case, you can choose between train, aircraft and car. In the second case, you are already late.

You are using value analysis when you say "I need to go daily to my office" and buy a bicycle. You are wasting your money and damaging our planet when you say "I need a car, and I will use it to go daily to my office".

Well, if you apply these methodologies to guitar sound, you will rapidly admit that in most of the situations, there's no room in the requirements for a dedicated guitar amp: create the sound in small boxes, plug that in the mixer of the band or of the studio, and you're done!

❑ The chart below is an attempt to position all the gear approaches that one can cross one day or another. One can easily see that they are complementary, which means that we usually call an idiot anyone who is in the opposite corner...



The vertical axis corresponds to the global method used to create sounds. It ranges from bottom-up (the natural approach of someone who experiments new sounds with his gear and selects what he likes) to top-down (the methodology used by technical designers, to create gear that will respond to specific requirements).

The horizontal axis corresponds to the focus of the musician: instrumentists aim first at having a good musical sound and let the sound engineers do the rest, while a person trained to the studio approach will aim at providing a sound that merges easily into a global band sound.

- We have all, one day, been geeks who stack several stompboxes and amps without really giving a dime about the sound of the band.
- We can become a Pure Player: we have a sound in mind, and spend fortunes in the best amp and the best effects to get the killer sound we are dreaming about.
- We can also mature a little, and become a Creator, someone who keeps his arsenal of stompboxes and combos, but knows that the sound of the band is the result of a collective effort.
- The fourth quarter is a place where you will find the Sound Wizards: people who have a sound in mind, know how to achieve it, and have eliminated the eternal quest for weird sounds that may, some day, be useful to play music.

Tom Scholz belongs now to the wizards quarter - after having certainly experimented the pleasure of being a geek in his teen age. He spent several years, before creating the Rockman line, in the Creators area, fighting to get his sound out of the gear that was available.

❑ The result of his efforts is a consistent line of sound design tools for the guitarists who care about the collective sound of their band. The technical choices of SR&D are clearly high-end oriented, while the ergonomics and features of the Rockman items avoid the traps of generating weird and useless sounds. What you get when you start playing with Rockman gear is a guaranteed sound quality, and, most of all, you will control your sound instead of letting your gear do what it can.

❑ You may, of course, prefer other approaches. I will just quote freely Bob Cedro, who worked during 11 years with Tom Scholz, and told me something like "Consider that Tom has created Rockman like a painter: you can look at his painting, you may like it or not, but you cannot change what the painter meant when he painted it."

And now, like many Rockman afficionados, I look at the painting, and I'm still wondering if I really understood what the painter meant...