POWER USER

Motif XS: EQUALIZERS – VCM EQ 501

EQ, or Equalizer, is arguably the most useful and powerful weapon you have in your processing tool kit. It is the one professional tool that has been deemed safe enough for even the *average* consumer to use. But in reality it is perhaps the most misused processor in the field of audio. Hopefully, by the time you finish this article you will have a new perspective on the use of the equalizer and a new appreciation for what it can accomplish for you in terms of making your music shine.

The first introduction for many to the Equalizer is the so-called, *tone controls* on your home hi-fi system or on the radio in your car. Typically, you have a Bass knob and a Treble knob. People don't know what to do with them so they turn up the Bass to 3 o'clock or better, they have the Treble knob set at 3 o'clock or better. If they have a very nice system they may also have a Mid-range knob; they do not know what to do with it so they turn that up to 3 o'clock or better, too. If there is a Loudness Contour button, they punch that in, too. Why? … because it sounds louder and louder is better, right? Well, yes, louder always sounds better – whether it actually is or not, is the question. They are not sure about any of what they have done so they just turn everything up. If this article is effective at all, you will no longer be one of these people.

As a musician you have (or should have) an advantage over the *average* consumer, or non-musician, when it comes to intelligently using the equalizer. That is the purpose of this article: to make you aware of the natural advantage you have and how to put it to use. We will define several of the fundamental terms, which are all **music** related, that should help you learn to use the equalizer in your projects.

We mentioned on your home hi-fi the knobs are referred to TONE controls. You may be thinking, "I thought on a synthesizer the Filter was responsible for the instrument's tone". And you would be correct. An Equalizer is basically based on a series of filters and amplifiers. Indeed filters are responsible for what we as musicians refer to as tone or timbre, however, what does this mean in terms of what we hear and how we hear? The Equalizer is responsible for the **harmonic balance** of the music program. You can make a particular frequency band (range) louder (boost) or softer (cut) by raising or lowering the Gain control for that segment. In the Motif XS there are many different equalizers: at the Element level, at the Voice level, at the Performance level, at the Mixing level and finally at the overall Master output level. There are several different types of EQ: 2-band, 3-band, 5-band, single band parametric, straight boost. In this article we will explain the device in general and the new special VCM EQ 501 in specific. The VCM EQ 501 is an Insertion Effect and is a type of equalizer called a parametric EQ. This means that you have control over the Bandwidth (or Q), the Frequency range, and the boost/cut function. The other type of EQ you may be familiar with is a Graphic EQ – where in contrast the frequency and bandwidths are fixed at particular predetermined positions. The parametric allows you to select the frequency and the bandwidth of each.

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Background

Let's open our discussion with a look at sound. There is noise and there is music. Noise is chaos and music is order. What makes music pleasing is that there is structure and some consistency in the nature of the vibration. Sound is vibration. When it is random we call it noise and when it is structured we call it music. Picture in you mind's eye a guitar string or a piano string in motion. When a string, attached at both ends, is plucked/struck it goes into vibration and it will maintain a consistent number of vibrations per second. We call this consistent vibration its frequency – measured in cycles per second or Hertz – named after the scientist who did work in this area. The number, or frequency, of the vibrations is what musicians call its pitch. We have all heard that the "A" above middle "C" is identified as "A440". This means that any object that vibrates *consistently* at 440 times in a second will give off the pitch we call an "A".

Vibration does not necessarily have to be sound. I must state this as not all vibrations can be heard, after all, light is a vibration at an extremely high frequency. The frequency response of the human ear is approximately 20 cycles per second up to about 20,000 cycles per second. Your mileage will vary. This estimate can actually be slightly lower and higher depending on the individual. But this is the measurement for the average best pound-for-pound athlete on the planet, a seven-year old boy (go figure!). As you get older your frequency response naturally rolls off in the high-end – this works out great because the older you get the less you want to hear screeching music. We have all heard that dogs can hear higher frequencies than we human beings and that many animals apparently hear lower than we can. But vibrations that occur between 20-20,000 cycles per second fall into the human audio range.

We said the equalizer affects *the harmonic balance* of the audible program. **Harmonics** is a term we need to understand as musicians. Technically, it is the whole integer multiples of the fundamental note. Let's take the "A440" as an example. This is the fundamental pitch – to find the next higher harmonic you would multiply 440 x 2 = 880. To find the next harmonic you would multiply 440 times the next whole integer, 440 x 3 = 1320; and so on. So every whole number multiplied by the fundamental gives us a higher harmonic. Each harmonic has different amplitude (volume) giving each sound a unique identity. Harmonics are like the *fingerprints* of the sound. Every sound has a unique harmonic signature much like every human has a distinctly individual fingerprint. Our ear-brain can recognize sounds because it is uncannily able to recall a remarkable number of sound IDs.

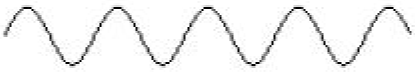
Here is an example. Say you are in a totally dark room and behind you and to your left someone drops a coin. You immediately turn to your left and say, "That was a quarter, approximately 10 feet away". Not only are you able to pinpoint where the sound occurred, you are able to identify the amount of money and from the sound you can now tell a lot about the room you are in and from what material the floor is made. A quarter has a distinctly different harmonic content from a nickel or a dime or a penny. And it does not sound like a half-dollar or a silver dollar. It sounds different bouncing off of wood then on metal or concrete. In fact, without a thought you realize that what I am saying to you is true. These are sounds you have heard (of course, if you live in Europe or Asia you must substitute coins of your own currency). But in your own mind you know you could tell them a part.

The sound hits your left ear an instant before it reaches the right (at 1100ft per second, estimate the time it takes to travel the few inches between your ears) and you immediately turn to the left. The reverberation and the phasing of the sound bouncing off the walls and ceiling, if any, immediately detail the environment – you instantly know if you are indoors or out, and how large an area you are in at the moment. And because of the keen awareness of your consciousness, you can even tell if the coin was dropped and from what height or thrown at you – the threat mechanism analyzes the situation. Cool!

You can recognize human voices by the harmonic content of the person speaking. Each of us has a unique harmonic structure to our voice. You can even tell a friend over a device with as poor sound quality as through a telephone. Telephones have a frequency response of about 300-3,000 cycles per second (so don't waste your time playing music to some one over the phone). But even in this severely narrow frequency region you are able to easily recognize your friend's voice from someone selling a new long distance service.

Say a trumpet plays an "A440" and then a trombone plays an "A440", do you think you could tell them a part? Of course, you could, no problem. Why? Because the harmonic content of the instruments are different – even though they are both brass instruments the shape dictates that they have different tone (timbre), i.e., different harmonic structure.

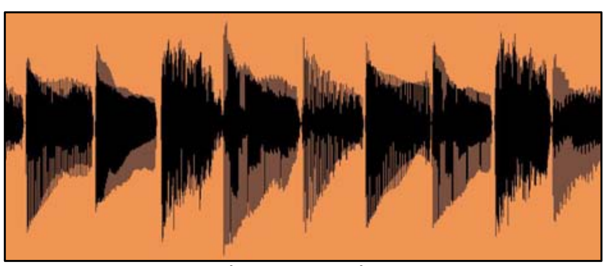
How does this happen? And what makes the



harmonic content different. We said harmonics are whole integer multiples of the fundamental and in any most musical tones there are actually several tones happening simultaneously. We will now talk about the harmonic series – and as musicians this should be somewhat familiar.

Picture a piano string in motion giving a pitch of "A440". If you have a high-speed camera and took a series of photographs you would see one where there was exactly 440 sine waves between the bridge and the nut. Another picture would reveal a contortion of the string with exactly 880 sine waves, and another would reveal 1320, and yet another 1760, and so on. When a string is sounding the fundamental "A440" it is also simultaneously, at softer levels, giving off the rest of the "harmonic series". Just *how loud* each of the higher harmonics is and when it occurs in time, is what gives each sound its unique signature. The ear-brain can detect minute variations in the volume of these harmonics and it is this that we identify when we identify a sound.

The relative volumes of the harmonics can even change overtime – this also is part of what the ear-brain memorizes and anticipates as it looks through its memory banks to identify the source.



It is very ironic that nowadays it is common practice to visually show the waveform in software and firmware samplers and recorders (diagram above). For us 'old-school' recording engineers nothing could be stranger – because looking at the waveform really tells you very little about how it sounds, while hearing it tells you everything. Could you tell a trumpet from a trombone by looking at the waveform? Doubtful… But our eyes are such needy things. The ear is phenomenal and makes the best identifier when it comes to sound – be it random or structured. It can hear sounds a trillion times louder than the softest sound it hears. A trillion! 1,000,000,000,000

Okay, let's take a closer look at **frequency**. As musicians you know that each time you double or halve the frequency you move up or down a musical octave. The piano is a great instrument to do this on because very few instruments that go as high go as low, and very few instruments that go as low go as high. We said the "A" above middle "C" was "A440". The "A" below middle C is "A220", and the "A" below that is "A110". And below this we have "A55" and the lowest note on the piano is "A27.5". Going up the scale we have "A880", "A1760" and "A3520".

Just because I'm sure you are curious the highest note on the piano is "C4186.009". So the full range of a concert grand piano is from "A27.5" ~ "C4186.009". We said the human ear can hear from 20-20,000. That leaves a lot of room between the highest "C" and the highest audible frequency. What is in that area? Is it important? Harmonics; and are they important!?! …You bet. They are what give sound its intelligibility. Without the upper harmonics the sound is dull and uninteresting to the ear-brain. The way it was explained to me when I was a young recording engineer is:

If the eye is drawn to things that shine and sparkle, then it is the upper harmonics that sparkle for the ear.

Okay, what does all this have to do with the equalizer. A knowledge of the pitch (frequency) that is sounding will help you when it comes to using an equalizer – relating that to the musical instrument or keyboard will definitely help you target the proper area. A working knowledge of the harmonic series and the importance of upper harmonics will help you when you are manipulating the **harmonic balance** of the sound. This is what you do when you EQ something - you are changing the balance of the harmonics by making certain frequencies louder or softer. And the danger is that you run the risk of making a sound, unrecognizable.

The natural instinct is to listen to a sound and look for what you are not getting enough of… this is why most civilians (a kind term for the "technically unwashed" or average consumer) **only boost** with their equalizer controls. Once you learn to listen closely, you start to hear something and say to yourself – 'what am I getting too much of...' By removing some problem areas you can actually make what you want stand out. If you only boost all the time you sometimes only compound the problem. We'll give you a good working example of this with a Kick drum later in the article.

Equalization in the Real World

I became a recording engineer the old fashioned way… I ran for coffee, setup microphones, cleaned tape machines (yes, it was the stone ages). I did an apprenticeship at a small jazz studio in New York City. The studio was affiliated with a record label and it was a very comfortable atmosphere both to learn and to record music. I got there because I needed to know how to get *the sound* I wanted when I recorded my band. I realized in my first recording experience as a musician that I could not express myself to the engineer. I knew what I wanted but could not speak 'the language'. I was young, and there is nothing wrong with being young as long as you realize that you are and that you have time to learn. So that is what I did. I said to myself: "Self, you need to learn about this recording stuff if you are going to be a musician." And that is what I did. After each session I would ask the engineer, when this happened why did you do this, and when that happened why did you do that… it was a process. Finally came the day I was "flying solo" – doing my first recording session.

It was the guitar player's date. He showed up hours early. I knew he was serious because he had his own foot-rest and his hands were meticulously manicured. I got him a comfortable chair, setup a pair of U87 microphones – one over the tone whole and one above the nut and fret board. I spent about 5 minutes listening to him play, right there in the room, and then retired to the control room. I then went back in and adjusted my microphone positioning until I found a good distance. Notice I did not immediately go to the console's EQ (and assume my first mic positioning was golden) – ***moving a microphone is always a better first solution than using an EQ.*** The distance from the source is a better harmonic balance than a knob on any EQ, period. I was pleased. I was getting this big, warm rich tone from that acoustic guitar. So big you felt you could walk between the strings. Oh, yeah, this was good. In walks the bass player. I set up a U47 (a favorite back in the day for acoustic bass). And I was getting a big, warm rich tone from that acoustic bass.

Now the guitar and bass started playing together. Hmmm! The big-warm-rich guitar was getting in the way of the bass. I made some adjustments to the EQ so that the guitar and bass worked together. In walked the piano player. We had a very fine piano in the studio and I had my own 'thing' for recording piano – it is my instrument after all. I did my 'thing' on the piano sound. Now the three of them started to play together, and I realized, once again, I needed to adjust the bass and the guitar to accommodate the piano. Okay. After some more minor adjustments, I was happy with the sound of the three of them.

In walked the drummer, and 40 minutes later I had a decent drum sound… Drums… need I say more? Then the flute player… I put him in the vocal booth. This allowed that I could still get clean isolation yet not have the microphone so close to the flute that it would give an unnatural sound. The isolation booth allowed the microphone to be far enough away to give the flute some airy-ness. Mic the nose or the chin of the player (or singer) – this avoids all the plosive wind puffs that can sound like a storm. You have to tell them to ignore the mic position, because they are used to being right in the mic because live it is a battle to be heard – so good mic technique goes out the window. Most people have no concept how to get the best sound from a mic – they never get beyond fighting to be heard.

Okay, a good hour and a half of setup and we are ready for a run through. This is when the big learning moment occurred – I call it the "AHA! Moment" (some say "Eureka", same thing). They were running through the first number and it was sounding excellent – really excellent. Everything well balanced and clear – I could hear everything just like I wanted it. The learning moment came when I pressed the solo buttons in turn on each instrument. That is when it hit me. I soloed that guitar – and there was this much thinner guitar sound, nothing like the big warm rich sound I had initially. I undid the solo button and the guitar sat right in place with the other instruments. I soloed the bass – and there was this much thinner bass sound, nothing like the big warm rich sound I had initially. I undid the solo button and the bass sat right in place with the other instruments.

Wow! No way I would EQ these instruments like this as individuals – but they were perfect in the context of the ensemble. They supported each other well. And the bottomline is that is what you are going for, the ensemble sound. Because it is a band not a bunch of individual sounds to be heard alone. It is all about how they work together. AHA! Eureka… hurray! I had learned the first and most important lesson in recording – make the final product work. This was not "solo" instruments, this was an ensemble.

VCM Technology

The geniuses at Yamaha's K's Lab[1] in Hamamatsu have created (recreated) the response and behavior of the classic Equalizers of the past. What VCM (Virtual Circuit Modeling) does is not just mimic the device's results (as is the case with most of the modeling devices out there)… what they have done is model the actual electronic components used in the classic equalizers of the past. The tubes, resistors and capacitors used to

[1] K's Lab is a design team at Yamaha Japan headed up by Toshi Kunimoto (the designer of the modeling synthesizers VL1, VP1 and technologies like FDSP).

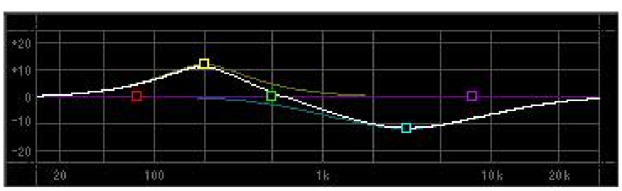
build these devices, their tolerances; their response to heat variation and how they were used in the actual circuitry of the devices is what has been modeled. This means that the input-output signal flow will behave the same as the original device under similar conditions. By modeling the components and the circuitry they can virtually build almost any device. The unique charm of these classic devices is what has been captured with the VCM technology.

VCM EQ 501 Parameters

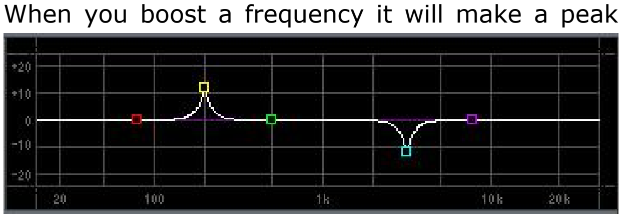
A close look at the front panel of the VCM EQ 501 will reveal why it is called this: it is a 5-band EQ.

The three brown knobs on the top row are labeled “Q”. The second row of five knobs is the “Frequency” setting. The five knobs of the bottom row are the “Gain” boost or cut. In the upper right corner is the overall OUTPUT.

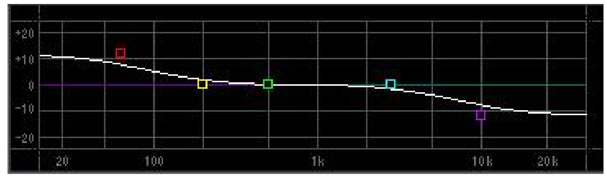
Q – is a term that denotes how wide a frequency umbrella each band will cover (and you thought it



was a group of omnipotent beings that live in the Q-continuum); Another word for “Q” is Bandwidth. When you boost a frequency it will make a peak



(looks like a mountain peak) and when you attenuate a frequency it will look like a valley.
A wide setting (a low Q number, top diagram) will cover several octaves of frequencies, while a narrow setting (a high Q number, bottom diagram) will be a very small range of frequencies. Notice the lowest band (1) and the highest band (5), do not have a “Q” control. This is because these two ranges are not peak/valley type, they are a type that is referred to as “shelving”.



Rather than create a peak (mountain) when you increase the gain, it creates a plateau (flattens) and all frequencies below the low band are effected. In the case of the high band all frequencies above are effected. Basically they act as a HPF at the low-end and an LPF at the high-end. An HPF is a High Pass Filter – which you can take literally: it allows higher frequencies to pass and blocks the lows. And conversely, a LPF or Low Pass Filter allow the low frequencies to pass and blocks the highs.

FREQ – is frequency. And is simply, as we have explained, the center pitch of the peak or valley. And in the case of the shelving bands it is the point at which all frequencies below or above start to be boosted or cut. Frequency is measured in cycles per second, or Hertz (Hz); a parameter named after the man did groundbreaking work in this area.

GAIN – is output for that band. If the Gain is set at 0 none of the other settings matter, as no boost or attenuation can take place. Gain is measured in dB or deciBels; Bell is Alexander Graham Bell, who did groundbreaking work in microphones, speakers, telephone, etc. dB is a ratio – a comparison of one setting to another. In general, a change of 1dB is hardly audible, 2dB change will be recognized by a majority of listeners (if they are paying attention), while a change of 3dB is clearly audible to everyone.

OUTPUT – In the upper right corner you see the **OUTPUT** knob. This is the overall output post the EQ. When signal first enters an EQ it passes through a filter which drops the overall level. The OUTPUT is to ensure that you can return the signal to what is considered UNITY gain – so there is not overall loss of level. An equalizer is a combination of filters and amplifiers. As we will learn, you do not always boost (at least not if you are wise), the OUTPUT can be important to give you back unity output post the signal going through the equalizer. Believe it or not, cutting (attenuating) the output gain at certain frequencies can be beneficial to a sound.

VCM EQ 501 Parameter Ranges

The **Q** setting (also referred to as Bandwidth) is available on the center three bands. The widest setting 0.50 through to narrow setting 16.0

The **Frequency** ranges are as follows:

Low (band 1)	**31.5Hz ~ 2.0kHz**
Low-mid (band 2)	**50.0Hz ~ 20kHz**
Mid (band 3)	**50.0Hz ~ 20kHz**
Hi-mid (band 4)	**50.0Hz ~ 20kHz**
High (band 5)	**500Hz ~ 20kHz**

The incremental steps between frequency settings are at 1/12 of an octave (as a musician you would understand this as approximately every half-step through out the music range). Without going into a discussion of Equal Temperament scaling – suffice it to say that this tuning is based on an octave being divided into twelve equal parts.

The **Gain** range is plus/minus 12dB for the two outer (shelving) filters and +/-18dB for the 3 peaking bands. You should resist the urge to use the Gain controls at the extreme setting – it is not that it is never necessary, it is just that plus or minus 12dB is a huge amount and plus or minus 18dB is a tremendous amount. Do not turn a knob like a guitar player… sorry for that one, but as you know, guitar players amps are built so you turn everything all the way up (to 11), **don't**, this is professional audio gear. It is the very rare occasion that you need that type of severe Equalization. If you think you need that much, you might want to consider that you have selected the wrong sound.

The **OUTPUT** can be increased or decreased by +/-12dB.

In the upper right corner of the VCM EQ 501 you will see that you can select from several PRESETs:

Flat – a condition where all settings are neutralized (no gain or attenuation at any frequency). There are only two other presets: "**Radio speaker**" and "**Dance BD & SN**"

Typically, you cannot pre-set equalization, as it always will depend on the type of signal coming in and this can vary per session. The preset given here are special general 'effect-type' equalization settings that when applied to program will give the general overall feel of an old radio program or a typical dance bass drum/snare combination.

The object, when working with an equalizer, is to listen to the program (the source signal) and adjust the sound accordingly. Let's do an experiment to get a feel for how to approach equalizing a sound. Call up the following Drum Kit in VOICE mode:

PDR: 004 (A04) Dry Standard Kit

What we will do is route the kick drum that is on note C1 to the INSERTION "A" effect. INSERT "A" is already assigned to VCM EQ 501. Within a drum kit you can route each individual drum either to the System Effects (Reverb/Chorus) **or** to an Insertion Effect A or B

- Press [EDIT]
- Press Track Select [1] to view the ELEMENT-EDIT (Key) parameters.
- Press note C1 on the keyboard to recall the kick drum parameters
- Press [F1] OSCILLATOR
- At the top of the second column route the note to INS A

- Press [COMMON EDIT]
- Press [F6] EFFECT
- Press [SF2] INS A to view the parameters of the VCM EQ 501

Recall the FLAT setting. Do so by highlighting the Preset parameter (upper right) and press [SF6] LIST. Now move away and re-select FLAT. This will "zero out" the EQ Gains to +0dB.

- Move the cursor down and highlight the MID band.
- Change the **Frequency** from 1.00kHz to **800Hz**

This is what we used to refer to as the "cardboard" frequency for kick drums. Here is what I mean:

If we want to equalize a kick drum what you must ask yourself is: "What about this sound is important?" The kick drum (also called the Bass drum) is a part of the drum kit that is struck with a foot-pedal. There are two distinct parts to the sound.

1) the mallet striking the drumhead – "click"
2) the response of the wooden shell – "boom"

These are at two distinctly different frequencies. Any percussive strike is going to make a sharp spiky sound – in this case it is typically over 1kHz and can be as high as 2kHz. The response of the bass drum body (shell) itself is down low and can be 250Hz and lower down to as low as 80Hz or even lower (it depends on the drum). It helps to picture in your mind's eye the sound in slow motion. Imagine a mallet striking the drumhead, "click", and then imagine the drum responding "boom".
If you need more of the good stuff… (please take notice that I didn't say "if you need more bottom"… because I am not just interested in the

bottom[2]). I want the click of the mallet strike to give it definition… by getting rid of the cardboard frequency, 800Hz, I should hear more bass response and more attack. Instead of boosting, try lowering the GAIN of the MID band at 800Hz, the more you lower it, the more bottom you hear and the more the attack stands out. You accomplish the same thing as if you had boosted the bass and boosted the EQ bands at the prime frequencies for the bottom and top ends of the sound.

Do you feel that? As you lower 800Hz, you actually perceive it as having increased the bass. For fun, try boosting 800Hz (cardboard)! Hmmm! Hope you are having an AHA! moment.

It is a good philosophy when listening to a sound to think about not what do you need *more* of, but what can you *remove* to improve the sound. This will combat the natural urge to always boost. I am not presuming to tell you that you should never boost or that boosting the output of a frequency is bad. I'm just trying to impress upon you that in the digital world, many users are **distorting** their music (it is an epidemic really) simply because they never really learned about the fundamentals of audio. Each small distortion may not cause the catastrophic digital noise that obliterates all sound, but what it does is add to an overall harshness of the sound. You can avoid overloading anything if you think "less"… Less is more.

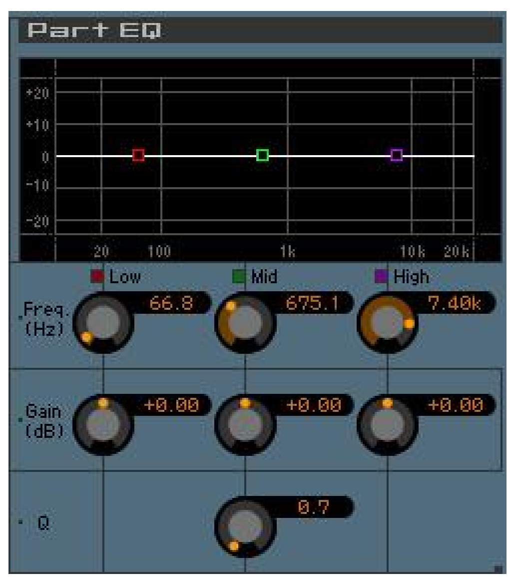
Some Things You Should Know

Equalizing sounds in a synthesizer is quite a bit different from equalizing signal coming in from a microphone from the following standpoint: The sounds in your Motif XS have already been recorded. They have already been EQ'd. They were recorded in the highest quality situation that Yamaha R&D could muster. An unbelievable amount of care has already been taken to ensure that the sounds you hear are ***harmonically balanced*** and useable within the synthesizer. Keep this fact in mind as you tweak sounds. If you find yourself radically altering the harmonic balance of a sound, perhaps you have selected the wrong sound. Perhaps there is another sound closer to what you are envisioning with your mind's ear. When applying the VCM EQ 501 to a microphone input[3], keep in mind that placement of the microphone is your first control for harmonic balance. If you are using a unidirectional microphone, there is something called the *proximity effect*. The closer you are to this type of microphone pattern the more bass response you will hear. And the distance from the microphone will always make a world of difference in how the signal sounds.

The VCM EQ 501 is always found as an INSERTION EFFECT. This means you can use it on as many as eight PARTS of multi-timbral mix. I never worked in a studio that had 8 EQ's of this quality (and I've worked in some of the best studios in the world). The VCM EQ 501 is based on the components of some of the classic top-shelf vintage EQs. Remember you will not need to equalize every instrument, however, the Motif XS gives you additionally a PART EQ for each of the 16 multi-timbral Parts of a MIXING setup.

PART EQ

Each channel of a MIXING setup will have its own PART EQ. This is a 3-band parametric equalizer, peaking MID-range, with the high and low bands set to shelving.



See you know what that means now! If, as I mentioned the sounds in a synthesizer like the XS have already been meticulously equalized and recorded, why then all this available EQ? Excellent question… the answer would be to create the

[2] Of course this can vary depending on the type and genre of music you are doing.

[3] This will be the subject of another Power User article on Sampling in the Sequencer.

ensemble sound. Remember my story about my first session. What I learned was that even when you make an input sound "near perfect" by itself, that does not necessarily mean that it will work in an ensemble. It is the subtle adjustments that allow the bass to carry the bottom end and not the guitar, to allow the drummers cymbals to sparkle in the upper regions. You are looking for gaps to fill and to clear harmonic areas so things can be heard. Pan positioning also is a tool that should be mentioned – by positioning an instrument in the mix you can give it "space". You cannot feature every sound – it is not a featured sound if every sound is featured.

VOICE MODE EQ

In Voice mode *every* Voice has its own 3-band EQ that can be stored within each Voice. This EQ, is a similar unit to the PART EQ described above; it has a parametric Mid-band and shelving Low and High bands. This EQ is useful when playing a sound "live" in VOICE mode. It will be stripped off when the VOICE is placed in a PART (either in Performance mode or in Song/Pattern mode). This is pretty standard. Because as we have tried to point out, when a VOICE is combined with other sounds in an ensemble you can count on needing to rethink the EQ for that sound. You reach this VOICE mode EQ from the main PLAY screen by pressing [F3] EG/EQ.

MASTER EQ

The Master EQ functions as an overall equalization of the entire mix output. It is a 5-band EQ much like the VCM but not of the same design.

Notice that above the Low and High bands you can select the "Shape" – this selection is between a peaking type or shelving type filter band.

Summary: We did not get involved with the individual Element EQ – basically because that will better be dealt with in an article on Voicing or Sampling. But in way of summarizing the Equalization process we must state that while there is some right and wrong, for the most part equalizing is subjective. This means it is really up to you. Used wisely EQ can help the clarity and intelligibility of your mix, and used poorly it can turn a good sound to mud. In general, EQ individual sounds to sound as good as you can make them. But remember once you place them in an ensemble their role changes and so should the equalization. The Master EQ is one that you can use to adjust the overall sound. Let's take a look at how you might be using your Motif XS.

You will want to use the Voice mode Insertion Effect EQ (VCM EQ 501) to make a Voice sound balanced and clear. An Insertion Effect is part of the Voice itself. For the very important sounds (featured sounds) in your Mix, when using the sound in a sequence, having the VCM EQ 501 is a big advantage. Remember any 8 PARTS can recall their Dual Insertion Effects from Voice mode. You will want to treat the VCM EQ 501 as something <u>special</u>.

When you go to play live, you do not want to have to adjust this equalizer (as this is like editing the VOICE), you should, if at all necessary, use the Master EQ to adjust your sound for the room that you are currently playing. This is where the Master EQ is used. The Master EQ is the last thing before the outputs.

The MASTER EQ is global for VOICE mode, and is available for storage in each PERFORMANCE and each MIXING setup.

The VOICE mode Master EQ settings are found in:

- Press [UTILITY]
- Press [F3] VOICE
- [SF2] MASTER EQ

(Note that this is actually a 5-band equalizer and measurements are in Hz (freq), dB (gain) and Q curve). This Master EQ will effect each and every VOICE, in addition to the aforementioned 3-band EQ and the Insertion Effect (VCM EQ 501).

In a Performance and Song/Pattern MIX the MASTER EQ is local within each setup. Each Performance or MIX is its own autonomous world. To see the Master EQ in a PERFORMANCE or MIXING setup:

- Press [EDIT]
- Press [COMMON EDIT]
- Press [F2] LEVEL/MEF
- Press [SF3] Master EQ;

The MASTER EQ will equalize all the Parts of the PERFORMANCE or MIX together.

The Equalizer affects overall output level. It increases output level when you boost and it decreases or attenuates level when you decrease. We need to mention one more item about your ear-brain and its ability to hear. We said that it can compress sound over a range of 1 trillion times the softest sound you can hear, and that its frequency response was approximately from 20Hz through to 20kHz, but this changes when the volume and temperature change. We don't have to worry about temperature too much, but the volume level that you playback your music has a profound influence on the frequency response of your ears.

Discovered by Sir Isaac Newton and explored by two scientists, Fletcher and Munson, the Fletcher-Munson effect describes the phenomena as you lower the volume (intensity) you are less likely to perceive high and low frequencies. Story goes: When Newton was sitting under that famous apple tree pondering gravity, he noticed that as the sun went down, he could distinguish the green of the leaves for a longer time than he could tell that the apples were red. This indicated that his eyes ability to differentiate frequencies of light (color) changed at different intensities. The same is true with sound; as the overall volume is lowered you hear less at the extreme ranges. That is, you hear less lows and less highs but you continue to hear the mid-range well. This is the purpose of the Loudness Contour button on your home stereo. It adds a little extra punch to the lows and highs and is intended to only be used when you have to playback at very low volume – when using music as a backdrop for a meal or behind guests talking. Basically, when the music is not the feature – it adds a "smiley curve", called this because the boost at the low-end and high-end makes a curve that looks like a smile. Punching it in at any other time is simply a misuse, or a lack of understanding as to what it is designed to do.

If you only want to make the overall signal output louder, don't do it by turning up the EQ - simply turn up the main VOLUME output. (The one control that is 100% consumer safe is the VOLUME knob!) Most home hi-fi systems and car radios are balanced harmonically very poorly – and it is mainly operator error. And this is due to the fact that the average person does not understand the difference between the Volume control and the Equalizer. They only know to turn it up. The scary thing is they think this is a good sounding audio system. A friend asked me what I thought of his car stereo, I've learned to ask first if they really want me to answer or are they just making conversation. My years of working in the recording studio make me very unforgiving of the typical car stereo – in spite of how much you can spend on it. And please don't ask me about the typical computer speakers…!!!

Last thing: Your studio monitors; find yourself a good pair of relatively flat studio monitors. Flat does not mean that they do not sound good, not at all. Flat means that they do not **color** the sound – they give you what you send in – and, trust me, that is what you want from your monitors. Nothing extra, nothing missing… just what you send in. If you are equalizing to compensate for a deficiency in your speakers, this is a bad thing for your mixes. Yamaha NS10M gained a reputation for "not lying" – giving you just what you sent in. Again, you will hear some people say "NS10M's had no bottom". Not true. They did not hype the bottom end. Placed at an optimum distance from the engineer (called the "sweet-spot"), they were near-field monitors, after all, the NS10M would give exactly what your mix was doing. They became a standard because of this. Today, studio monitors have built-in amplifiers allowing the design engineer to ensure that the tweeter gets exactly the right amount of power versus the woofer – separate power amplification for each. This insures that the concept of the design is heard as intended.

The typical consumer buys speakers based on comparison of volume (which is mainly its efficiency, not its quality). They will buy the loudest speaker – louder sounds good to us. But all systems have a way to control the listening volume (a big knob).

- What do the speakers sound like at equal volume?
- How is the stereo imaging?

If you are comparing speakers at a store resist the urge to be the bozo who buys the loudest speaker – ask them to adjust the volume so that you can determine how they **really** sound at optimum listening volume. The volume you can balance – the quality you cannot change.

It takes years to get used to listening at an *appropriate volume* – if you listen too loud, everything starts to sound good. If you listen too soft you get less bass and less highs. These things you must anticipate, learn, know and know how to use to your advantage. While the *appropriate volume* will make most musicians say, "turn it up"; it is the best listening level when trying to mix music. If it works at moderate volume it will work when you make it louder - but stuff mixed loud does not necessarily translate to lower playback volumes. I'm sure you have recordings in your collection that only sound good when played back really loud and yet you have others that seem to work at all playback levels.

Having knowledge of how your ear's response changes with volume levels is important. Serve you well, it can, hmmm! Hope there was an "AHA!

Moment" for some of you in here somewhere and that it served as a good review for you others.

Enjoy!

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