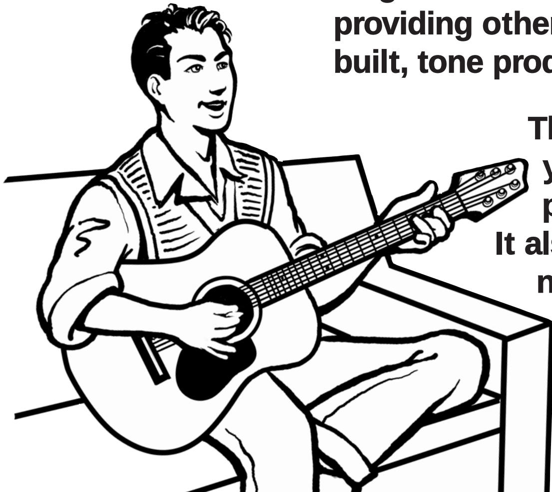
duncan|turner acoustic research

EQUINOX™

Instructions for EQUINOX Parametric Preamp

Ever notice in certain live or studio settings, you sound truly great? Notes seem to leap out of your instrument. Your tone is perfect. Your ability to express yourself is heightened. Unwanted feedback vanishes. And other times, you sound, well… flat?

One of the primary reasons you sound better at certain times is the gear in your signal chain, specifically, the all-important preamp-equalization stage. We're D-TAR, a company founded by musicians and dedicated to providing other musicians with highly effective, easy-to-use, quality-built, tone products. Equinox™ is just such a product.



This manual will show you how to make the most out of your Equinox. You can use Equinox as a three-band, true parametric, equalizer for "surgical" control of your tone. It also works as a stand-alone preamp. And it has a two-band notch filter to eliminate unwanted feedback without killing your tone. Equinox is a truly versatile piece of high-end audio gear. But most importantly, it will help you achieve truly great sound. Every time.

"WITH RESPECT TO ACOUSTIC TONE"

Introduction & Product Overview:

First, we'd like to thank you for purchasing the Equinox™ preamp/parametric equalizer. We hope you will get many years of use and musical enjoyment from it.

Equinox is the most versatile equalizer/preamp in its class. It can be used as a preamp with electric and amplified acoustic instruments or it can be used in conjunction with a dedicated preamp or mixer/blender such as the D-TAR Solstice™. Designed for stage or studio use and made to meet stringent pro-audio specs, the Equinox™ parametric EQ can enhance the tonal response of your instrument and sound system as well as control feedback under difficult sonic circumstances using the two tunable notch filters. Both parametric EQ and notch filtration can be independently switched in or out, so you can use the functions you need, but completely bypass whichever function you do not need.

Using Equinox as a Preamp

Unlike most equalizers, Equinox can be used as a standalone preamp for ultra portable instrument amplification systems. In this mode, simply plug your instrument into Equinox™ input, and then plug Equinox™ output into a power amp and speakers. Equinox features a 20 dB gain switch to choose between instrument level signals or line level signals normally associated with pro rack gear. The "+20dB" button, when depressed, provides an additional 20dB of gain in the input stage. This changes the Equinox from a unity gain, Line level device, to an Instrument level device capable of accepting low level input signals and functioning as a stand alone preamp. Used either with a separate power amp and speakers or matched to a "powered monitor," Equinox may be all you need for amplifying acoustic instruments, keyboards, or bass. The high input impedance of the unit makes it compatible with piezo pickup systems whether or not they are equipped with an internal preamp. With the versatile EQ and notch filtration, Equinox can be counted on as an elegant solution to tricky amplification challenges.

Setting the Input Level

The first step is to determine whether or not to use the +20dB gain boost. In general, the +20dB button should be engaged for signals that are 0.5Vrms or less. Most musical instruments put out signals that fall into this category. Effect loops that are classified as –10dBu will also generally work with the +20dB switch engaged.

For signals that are between 0.5 and 9.0Vrms, the +20dB button should be disengaged (out) or there will be excessive clipping. Line level devices fit into this second category. This includes effect loops classified as +4dBu.

If you are unsure of the signal level coming out of the device you are using to drive the Equinox, use the following procedure to determine the best setting:

1. Turn the Output Level control all the way down.
2. Turn the Input Level control all the way up.
3. Disengage the +20db button.
4. Plug the output of the driving device into the input of Equinox.
5. Play your instrument at the hardest level that you expect to achieve during your performance. You should see the overload indicator light up occasionally on the highest peaks. If you never see it light at all, try engaging the +20dB button. You should now see the indicator light up.
6. If the indicator seems to be lighting more often than just the peaks, reduce the Input Level control just slightly. In nearly every situation, the best noise performance will be achieved by setting the input level control between 3 o'clock and maximum level.
7. If you are driving a power amp directly from the Equinox, set the Output Level control to produce the desired playing level. If you are driving the return of an effects loop, it will be necessary to carefully set the Output Level to avoid overdriving the next device in the signal chain.

Using Equinox' EQ Functions

Parametric EQ takes a little practice to use, but allows very precise control over tonal shaping. You can think of Equinox as a sophisticated three-band EQ with boost and cut controls for each of the three frequency bands, much like the EQ on Solstice or on many musical instrument amplifiers. But with parametric EQ you have greatly enhanced control over the exact center frequency of each band as well as having control over how wide a band is boosted or cut on either side of the center frequency.

To get a feel for how parametric EQ works with Equinox, get down to basics: Switch off the notch filters by pushing the Notch Bypass button in. Next, make sure the EQ Bypass button is out. Now, turn the three "Gain" knobs to the 12 o'clock "no effect" positions. With no boost or cut, you should hear little or no difference whether you switch the EQ Bypass button in or out. With the button in the out position (EQ engaged, not bypassed), turn the 200-2KHz Gain control up to about the 3 o'clock position, the bandwidth control to the 12 o'clock position and note the change in tone. Try sweeping the Frequency control from the lowest point to the highest. You are changing the center frequency or note at which the EQ is taking place. Next, leave the Frequency knob in one place, and try changing the Bandwidth, you will hear the EQ effect go from broad band to very narrow as you focus the EQ down almost to a single note. Now try more radical combinations of Bandwidth and Gain to hear those effects. You will probably never need to use the most extreme combinations of EQ and Bandwidth, but you have the capability of drastic tonal manipulation at your fingertips.

Having practiced using parametric EQ in one band of frequencies, you can now work with all three bands to enhance the tone of your instrument, tame "wolf" notes, and generally manipulate the sound spectrum. You can simultaneously enhance the deep low end, tame midrange "squawk", and bring out upper harmonic shimmer on your instrument. Or you can find a perfect sonic slot in a mix by rolling off lows and highs so your guitar or other instrument doesn't occupy another instrument's space. It is easy to overdo it with EQ, and it is always worth checking to see if what you've set up with your EQ knobs is an improvement or not. Using the EQ bypass switch, you can get a quick "reality check" whenever you need to.

Notch Filters - General Information

One of the biggest challenges in amplifying acoustic instruments is the feedback started and fed by particularly resonant air chambers or soundboards. The paradox of acoustic instruments is that the more acoustically live they are, the harder they are to amplify. Also, the larger they are, the more they will tend to feed back. It is much more difficult to get loud onstage volume with an upright bass, acoustic bass guitar, or dreadnought than with a mandolin or fiddle; as the tops start to act like microphones, and the low resonant frequencies of the air chambers couple all too readily to room and stage resonance.

Feedback frequencies tend to be quite specific, so if you can reduce just that one narrow offending band sufficiently, you can gain several dB of overall volume level on stage, while maintaining the tone you like. The notch filters on the Equinox are very narrow bandwidth, cut-only circuits. They can be precisely tuned to offending feedback tones and used to bring those frequencies under control while having little or no impact on neighboring frequencies.

Equinox features two bands of notch filter control with a wide range of overlap to enable you to notch out (or “cut”) major offending frequencies. These two sections will help control the feedback that often occurs at the natural air cavity resonance of a guitar body (you can also try a sound hole plug for this), the primary top resonance which is often close in frequency to the air resonance, and/or some other stray frequency which might be particularly bad.

Using the Notch Filters

Equinox has two bands of notch filtration with one covering a tunable band from 40 Hz (close to the low “E” on a bass) to 400 Hz (just below “A 440”) and the second covering 80 Hz (close to the low “E” on a guitar) up to 800 Hz (well into the midrange). Note that the “Gain” knobs of the notch filters are cut only, unlike the EQ knobs on the parametric stages that are boost/cut. They should be rotated counter-clockwise to cut; in other words, fully clockwise rotation will be flat. In using the feedback controlling notch function of Equinox, set your sound level to just below the onset of feedback...the point where you hear the resonant ringing of the instrument and room just starting to rumble. Make sure the “Notch” bypass switch is off and the notch filters are engaged. Start by using only one section of the filter by turning its “Gain” knob to the maximum cut, and then sweep the Frequency control through range. You should find a point where the ringing diminishes and you are able to then raise overall system gain without feedback. When you have reached the desired stage volume, try backing off on the amount of cut you are using so the notch filter affects less of the bandwidth on either side of the feedback note. Continue backing off the amount of cut until you just begin to hear the ringing again, then increase the amount of cut slightly. If you find it necessary to increase your stage volume during the night, it will probably be necessary to increase the amount of cut in one or both of the notch filters. The trick is to use as little cut as possible and still not have problems with resonant feedback.

If you then find another frequency which is feedback prone, you can use the same process on the second section of the notch filter to locate and suppress it.

Patching Equinox into an Effects Loop

You can also "patch in" your Equinox for live sound or studio applications to gain ultimate control over your sound. By using one or more Equinox in each effects loop(s) of a preamp such as D-TAR's Solstice mixer/blender you can gain the ultimate control over your live sound, the sound of instruments in the studio, or "fix it in the mix," should you discover sonic problems in recordings. Equinox is compatible with any guitar or bass amp with an effects loop, as well as with PA and recording consoles. Using an Equinox together with a Solstice can be accomplished with a single stereo cable connected from the “Input” jack of the Equinox to either of the “Insertion Point” jacks on the Solstice rear panel. Input and output connections can be made on a single cable, keeping cabling simple and clean.

For more information, see Setting the Input Level on page 1.

Front Panel

Technical Specifications:

Input stage Extremely low noise and low distortion (<0.001%). Gain adjustable with external switch from unity to +20dB.
Maximum input signal 10 VRMS @ unity gain setting / 1 VRMS @ 20dB gain.
Slew Rate 22V/us
Input impedance 4.7 Meg ohms
Low Frequency Band
- Frequency adjustable from 39 Hz to 408 Hz
- Q (bandwidth) adjustable from .58 to 9.4 (~2.5 to .15 octaves)
- 15dB of boost and cut available

Midrange Frequency Band
- Frequency adjustable from 212 Hz to 2.35 KHz
- Q (bandwidth) adjustable from .58 to 9.4 (~2.5 to .15 octaves)
- 15dB of boost and cut available

High Frequency Band
- Frequency adjustable from 1.8 KHz to 20 KHz
- Q (bandwidth) adjustable from .58 to 9.4 (~2.5 to .15 octaves)
- 15dB of boost and cut available

Notch Filter #1
- Frequency adjustable from 40 Hz to 400
- Q (bandwidth) fixed at 9.0 (~.2 octave)
- 15 dB of cut only available

Notch Filter #2
- Frequency adjustable from 80 Hz to 800
- Q (bandwidth) fixed at 9.0 (~.2 octave)
- 15 dB of cut only available

Misc. Features

Input Jack 1/4" Stereo, input on tip / output on ring, allows interface with D-TAR Solstice acoustic mixer using a single, stereo cable.
Bypass switches independently removes entire EQ and/or notch filter sections from signal chain leaving only high impedance, low noise input and buffered output.
Gain switch provides a 20dB gain boost. Allows selection of instrument level (enganged) or line level (bypassed) input.
XLR jack is a balanced signal output. It can be used to send a balanced signal to a mixing board or for recording a live performance.The typical level at the balance line out jack will be -10 dBu.
Power Supply Internal +/- 15V derived from 16VAC external, wall mounted power transformer.

Rear Panel

APPENDICES

A) A BRIEF HISTORY OF EQUALIZATION

The term "Equalizer" goes back to the time when few audio devices could reproduce a "flat" frequency response. The equalizer was designed to correct the anomalies, thus making frequency response equal at all points in the audio band. Some of the earliest audio spectrum shaping devices were passive filters similar to the treble cut controls found as "tone controls" on most electric guitars. These are "cut only" circuits, which can only take out the frequencies to which they are tuned using resistors, capacitors, and inductors (coils). One of the most famous of these cut only filter sets was the Altec "AcoustiVoice" system used to "tune" PA systems to rooms.

With the use of active circuitry, first with tube gear and then solid state, the next generation of equalizers could work in fairly broad ways to cut or boost bass, middle, or treble frequencies. The invention of operational amplifiers, ultimately miniaturized as integrated circuits, allowed much greater complexity in circuit design, and that allowed equalizers to become much more fine tuned to the needs of audio professionals.

Recording engineer and electronics designer George Massenberg generally gets the credit for developing the modern parametric equalizer. He did so to solve the thorny problems he came across during critical recording sessions that could not be solved using the relatively crude EQ of that time (1960s).
The problem with many EQ designs it that they work too broadly to take out or enhance specific parts of the audio spectrum.
In coming up with a way to address each parameter of the shape of possible EQ curves over the entire audio bandwidth, Massenberg helped invent one of the most versatile electronic instruments in the sound engineer and musician's toolbox.

B) WHAT IS PARAMETRIC EQUALIZATON?

"Parametric" refers to the three parameters that can be controlled with an audio equalizer:
1) The center frequency or note that is to be manipulated
2) The bandwidth (or "Q") on either side of the frequency selected
3) The amount of boost or cut, which can be applied to that frequency band
Thus a true parametric equalizer will have three knobs per section; a frequency sweep control, a bandwidth control, and a boost/cut control with "flat" being the 12:00 setting. With parametric EQ, you can work with surgical precision to correct or enhance very specific tonal problem areas. By gaining precise control over the area of the audio spectrum you need to shape, you can use just as much EQ as you need while minimally affecting neighboring frequencies.

C) TONAL ENHANCEMENT WITH EQ

The application of EQ is more of an art than a science, and one of the cardinal rules in audio is to do as little as possible to achieve your desired results. It is very easy to find yourself getting further and further away from your intended goals the more you twist knobs. With the bypass switches on both the parametric EQ and notch filter stages of Equinox you can quickly compare your knob settings to the straight-through sound to see if you are moving in the right direction with your EQ settings.

One of the uses for parametric EQ is to tame boomy, strident, or otherwise obnoxious tonal characteristics from various instruments by using the Gain control. One recording engineer trick to finding the right frequency to cut is to go in the opposite direction; turn the EQ control to full boost and sweep through the frequency control with the Bandwidth set fairly narrow. When the sound gets ultra-obnoxious, you know you've found the right frequency band to cut. You can then work with the Bandwidth control to hear how wide a range of frequencies should be cut to achieve the sound you're after.

D) USING EQUINOX AS A DISTORTION ENHANCER

One of the more unusual uses for parametric EQ is as a "pre-preamp" between an electric guitar and the front end of a tube amplifier. By turning up the boost knob, experimenting with the bandwidth control and sweeping the frequency knob through on the lowest band of EQ, you can "voice" the overload distortion characteristics of the amplifier yielding a rainbow of great tones. Additional use of the other two bands can further enhance this voicing or pre-emphasis of the frequency spectrum you are sending to the amp.

GLOSSARY OF SOME AMPLIFICATION TERMS

AMPLIFIER
A device for making small electrical signals bigger. The amplifier was actually invented in the late 1800's before there were any devices that could make building one possible! In music, the term often refers to a self contained "combo amp"- an electromechanical device combining a preamp, amplifier, and loudspeaker usually including some kinds of tone shaping circuitry. With Solstice, you have the option of "custom building" your amplifier by matching the preamp/mixer/blender stage to a separate power amp and loudspeaker system.

BUFFER
A preamplifier designed to isolate the source from the next stage of amplification. Buffer amps have high input impedances and low output impedances and can also feature some "gain" or signal boosting capability. Buffers are required with piezo crystal or piezo polymer pickups, and are often built into acoustic-electric guitars. Solstice features high impedance buffer stages for both channels so you can use either active or passive pickup systems. D-TAR makes several on-board buffered pickup systems for acoustic instruments.

CARDIOID MIC
A microphone designed to be more sensitive in one direction than in others, a directional mic. Cardioid mics are used more often than other types on stage because they make it easier to isolate one voice or instrument from the others for mixing. Hypercardioid mics, sometimes called shotgun mics, are designed for use at a distance as they can be aimed at the sound source and used from many feet away. The good old Shure SM-57 and 58 are cardioid mics.

CHORUS
An electronic device that can split a signal, mildly shifting the pitch and timing of one part, then mix it back in with the original signal. The effect is roughly like several people (they're the chorus) playing the same part at the same time. Solstice has effects loops to allow convenient interface with chorus effects.

COMPRESSOR
A processor that "squeezes" the dynamic range of the signal by limiting peaks and bringing up the level of soft passages. A limiter can be used to fatten a sound or give it more apparent sustain. If you've ever wondered why music sounds kind of flat on FM radio compared to live, overuse of compression can be one reason. On the other hand, the Beatles used tons of compression on their acoustic guitar recordings and it sounds great. Can be used in Solstice effects loops.

CONDENSER MIC
A microphone in which an electrically charged diaphragm moves with sound waves while a charged back plate stays stationary. Because the diaphragms of condenser mics can be made very light in weight, the frequency response can be very good with a condenser mic. Neumann mics, considered by many to be the ultimate mics for recording voice and acoustic instruments are condenser mics."Condenser" is an old term for capacitor. Condenser mic derive their name from the fact that they are sensing the change in capacitance between the diaphragm and the backplate and converting it to a signal voltage.

CONTACT PICKUP
Sometimes called "soundboard transducers" are most often piezoelectric accelerometers (acceleration monitors). They put out an electrical signal that is an electrical equivalent to the mechanical vibrations occurring where they are placed. The D-TAR Perfect Timbre is an excellent example of this type of pickup.

dB
Yes, lower case "d", upper case "B". The "d" stands for "deci"; the "B" standing for Alexander Graham Bell, inventor of the telephone, teacher of the deaf, and co-conspirator with Glenn Curtis in attempts to steal the secrets of controlled flight from the Wright Brothers (see the movie *Winds of Kittyhawk*). A decibel is a unit used for comparing ratios of signal strengths for acoustical loudness or for electrical audio signals. An expression of dB must be referenced to some other level, as it is not an absolute entity, but rather the logarithmic ratio between one signal strength and another.

0 dB SPL
Is the level at which half of the population of humans with unharmed ears (no Blue Cheer or Barry Manilow music…) can perceive a sound. The "SPL" stands for "sound pressure level."

0 dBV
Is a voltage reference equal to 1.0 Volts rms.

0 dBu
Is a voltage reference of .775 Volts rms, the "u" meaning unterminated, or going into an infinite impedance load. This is the standard used throughout the recording industry.

+4 dBu
Is the standard "pro" audio voltage reference level which is equal to 1.23 Volts rms.

-10 dBV
Is the standard reference level for consumer and some home studio gear. Often found with gear using RCA connectors.

DI (see also "DIRECT BOX")
The British term now common in the US for "directly interfacing" a pickup signal into a recording or PA console, thus bypassing amplifiers, speakers, and mics. Used especially for electric bass to get a clear tone. Many home enthusiasts directly connect their acoustic guitars to the recording device to gain better isolation from track to track than they can get just using microphones.

DIAPHRAGM
In a microphone, a thin, stretched, plastic film, the equivalent of your eardrum. The diaphragm vibrates with sound, then transforms that acoustical energy into an electrical signal that can be amplified.

DIGITAL DELAY (DDL)
A signal processor that converts analog signals into a stream of digital information that can be delayed and mixed to give echo-like sounds. Digital delay is usually included with other sonic colorings in multi-effects processors.

DIRECT BOX
A device used to buffer or isolate guitar and bass signals so they can be "DI'd". Many of the direct boxes designed for electric guitars and basses do not have a sufficiently high input impedance for interface with piezo pickups. Direct boxes can either be passive, using transformers, or active, using tube or transistor based circuitry.

DYNAMIC MIC
A microphone which works like a backwards loudspeaker. The diaphragm is attached to a small coil of very fine wire that is surrounded by a magnetic field. When the diaphragm and coil vibrate with sound waves, a small electrical signal is generated in the coil that can be amplified through a mic preamp and other devices. The Shure SM-57 & 58, two of the most common mics used in clubs and studios, are dynamic mics. Dynamic mics are noted for being tough; the mic you can drive a nail with is probably a dynamic.

EFFECTS LOOP
A set of jacks on an amp or preamp which allow sending a signal out to an effect and bringing the modified sound back to the main unit. The advantage of an effects loop is that it is buffered (yes, same concept) on the output and input, the effect will "see" a predictable impedance and level, and the modified signal can be master volume controlled in the main amp or preamp.

ELECTRET MICROPHONE
Miniature mics that work on condenser mic principles but have permanently charged polymer diaphragms. Electret mics have miniature preamplifiers built in and require low voltage DC power (usually 1.5 to 18 volts) often supplied as "phantom power." The Seymour Duncan Mag Mic uses an electret element for it's second source with "on-board" blending.

EQUALIZATION or EQ
An electronic means of shaping frequency response; the term generally refers to sophisticated tone control circuitry. Originally used to mean correction for the unequal frequency response of old PA, recording and playback gear.

EXTERNAL MIC
Generally referring to the good old practice of standing in front of a mic on stage as opposed to installing a mic in your guitar. You've seen them, you've used them, and now you know what they're called. There are now some bracket devices for mounting an external mic on your guitar.

FEEDBACK
Yowl, howl, etc., feedback by any name is the sonic nemesis of the performer. It happens when amplification goes beyond control, and the amplified sound itself is re-circulating and becoming further amplified. The sonic equivalent of Chernobyl—audio meltdown. "Ringing" is the precursor of feedback and refers to a barely controlled resonance just shy of feedback. D-TAR's Equinox features two notch filters designed to combat feedback.

FLOATING PICKUP
A magnetic pickup mounted to the end of the fingerboard on a guitar or to some other non-vibrating part of a musical instrument. Floating pickups are sometimes used on archtop acoustics so the adding of a pickup will not interfere or change the vibration pattern of the top. Seymour Duncan makes a variety of floating pickups including the Bob Benedetto signature pickup for use with archtop guitars.

GRAPHIC EQUALIZER
An equalizer that uses sliding potentiometers (slide pots) to control the level of the signal in various frequency bands. Called so because the knobs form a graphic representation of the frequency contouring. Graphic equalizers are generally either "1/3rd octave" or "1/10th octave" referring to the width of the audio bands covered.

HUMBUCKING PICKUP
A type of pickup using two coils to cancel magnetically induced hum. Invented by Seth Lover at Gibson in the 1950's, the "humbucker" is noted for it's loud and warm sound. Check the Seymour Duncan website for the world's most complete selection of humbucking pickups.

Hz
Formerly known as "Cycles per Second" and named after Heinrich Hertz (not of the auto rental company…), the scientist who in the late 1800s was a pioneer in producing and detecting electromagnetic radio waves.

IMPEDANCE
A measurement of the resistance to the flow of AC (which is what audio signals are); impedance is affected by resistance, capacitance, and inductance in a circuit and is also frequency dependent. Impedance is often mistaken for resistance and is also incorrectly thought of as being a measurement of the voltage from a pickup. In practical terms, you want low impedance sources feeding into high impedance loads; this gives maximum accuracy in signal transfer.

INTERNAL MIC
A microphone, generally an electret condenser mic, mounted inside an instrument.

LIMITER
A limiter keeps hot signals from overloading the next stage of electronics. Les Paul takes credit for inventing the limiters as used in recording studios. He related that he got the idea from watching Mary Ford turn her head while singing loud passages as she watched the recording VU meters. She physically limited the input signal to the mic with this technique.

LINE-LEVEL
The voltage level at which most pro gear sends pre-amplified signals to other devices such as equalizers, limiters, compressors and power amplifiers. Generally considered to be +4(dBu) or 1.2 Volts RMS.

MAGNETIC PICKUP
A pickup that consists of a magnetic structure and one or more coils of very fine wire which "transduce" or transform the vibration of plain steel or steel cored wound strings into an electrical signal.

MIDI
Musical Instrument Digital Interface, the computer language used in modern synthesizers and signal processors to "communicate" with other devices.

MINI-MIC
Miniature mics derived from hearing aid and CIA "mic in the martini olive" technology. These are generally electret mics, a simpler variation on the condenser mic.

MIXER
Used to combine or mix multiple sound signals into a mono, stereo, or other simpler signal to go onto tape, a CD, or through a PA system. Also refers to the person who does the mixing, not to be confused with re-mixer, the person who doesn't mix live, but works on mix-downs of pre-recorded mix-ups.

MONITOR
Generally referring to a set of speakers aimed at musicians used to give performers a chance at hearing themselves on stage. Watch for "In-Ear" monitors, the latest thing in stage monitoring; these are like hearing aids for musicians. The term "Monitor" implies accuracy as well, as in "Studio monitor speaker."

"NATURAL" SOUND
Often achieved with the most unnatural of means, natural sound is the Holy Grail of most acoustic musicians. To hear it, try listening to truly acoustic music. Our goal at D-TAR is to help you achieve the most natural sound you can get … plugged in.

NOTCH FILTER
A specialized kind of equalizer that can be tuned to "notch out" problem frequencies without affecting neighboring frequency bands. Usually used to kill feedback frequencies. Equinox features two switchable notch filters.

OMNIDIRECTONAL MICROPHONE
A mic that picks up sound more or less equally in a spherical pattern all around the mic's diaphragm.

ONBOARD and OUTBOARD
Generally refers to where pickup buffering and/or EQ stages are located. Onboard in your instrument, outboard is somewhere else, man.

PA SYSTEM
Originally "Public Address" system. Do you remember, "Would Johnny Brown please come immediately to the principal's office?" Some of the first PA systems were used in department stores and schools. Now the term refers to sound systems designed for amplifying live music.

PARAMETRIC EQ
A type of equalizer that allows continuous control over three parameters: frequency, bandwidth, and amount of boost or cut. While a bit harder to intuitively understand than graphic equalizers, parametric EQ is preferred by pro audio engineers for fixing specific sonic problems without affecting other frequencies as happens often with graphic EQ. D-TAR's Equinox is a three band parametric EQ with two bands of notch filtration.

PHANTOM POWER
A system in which DC current is run up the same cable used to send the signal down to a mixer. Used most often in the studio to power high end condenser mics, but sometimes used for powering mics and other electronics inside guitars.

PHASING
The relative polarity of two or more signals that contain similar information. In-phase signals add together while out-of-phase signals tend to cancel.

Q
In more musical terms, this refers to the "Bandwidth." At its narrowest setting, you can come close to being able to boost or cut just a single note. Set wider, you can affect a full octave-wide set of frequencies. At a Q of approximately 1.3, the bandwidth is 1 octave. As Q goes up, the bandwidth gets narrower, so at a Q of 2.6, the bandwidth is 1/2 octave. Conversely, at a Q of 0.65, the bandwidth will be 2 octaves. The "bandwidth" or "Q" control is one of the features that make parametric EQ so versatile.

TRANSDUCER
Any device that changes mechanical or acoustic energy into an electrical signal or vice versa. Mics, pickups, and loudspeakers are all transducers. The term transducer is often used with accelerometer style piezo pickups, but is not exclusive to such pickups.

TRANSIENT RESPONSE
The quality of how fast a preamp, amplifier, or signal processor responds to an input signal. Related to "slew rate". Fast is good, slow is bad.

TUBE
An electrical device that can amplify low-level signals into higher equivalents. Tubes are the oldest technology for this purpose and are still preferred by many in preamps, direct boxes, and amplifiers. They're made of glass like lightbulbs, and boy, do they get hot!

TWEETER
A loudspeaker designed specifically for high frequencies. Tweeters usually cover the range from 3000 or 4000 cycles (3 to 4 Kilo Hertz) on up to 20 K Hz. Think "Tweetie Bird."

WOOFER
A loudspeaker designed for reproduction of low frequencies, generally from 20 Hz to 1 to 3 K Hz. "Midrange" drivers are used sometimes to cover the frequencies between 1 K Hz and 4 K Hz.

Limited Warranty

D-TAR offers the original purchaser a one-year limited warranty on both labor and materials starting from the day this product is purchased from an Authorized D-TAR Dealer. D-TAR will repair or replace this product, at its option, if it fails due to faulty workmanship or materials during this period. Defective products should be returned to your USA dealer, international distributor, or sent direct to our factory postage prepaid along with dated proof of purchase (e.g., original store receipt) and a RMA number clearly written on the outside of the box. Please call our factory at 805-964-9610 for an RMA number.

This warranty does not apply to damage to this product or an instrument caused by misuse, mishandling, accident, abuse, or alteration. Product appearance and normal wear and tear (worn paint, scratches, etc.) are not covered by this warranty. D-TAR reserves the right to be the sole arbiter as to the misuse or abuse of this product. D-TAR assumes no liability for any incidental or consequential damages, which may result from the failure of this product. Any warranties implied in fact or by law are limited to the duration of this express limited warranty.

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5427 hollister avenue
santa barbara ca 93111.2345
PN#501055-120 Rev. B

duncan|turner acoustic research

5427 hollister ave santa barbara ca 93111.2345
tel 805.964.9610 fax 805.964.9749 www.dtar.com

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