VINKTRONIC LABS ART PRO MPA SUPER TUBE PREAMP
V 1.1
USER MANUAL

Thank you for purchasing the Vinktronic Labs ART Pro MPA Tube Microphone Preamplifier. I have been studying tube design and theory for 25 years. This piece is the result of many years spent modifying, altering, studying and tweaking different pieces of new and vintage tube equipment. I have designed this piece to be user adjustable when creating the subtle harmonic distortion characteristics that many listeners and recordists find desirable. It emulates many of the different vintage tube processors of the early days of music recording. This machine however, allows you to adjust the harmonic distortion, dynamic equalization and non linear gain limiting by use of 2 stages of tube preamp redesigned with a unique circuit using constantly variable controls to carefully tweak the sound character. It has also has improved signal to noise ratio that modern recording studios require for microphone or line level preamping (the unit has been converted to be used at line level for audiophiles and final mixing in a studio). Please take a minute to read through this manual before using the preamp so you can get the best use out of this piece. I will also explain a little theory of harmonic enhancement and dynamic equalization as it applies to affecting the tonal quality of musical signals when the total audio volume is changing. As an overview, I will admit the preamp has been "voiced" for semi-professional home recording studios, where the musician is using a stereo pair of quality microphones, small or large diaphragm condensers with wide bandwidth and expecting to capture a very "analogue" sound field. The device can also be used as a single channel preamp, but it's capability in offering 2 well matched channels of vintage character can create a remarkable stereo recording. The preamp is not flat in frequency response or super low distortion. It is attempting to mimic a vintage sound the ear likes to hear, but also creating a slight characteristic of dynamic sound equalization for "mechanical transducers" when capturing the sound waves. As an anecdotal point, microphone capsules shouldn't necessarily be round, or square, but perhaps oval-like in shape to accommodate hearing patterns and to reduce foil plate resonances within the diaphragm itself. Such a microphone would probably have a slight natural equalization of audio which accommodated the fact that it is a mechanical transducer. Specifically, good microphones should have no narrow "pass band" resonances (peaks in the response), but in many cases (especially near field micing) it is acceptable for the mic to have slight bass or presence enhancement. Infact many microphones do, and the equalization/harmonic switches attempt to accommodate many microphones.

1
GETTING STARTED

1 - Plug in the microphone. It will be positioned when everything set up to be at a distance from the instruments of which the gain and level controls are somewhat at ½ way to ¾ way up to produce a professional 0 dbm signal level out, as shown by the MPA

meter. Be aware the output signal level of the MPA may be too high the expected signal level input of many semi pro processor devices such as reverbs or equalizers. Their input levels may require to be turned down. I have not encountered a situation yet where real pro equipment cannot take the true 0dbm level output with peaks of dynamics at +14db (the unit intentionally limits more excessive signals above +6db mostly though only on complex signal or severe transients).
2 - Let the unit warm up at least 5 -10 minutes.
3 - Switch on the 48 volt phantom if required by the microphone. Also 5 minute warm-up.
4 - Microphones can be placed a little further back than typical as the preamp has a different way it "preamplifies" and slightly gain rides the signal with harmonic and equalization enhancement. Close micing is acceptable though as there is plenty of dynamic headroom available. In my preamp circuit it is virtually impossible to overload the gain stages unless the LED or meters are totally continually maxed out. This situation will nonetheless create a usable soft overdriven tube sound which some original owners of prototypes liked for certain effects, such as stressing a sense of sonic "urgency" on vocals or bass, so this effect is left in but lessened on this newer "audiophile" version. In general however omni-directional mics work better for close field micing (6" to 3') picking up sound all around the instrument and cardiods work better 2 to 6 feet away, as they focus on musical sounds which are mostly in front of the mic capsule.

2
THE INPUT CONTROL

1 - Set the INPUT GAIN as high as possible to affect the tube LED's to run near the red when playing an instrument at the loudest volume.

a- When the LED's are running in the red, you will get more tube saturation/ "analogue tape" character, as the tube circuit now truly allows a soft symmetrical tube limiting of strong dynamics and peaks. Very few tube circuits actually do this. Most designs clip somewhat asymmetrically and impart a sonic fussy grit to the signal. With my unit, more desirable 2^{nd} and less 3^{rd} harmonics are created, but this is gain limited so as not to cause an obvious fuzzy tube distortion when excessive signal is present. Again I feel it is better to create a little soft limiting of dynamics rather than soon clip the audio at a higher level.

b- When the LED's are running under the red, the circuit will produce more 2^{nd} harmonic presence which is adjustable in phase and equalization by the tube cathode bias control (adjustment will be explained later).

c- When the LED's are running low (in the green/orange area), it will give a more clean sound with virtually no 3^{rd} harmonics added to the music. In general, the distortion now is mostly predominated by 2^{nd} and 4^{th} harmonics, but as will be discussed later, this is adjustable as in phase or out of phase harmonics of the original audio signal. These preferred harmonics at lower level are gain ridden up to provide a more obvious tube "thickness" and "presence" depending upon use of cathode bias and equalizer controls. This is the subtle but important sonic effect which can be heard by listeners as the "tubes are warm yet detailed sound" but now at low signal levels far away from the overdriven tube effect creating a sonic texture very much more euphonic on a wide range of recording situations. This is a crucial design criteria which sets my unit apart from most modern units. Some vintage preamps and compressors were able to achieve this but did not have enough adjustment versatility. As an anecdotal point, I've encountered too many modern designs which are inconsistent to deliver this effect. The same model will seldom create this effect consistently or at all (especially with aging use), hence the wide range of users opinion of what they hear. This is due to subtle differences in aging circuits, power supplies and especially tube selection as it applies to a particular circuit design. It is important to note here and break the myth that certain name tubes are better or worse, but really from an engineering standpoint it is the actual circuit design which will determine the tubes performance to create the euphonic second harmonic trick. There are good and bad tubes past and present. People should not be so compulsive about a brand name, especially now that some modern 12AX7 are so good. My circuit is designed to be capable to adapt a part or two and then accommodate a wide range of tubes if necessary, more manufacturers should do this. From a personal perspective, I've seen countless expensive tube designs which were running really poor but the user thought it was great, until I fixed it correctly and then was told, "Wow I didn't know it was supposed to sound this (better) way!".

3
THE OUTPUT CONTROL

1 - Set the output control so the VU meters hit or go above the 0db mark.
2 - When the OUTPUT CONTROL is set higher and the meter is reading into the +0 to +3 region (it is OK to pin the meters occasionally) a slight volume leveling will start to occur. This gives instruments a fuller, more forward sound, although slightly compressed, the deep bass impulses and high frequency transient peaks will not be reduced as much or lose clarity. More high frequency harmonics are added and the middle range will be more controlled, slightly veiled with presence. This is not a simple compressor. It is a subtle logarithmic level reduction of volume of sonics, as well a slight gain riding of lower level signals keeping the "tube" sonic effect present. It also does not compress the lows and highs as much as the midrange, trying to approximate the "loudness" equalization that a human ear/brain perceives. The meter circuit electronics and needle movement calibration are part of the tube harmonic gain riding effect and actually forcing the tube to maintain 2^{nd} harmonics from the tube at -20db and less at 0db level. This is the most important trick my MPA does. Again it is important to note my design forces the tube to create a truly pleasant tube 2^{nd} harmonic distortion at lower levels (-20db) as opposed to most circuits whereby the tube must be driven to near clipping to get a tube "stressed" character. Use of the cathode bias control forces the tube to create in phase harmonics and warming up the sound character or adjusting out of phase harmonics from the tube will affect a slight presence or "sparkle" to the detail of music. No need to overdrive the tube. More control to enhance low level signals which create a bigger, richer sonic image to the ear.

4
THE +20db GAIN SWITCH:

When this switch is pushed in it will boost the overall gain throughout the unit, giving a more forward, thicker powerful sound. When the switch is out, a gain matching and equalization circuit is turned on. This will give the impression that the mics have been moved farther away from the instruments and the gain has been automatically turned down. This circuit also attempts to phase correct and equalize the mic signal tonal quality to be more distant but still big sounding. Admittedly this is my perception of what sounds good to me, when using wide bandwidth microphones placed in an intimate (2 to 6 ft. mic position) set up or remixing an audio track which seems too forward. The sonic picture will be more laid back. An anecdotal point here to make is that many older mics (well respected and much recommended but not necessarily the most honest sounding) will not have the true wide bandwidth which I feel is important to create a natural soundscape. Many of these older mics were design for close field recording and even the most famous names do not have bass bandwidth down to 30-60 cycles or past 12-14 kHz. Sorry to say many of the new generation of an Asian made mics are remarkable in dynamic range and audio bandwidth. When older mics are placed close the bass predominance effect and treble clarity will be more apparent. This was part of their design criteria. They can take high signal levels and the mic elements have clarity a high SPL. Where as the more modern efficient mics with wider band width and dynamic range appear to blur up a little at close micing but sound incredibly clean at a little more distance. Actually some of this mic distortion is really the mic preamps within the mic itself slew limiting or clipping. I admit I prefer to get the microphone a little further away now that the mic can "hear" more completely the whole instrument. I have tried to allow controls to adjust for either use. This is a sonically dryer and thicker sound and will sound excellent when processed through a good reverb. Move the input or output controls to make up gain if necessary.

5
HARMONICS SWITCH:

The Harmonics equalization switch is a 3-position toggle switch. Each position affects the general tonal balance and somewhat the tube compression of sound. Each position will alter the compressor tracking and affect a slight timbre change of the music.

- a- The up position is good for close micing with older warm sounding mics. It gives the impression of the mics being moved back but keeping brightness or detail. If the mic or instrument is too fat this will balance out the tone quality and lessen mic predominance effects of reducing the bass, accenting the high frequency content. Especially use full equalization for ribbon microphones.
- b- The middle position is good for lead instruments or vocals. The harmonic equalization circuit keeps the sonic information strong and forward and the most natural setting, flat frequency response, less gain riding of high and low sonics. This is probably the best setting for ensemble instrument recording or remix of stereo program (I use an MPA on my CD player).
- c- The down position is good for near field mics which are too noisy or have too much presence (close micing of piano, acoustic guitar and other acoustic

instruments). It allows the low and midrange harmonics to dominate but still keeps detail in the high frequencies. This is the “warmer” setting of the preamp. Very useful on microphoning with bright mics or music which is too edgy or strident. It is also my opinion of the warm tube sound and works best when recording in bright rooms or instruments which are naturally too bright. Drums, reed instruments, voice and 12 string guitar can now sound very rich in harmonics with out the “edgy” or strident quality.

6
TUBE BIAS EQ CONTROL:

This control affects the bias and gain of the tubes and creates the harmonic distortion being introduced into the signal path by adjusting the DC cathode bias and gain of the two stages of tubes.

a- Start with the bias control in the middle position. This gives a fairly neutral tube character to the sound, only a slight amount of tube character with typical gain a tube would want. Probably preferable when complex or harmonically already rich musical information is passing through.

b- Turning the control to the right (clockwise) will add more in phase 2nd harmonics. This will give a velvety, slightly veiled sound character ideal for use on bright mics, guitars, pianos and vocals. Creating about 2% 2nd harmonic distortion with the control at the maximum setting. Sometimes perceived as a “dryer” sound.

a- Turning the control to the left (counter-clockwise) will add more out of phase 2nd harmonics. This will thicken the bass and midrange with more “edge” to the sound character or presence to guitars and voice. This effect will be subtle, the more complex the overtones the more obvious the effect will be. The control will create up to 2% harmonics at extremes of knob position. No other device I know of allows the user to vary the strength of harmonic enhancement, to be in phase or out of phase, with attendant equalization, and a subtle gain riding of those harmonics. Remember too the harmonics are naturally part of and created only from the original musical signal passing through the circuit. If the signal level becomes louder, the ratio of harmonic in my circuit design will stay about the same, as opposed to other designs where louder signals get the tube to sound “dirtier” or be missing at low signal levels.

Do not be afraid to experiment with different settings. All the controls on the front of the faceplate are always live. Try different combinations of control settings. Combinations of mics, placement and instrument will create a myriad of different sonic characters.

Example: The harmonics switch in the up position and the bias control turned to the left (clockwise) will give an “edgy” or sparkling sound. Very desirable on voice and acoustic guitar, especially to push them forward in the mix.

Do not be afraid to run the meters hot. At +1db/+3db meter swing is typical of peak signals. Again the unit will self limit to reduce overloading of the next device (mixer, reverb) connected to the MPA output. The logarithmic compressor, dynamic equalizer is designed to be only slightly obvious, but admittedly I have “voiced” it to work well with large plate diaphragm mics at midfield placement. 1 to 3 feet away from guitars voice, maybe 3 to 6 feet away from piano or drums. The compressor is similar in some ways to an RCA BA6 (Teletronics) design and attempts to slew limit the audio signal so as not to cause digital conversion distortion (slew rate limiting) problems. All digital converters regardless of price or manufacturers claims slew limit above 4kHz. 96kHz sample rate improves the slew rate whereby slew limiting starts to occur at 8kHz to 12kHz. Oddly most speakers or people truly cannot hear this problem but reverbs algorithms suffer obviously, if you listen very carefully.

Another unique feature on the modified MPA is the variable predominance filter. This circuit has been altered to be more “phase correct” and truly filters bass dominating the sound when using mics placed close to instruments. My circuit maintains a more phase correct bass even when attenuating, dipping out the “bass muddiness”, and less extreme deep bass, and pluck is also not lost. To be honest, I’m surprised more designers have not figured out this trick. It’s effect is very obvious to a trained ear with true wide band speaker monitors.

Since this unit is voiced to be used with a good large plate condenser microphone which have a wide dynamic range. The LED meter can often run into the red for most recording purposes. Constantly red LED meter will affect the tube to symmetrically soft clip causing an aggressive “overloading of tape” sonic character or overdriven guitar amp effect when totally red. Be careful this overdrive tube character is best used only on individual instruments or simple instrumental passages. Although original engineering samples of the modified MPA were used by hard rock and roll types who liked this subtle tube “grunge”, especially as a final mix effect. I left it’s capability in but now gain limit a little more for the audiophile recordist.

Also, I will admit there is a design of the tube circuit to intentionally “plump” the sound. It’s a slight tube “kick” (really a DC bias shift) which is inherent even in well done designs. This sound effect is for some people one of the most desirable effects, either you like it or you don’t. By keeping the meters and LED’s out of the red it will be little noticed.

The design attempts to gain ride just enough so as not to allow slew limiting affects to occur in digital converter which will probably be in the next device, reverb, eq, computer recording etc. This is a subject which will need more space to explain clearly, but simply put digital has great signal to noise but no dynamic head room at all. Most sample rates will still create slew limiting problems in the audio typically audible to the trained ear a slight stridency of high frequencies. This may not be too noticeable on most speaker monitors presently used. (Most tweeters have such poor high frequency response and horrible dynamic linearity that this speaker effect actually dominates, but I am predicting a truly great improvement in the next generations of studio monitors. Another anecdotal point, after 20 years of actually measuring so many "professional speakers" - I will tell you honestly I've seen very few tweeters which could do more than 30 to 50 db worth of dynamic range and get frequency response out to 20 kHz. Think about how often a true test report graph of such parameters presented to the purchaser of speakers. A good look at most of these "pro speaker" show them to be cheap Asian parts, or just simple archaic designs with fancy cabinets and trim on them. Sorry, I'm not impressed. However, high quality headphones can be far more revealing. The best ones are phase coherent and very linear in frequency response, but unfortunately subject to ear/phone cup coupling problems.

In closing I will be clear that this is not the perfect preamp. It wasn't intended to be. It is a great preamp with some novel user adjustable function which allows it so sound like many fancy priced items. It does have an excellent mic preamp front end which can adapt to any mic impedance, high or low, by use of well done multi parallel transistor input true low impedance stage with 3 stages of RFI filter (modified) to keep out as best possible all that crazy radio noise (kudos to ART). It also uses the tube for the best advantage of harmonic character (too bad they didn't use my idea). The output stage can drive any combination of balanced, faked balance line, reverse balanced line with any problem. ART did it right. Some very fancy units will blow the output amp if not in only 1 way. My metering attempts to truly show the dynamics and volume level of the musical sonics. This device can be used as a final mix processor, by use of the phone plug ins. This is my "swan song" for the music business, my friends, and for my own music. I give much credit to ART who supplied me willingly with schematics, parts, some advice and especially for designing a very clever preamp which was just the right thing for me to rev up. The quality of parts and honesty to put a real design into a box instead of an "op amp in a box" with a pretty face plate and marketing hype was refreshing. Most sound engineers will never really "get it". A fancy name will attract more attention, but some musicians can be slowly persuaded once they try a new thing.