THE MANLEY TNT MICROPHONE PREAMPLIFIER

OWNER'S MANUAL

Manley Laboratories, Inc.
13880 Magnolia Ave.
Chino, California 91710

Tel: +1 (909) 627-4256
Fax: +1 (909) 628-2482
www.manley.com

MANLEY® LABORATORIES, INC.

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INTRODUCTION

THANK YOU !

For choosing the Manley TNT Microphone Preamplifier. The name of this mic preamplifier "TNT" is a reference to the two different channels, one that is Tube and the other No Tube. We see people today hungry for a variety of preamps as a means of getting a variety of sounds and tones from the start. Of course, mic selection and mic position can produce a wider variety of colors, but once that is locked in, the preamp choice can be a significant factor in some sessions and that subtle difference in others.

The designer considers the reference to TNT (the explosive) might be a bit apropos too. Both share a reputation as a relatively powerful and particularly useful tool for the 'difficult' tasks while being relatively safe (well, much safer and more stable than Nitro-Glycerine, its main component). Manley would like to thank Steve Pogact for suggesting the name.

Most mic preamps are stereo, which is fine when one needs to record true stereo with matched mics (few are), but the most common situations where one can compare and choose a particular preamp, are the often the more relaxed overdubs. These are typically single miked or multi-miked where each mic might be a different type and different distance from the source. So the TNT might be the ideal cost-effective solution for these situations – maximum choices & minimum cost and rack space and where you need to know you are using the one of the best.

Some people use the TNT simply because it offers two different sounds and is different from their other preamps. Some people choose it because each channel may be a contender for the highest caliber preamp in either the "tube category" or "solid-state category" and they demand the best. And some users just like the unique controls and features. Of course, given a variety of sounds and features, different people will gravitate to one side or the other and one feature or another. This is to be expected and probably encouraged, however let us emphasize that we encourage you to dig in, use it to its fullest and form you own opinions and methods that work for you.

GENERAL NOTES

LOCATION & VENTILATION

The Manley TNT must be installed in a stable location with ample ventilation. It is recommended, if this unit is rack mounted, that you allow enough clearance on the top of the unit such that a constant flow of air can move through the ventilation holes. Airflow is primarily through the back panel vents and out through the top.

You should also not mount the TNT where there is likely to be strong magnetic fields such as directly over or under power amplifiers or large power consuming devices. The other gear's fuse values tend to give a hint of whether it draws major power and is likely to create a bigger magnetic field. Magnetic fields might cause a hum in the preamp and occasionally you may need to experiment with placement in the rack to eliminate the hum. In most situations it should be quiet and trouble free.

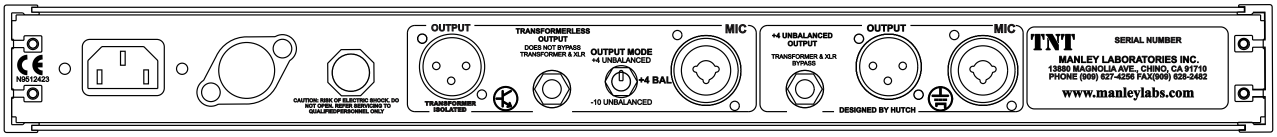
WATER & MOISTURE

As with any electrical equipment, this equipment should not be used near water or moisture.

SERVICING

The user should not attempt to service this unit beyond that described in the owner's manual. Refer all servicing to your dealer or Manley Laboratories. The factory technicians are available for questions by phone (909) 627-4256 ext. 325 or via our website at www.manley.com. Fill in your warranty card! Check the manual - your question is probably anticipated within these pages......

THE BACK PANEL



1 3 8 9 10 7 6 5 4 2

First connect all the cables, then turn on the power, wait 30 seconds, then have fun, as if we had to tell you....

1) **POWER CONNECTOR**. First verify the POWER SWITCH on the front panel is off (CCW). Use the power cable supplied with your TNT. One end goes here and the other end goes to the wall outlet. You know all this.

2) **VOLTAGE LABEL (ON SERIAL STICKER)**. Just check that it indicates the same voltage as is normal in your country. It should be. If it says 120V and your country is 220V, then call your dealer up. If it says 120V and you expect 110 it should work fine.

3) **FUSE**. Unplug the power cable first. The Fuse Cap needs a push then turn a quarter twist CCW to pull off. Fuses are meant to "blow" when an electrical problem occurs and is essentially a safety device to prevent fires, shocks and big repair bills. Only replace it if it has "blown" and only with the same value and type (500mA slow-blow for 120V, 250mA slow-blow for 220V). A blown fuse either looks blackened internally or the little wire inside looks broken. A blown fuse will prevent all the LEDS from lighting and will prevent any power and audio.

4) **TUBE CHANNEL XLR JACK INPUT**. (Left) Accepts balanced or unbalanced sources. The pin out is as follows: PIN 2 = Positive = Hot, PIN 3 = Negative = Low or ground, PIN 1 = Chassis Ground. Be sure that the PIN 3 is connected to the negative or ground and not "open" or a 6dB loss or loss of signal will happen. This input is transformer coupled. Switching PHANTOM POWER on, puts approximately 48 volts via 6.8k resistors (double regulated 46-47 vdc typical) on pins 2&3.

5) **TUBE CHANNEL XLR JACK OUTPUT**. (Channel One or Left) Transformer Balanced and Floating. Only for +4dBu pro levels. The pin out is as above. NOTE: Inserting a 1/4" plug into the "UNBALANCED OUTPUT" jack disables this XLR output !

6) **TUBE CHANNEL PHONE JACK OUTPUT**. (Channel One or Left) Unbalanced output only. Factory set-up for +4dBu pro levels and mono or 2 conductor plugs. NOTE: Inserting a 1/4" plug into this jack disables the XLR output because it switches the transformer out of the circuit to prevent any loading. This 1/4" output should be used if the goal is to record as clean and tight a low end as possible from tubes. This jack also should be considered if the destination or cables present a difficult load for the XLR transformer output (high capacitance from long cables, or low resistance from 600 ohm vintage gear or more transformers).

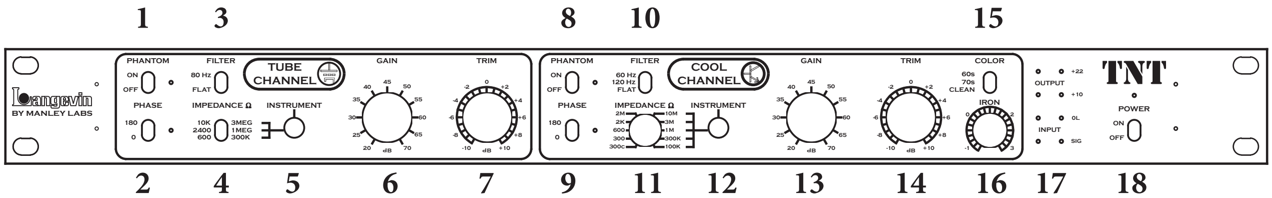
7) **COOL CHANNEL XLR JACK INPUT**. (Left) Accepts balanced or unbalanced sources. The pin out is as follows: PIN 2 = Positive = Hot, PIN 3 = Negative = Low or ground, PIN 1 = Chassis Ground. Be sure that the PIN 3 is connected to the negative or ground and not "open" or a 6dB loss or loss of signal will happen. This input is transformer coupled. Switching PHANTOM POWER on, puts approximately 48 volts via 6.8k resistors (double regulated 46-47 vdc typical) on pins 2&3.

8) **COOL CHANNEL XLR JACK OUTPUT**. (Channel One or Left) Transformer Balanced and Floating. The pin out is as above. The 1/4" jack has no effect on this XLR (both can be used at once) but the OUTPUT MODE switch should be properly set (generally +4 Balanced for maximum headroom) however it is worth mentioning that a transformer floating output is equally happy driving balanced or unbalanced inputs. But we will remind you that most unbalanced inputs are designed for -10 levels so in order for TNT's levels and LEDs to work as designed, if the destination is -10 consumer level the switch should be set for "-10 unbalanced" (usually ;>)

9) **COOL CHANNEL PHONE JACK OUTPUT**. (Channel One or Left) Balanced or Unbalanced output so a 2way or 3way(stereo) plug can be used. Factory set-up for +4dBu pro levels. This 1/4" output should be used if the goal is to record as clean and tight a low end as possible (transformerless). This jack also should be considered if the destination or cables present a difficult load for the XLR transformer output (high capacitance from long cables, or low resistance from 600 ohm vintage gear or more transformers). The OUTPUT MODE switch should be properly set (depending on the destination and wires) because these outputs do not use cross-coupled op-amps and are designed to properly drive either balanced or unbalanced systems providing the switch is properly set.

10) **OUTPUT MODE SWITCH.** Sets both the output level and whether both the XLR and 1/4" jack are balanced or unbalanced. Normally this switch should be in the **"+4 BAL"** or Center position. Some PRO gear may prefer the "+4 unbalanced" position but this style of input is becoming increasingly rare, however because the TNT loses 6 dB of headroom in this mode, it can be used for creative "drive" cleverness. Similarly the "-10 UNBALANCED" mode loses another 6 dB of headroom when 'mismatched' in typical +4 balanced systems. However, this mode is more likely to find service because a fair amount of semi-pro and consumer gear with RCA jack inputs are compatible with this level and 1/4" plug to RCA plug adapters are very common.

THE FRONT PANEL



TUBE CHANNEL

1) **PHANTOM POWER:** Toggle up turns 48 volt phantom power on (and the red LED) which is needed for most FET condenser microphones. Turn your monitors down because it may make a big ‘POP’. Avoid patching the mic lines if phantom is turned on. Some ribbon mics can be damaged if phantom is on and cables or patches are changed, so don’t use phantom with ribbons - or be careful.

2) **PHASE SWITCH:** Reverses the polarity (180 degrees) of the microphone signal. Sometimes needed in situations where two mics are used and sometimes useful for vocals when headphones are used. Amber LED is ON when PHASE switch is engaged.

3) **HIGH PASS FILTER:** Toggle up engages a basic 80 Hz filter. Used to remove excess lows, rumble and some air conditioning noise.

4) **IMPEDANCE SWITCH:** 3 position toggle changes the loading characteristics presented to the microphone or instrument. This can subtly change the sound of the mic. “**2400**” ohms is considered normal or typical. “**10K**” is a lighter load and may be appropriate for some sounds and often for ribbon mics. “**600**” ohms may tighten up the lows on some mics. Unlike many Impedance switches, the volume will not change by 6 dB, and will stay relatively constant making comparisons easier. The numbers to the right of the switch indicate the impedance given on the 1/4” Instrument Input, where 1 Meg simulates a typical amp. Higher usually means brighter.

5) **INSTRUMENT INPUT:** Plug your guitar or bass in here. A plug inserted in this jack will disable the XLR Microphone Input.

6) **GAIN SWITCH:** This rotary switch sets the gain for the first amplification stage. The steps range from +20 dB to +70 dB and when used with the TRIM (below) provides a range of +10 to +80 dB of gain. The bottom two LEDs (see #17) indicate “signal present” and “overload” of this first stage. If internal jumpers are properly set, one can turn this switch higher for overdrive and turn down the GAIN TRIM to optimise the level again. If the jumpers are set for “clean” then it may be difficult to overdrive this stage.

7) **GAIN TRIM POT:** This pot is typically used to finely adjust the gain as needed for the recording device or converter. The two top LEDs are associated with this knob, and are intended to help set optimal levels, which are well (about 10dB) below when the TNT clips.

COOL CHANNEL

8) **PHANTOM POWER:** Toggle up turns 48 volt phantom power on (and the red LED) which is needed for most FET condenser microphones. Turn your monitors down because it may make a big ‘POP’. Avoid patching the mic lines if phantom is turned on. Some ribbon mics can be damaged if phantom is on and cables or patches are changed, so don’t use phantom with ribbons - or be careful.

9) **PHASE SWITCH:** Reverses the polarity (180 degrees) of the output signal.

10) **HIGH PASS FILTER:** Toggle middle position is a 120 Hz HP filter, Toggle up engages a less drastic 60 Hz filter.

11) **IMPEDANCE SWITCH:** This is a 5 position rotary switch that both controls the loading on a connected microphone and internally directs and shares the signal between two different preamp circuits or topologies. The least amount of loading is the **2 MEG** setting and the next setting marked **2K** (2000 ohms) might be considered traditional or typical (most preamps were designed for 2000 to 3000 ohms). Both of these two only use the cascode FET preamp. The next two settings share both the cascode FET preamp and the current mode preamp, **600** and **300**. The final setting, **300C**, only uses the current mode preamp and relays bypass the sub-circuit used for mixing the preamps, so it has some purist function.

12) **INSTRUMENT INPUT:** Plug your guitar or bass in here. A plug inserted in this jack will disable the XLR Microphone Input.

13) **GAIN SWITCH:** This rotary switch sets the gain for the first amplification stage. The steps range from +20 dB to +70 dB and when used with the TRIM (below) provides a range of +10 to +80 dB of gain. The bottom two LEDs (see #17) indicate “signal present” and “overload” of this first stage. Use this switch in conjunction with the 60’s-70’s switch and the LEDs to control the amount of color or distortion. In this situation the GAIN SWITCH becomes a “drive” control too.

14) **GAIN TRIM POT:** This pot is typically used to finely adjust the gain as needed for the recording device or converter. The two top LEDs are associated with this knob, and are intended to help set optimal levels, which are well (about 10dB) below when the TNT clips.

15) **COLOR SWITCH:** With the switch in 60’s or 70’s, a circuit is added that is designed to clip in an interesting way that somewhat simulates the way magnetic tape, and guitar amps clip. Use the GAIN SWITCH to control the depth of distortion. In general best results are obtained when the desired effect is subtle and this circuit is just lightly driven and obvious distortion is minimal. Of course, sometimes more drastic effects are desired and the GAIN SWITCH can be turned up. Thicker distortion may take several processors. The 60’s - 70’s switch can alter the highs depending on drive levels.

16) **IRON EFFECT:** This knob adjusts how audible the output transformer may be from exaggerated at +3 to near zero at 0 and even becoming inverse or the opposite of a transformer at -1. This control is essentially “out-of-circuit” with the knob straight up (12:00) (like an EQ). The knob controls several subtle effects including low frequency level, low frequency distortion, high frequency level and high frequency dynamics. This control is often subtle and somewhat signal dependent.

17) **LED LEVEL INDICATORS:** Simple indicators to show signal presence, first stage clipping, and more or less appropriate levels for the next device. See page 000 for more details.

18) **POWER SWITCH:** With this switch UP, LEDs should come on and maybe sounds might come out the back XLRs......

BEGINNINGS

The TNT project began due to requests from fans of the SLAM! and in particular Lynn Fuston of 3D Audio Inc. The request was simply "Can you bring out just the SLAM! Mic preamps without the Limiters and other features. Then came the 3D Audio bulletin board "Dream Pre-Your ideas wanted" thread which had a lot of great ideas and diverse opinions. Perhaps the most obvious theme was that engineers were now using several mic preamps at their disposal for a variety of colors like they had always done with mics. This was actually a pretty new trend in '92. How about a box with a few different preamp topologies for different sounds?

We began to experiment with some simple discrete preamp design concepts and breadboarded a few approaches. Hutch also had developed the "Rapture" gain stage intended for a proposed digital converter that was impressively un-colored and which became the standard against which other 'experiments' were compared.

Then the 3D mic preamp summit "Preamps in Paradise" happened January 2004. This was a historic event in Tennessee with a panel of 10 notable preamp designers, and 7-10 famous engineers known with reputations as preamp connoisseurs. Amongst fascinating stories and a sharing of approaches, this designer was hearing a chorus of requests, a short list that included "variable impedance – but minus the typical gain changes" and "some *new* useful control or knob". Up to then, the solid state preamp we were working on was envisioned as a typical minimalist '2 knob' discrete channel in some ways similar to the tube side borrowed from SLAM!. But the engineers were asking for more control, and were describing approaches based on sonic characteristics such as clean / not so clean, bright / not so bright, transformer or transformerless and this resonated with the designer's session experience. Designers talk about discrete topologies, tube types and transformer details but recording engineers talk about sounds and controls and session techniques.

The initial concept of the TNT was to put two very different mic preamp technologies in the same box, and that each were to be as simple as "plug in a mic and it sounds fine", without a lot of controls to get in the way. In the end, the TNT did get new features and controls such as the IRON knob, 60's - 70's switch and its unique impedance control. And these were largely due to engineer's requests from the "Preamps in Paradise" event.

At the 3D Preamp Summit, the topic of "vintage-style" electronics came up, not because of huge desire from the engineers but more as a designer topic relating the headaches of recreating transformers and obsolete parts accurate enough to be comparable to the original. However where there was interest in old school style was when the recording engineers began to talk about how they did those classic 60's sessions, and the focus was on production technique and war stories rather than components and gear. Maybe the engineers were saturated with recreations of old gear and were craving both the magic of old sessions and yet new exciting toys to do their job with tomorrow. A bit of both.

So we went back to the ol' drawing board and back to the lab bench and experimented with a variety of circuits and topologies, but this time holding truer to the end result the user would appreciate rather than the internal workings that might have buzz word appeal. Already the SLAM! preamp was not "all-tube" but more of a hybrid FET-Tube cascode, so why stick to "all-discrete" or "vintage-clone" when the engineers seemed to be just concerned with sound or tone and occasionally hoped they could get a few new controls if possible.

The TNT solid state side evolved into a mix of discrete and op-amps, plus it ended up with the Rapture Amp for the line driver. Why this mix? Just our decision based on listening comparisons where our choices generally favored the cleaner or most true to the source as a base. The users could always add stuff that gave some color or texture onto that base and we provided a few too.

We did come up with a few controls that gave some possibilities for "tints" or "flavors" that could be dialed in. For the most part these were designed to be subtle rather than drastic because the TNT is a mic preamp and not an EQ or a typical processor. In fact even the IMPEDANCE switches were designed to not mangle the sound in unintended ways - they should be "just" impedance controls without baggage. Some users may expect bigger sonic changes from huge impedance changes or major audible effects from varying the IRON content, but the folklore surrounding those ideas is maybe more dramatic than reality. These controls were designed to reflect reality which seemed appropriate on a high end Mic Pre and tend to be more like tweaks and trims. And this was more in line with the original concept of a basic good plug-in-a-mic-and-go preamp.

The TNT was getting interesting. We now had a tube preamp from the SLAM! on one side plus two solid state preamps working together on the other side, and each of those 3 circuits had a unique sound or subtle flavor yet there was some common theme or style.

The tube channel is based on a JFET / Vacuum Tube Triode cascode circuit that is quite unusual in that it is a blend of old and new components. The cascoded combination allows for high gain, low noise, and low distortion without using negative feedback.

On the solid state channel, one of the preamps also uses JFETs in a cascode topology. The high impedance circuit operates in the voltage mode. The ideal voltage amplifier would have infinite input impedance and the current flow would be zero. We use a paralleled cascaded ultra-low-noise FET / op-amp circuit and the TNT requires a pair of those hybrids for balancing (8 discrete matched FETs). There are a few interesting twists here too. When phantom power is engaged, it uses the conventional 6.8 kOhm phantom power resistors plus the usual DC blocking capacitors. So even if the impedance switch is set to 2 Meg Ohms, the phantom circuit limits the impedance down to about 14 kOhms. But if phantom is turned off, TNT removes those resistors (rather than switching them to ground, as is standard procedure) and bypasses the DC blocking caps, so that you truly have 2 meg input impedance and DC coupling restored. And no free lunch here either. Compared to the current mode amplifier, the voltage mode amplifier typically tends to have opposite characteristics in terms of its strengths and weaknesses.

The low impedance circuit is based on a special Lundahl transformer designed to operate in the current mode. This allows the transformer to work down to near DC yet be very small physically. This was paired with an ultra-low noise, ultra-low distortion op-amp that won in our listening tests.

By combining the JFET and Current mode preamps we were able to create a variable impedance that uniquely sensed and amplified both voltage and current. And this in turn provided much better gain consistency while usually sounding a bit better than either approach alone. In other words the volume didn’t jump or shift as the impedance control was changed unlike most (or all) other approaches. Some switch transformer taps, but transformer frequency response is very source dependent. And some preamps had 20 dB gain changes as the impedance knob was adjusted. Now that users can accurately hear and compare the effect of variable impedance without huge gain changes and without significant frequency response changes, it becomes truly interesting to hear what effects there are. In most cases these effects are not life changing. Even where low impedance settings seem to affect the damping or tightness a dynamic or ribbon mic might exhibit, one then may be faced with an unfamiliar sound from a familiar mic. This may be a good thing sometimes, but often one picks a familiar mic for its familiar sound. And one may hear similarly questionable highs in the high impedance settings. One might hear excess sibilance or harshness that may be due more to cables and cable distance than anything else. In the end, one may be most comfortable with the two middle settings of 600 and 2400 as the idea of extreme impedances gradually lose appeal.

Maybe the TNT might be viewed as a bit of a myth-buster in regards to “variable impedance”. Yes, it is sometimes useful, but can often be subtle - not exactly the most important feature for a mic preamp.

And then the basic premise of a tube preamp and solid state preamp being very different sounding animals might be a myth too. At one time we had both preamps tuned and adjusted to be extremely clean and transparent. One might guess that two transparent gain stages regardless of the technologies might sound the same, which was essentially inaudible. We had to go backward and re-introduce some of the “flaws” to recreate some of the creative differences that we all expected. We added some internal jumpers that essentially un-trim the tube bias trimmer pots that are tweaked to set up minimum distortion.

Same thing with the IRON control..... So much has been said about the sound of transformers lately that many people expect that one component to almost be responsible for a product’s signature sound - wrong! Sorry another myth. Most modern transformers are pretty transparent when used properly.

In fact, we had to use several techniques just to make the IRON control ‘audible’ including designing the transformer with unusually low permeability laminations, driving it with a non-optimum source, and forcing DC into a tertiary winding to create more distortions. In other words it was a bit of a fight to make it audible enough to wrap a control circuit around it.

Perhaps it is because those near opposite characters of the two circuits, that when combined or blended the audible benefits and strengths of each prevail, while the weaknesses of each are minimized. Of course, the better each circuit is optimized, the closer they tend to meet in the middle (transparency). Perhaps it is because the approach of respecting both voltage and current; it results in a form of optimal power transfer. Optimal power transfer is a very old topic in electronics and it relates to the old 600 ohm impedances pro audio inherited from the telephone industry and the 75 ohm terminations we need to be concerned with for word clock and video lines. Maybe with complex sources, there is valuable information carried both in voltage and in current and maybe most mic pre inputs are not as simple as a basic resistor. In other words, to some degree a typical microphone is a complex source (especially dynamic, ribbon and transformer coupled condenser mics) and a typical mic preamp input may also be a complex load (especially if it is transformer coupled) and the cable and connections between the two might also be viewed as a combination of resistance, capacitance, inductance and distance. So maybe it is all too complex to grasp without some serious computer modelling, but maybe it can be easy enough just to hear in some situations - and we'll leave that up to you.

We would like to speculate that one of the complexities that you might experience will be cable length. While we were designing the TNT we noticed excess high frequency sibilance coming in at the highest impedance settings. We finally traced it to the mic cables that we were using. If we doubled the length, the problem doubled and if we used a very short cable the problem disappeared. And the problem wasn't apparent at low impedance settings. OK but why? Here is where we have to speculate.

At low Z settings, it may be akin to our old 600 ohm terminated lines that pro audio inherited from the telephone industry. And that standard was set up to reduce echoes in early long distance lines. It also resembles the 50 ohm or 75 ohm terminated lines used for video and word clock where cable reflections impact high frequencies. Normally we don't consider audio frequency cable reflections to be a concern because they don't seem to affect the 20 kHz frequency response or square waves on our 'scopes. So our speculation might be a question. How far do these cable reflections need to decay (in dBs given that -60 dB is 1/1000) before they do not intersect with our abilities to perceive transients? Or how many microseconds of reflections and down to what dB? Just use a shorter cable.

Impedance Issues and Microphones

OK, the above might be a bit of technical mumbo-jumbo and what you really want to know is what to expect and listen for when you change the impedance switch in your session. Simply, at very high impedance settings, there are usually a little more highs. At low impedance settings, with dynamic and ribbon mics the bottom often tightens up. In the middle impedance settings, the preamp may sound closest to what you have grown to expect with that mic because most mic preamps are medium impedance and typically 1000 – 3000 Ohms.

With high impedance settings, one may be affecting the mic and cable in a few ways. First, with transformer coupled mics like most dynamic, ribbon, and tube condenser mics, one might be setting up a high frequency peak in the mic's transformer that may have been intended for 1 kOhm to 3 kOhm preamp impedances. The opposite is also common, where very low impedances may cause the transformer to roll-off highs earlier than the designer intended (yeah, but its **your** mic and **your** session, so choose the setting with **your** ears). The other effect goes back to that rambling about cables and time domain effects. Listen carefully for excessive sibilance and what might be described as an artificial harshness, and what perhaps the most finely honed ears will hear as time-smearing in the top octaves. This effect is directly related to cable type and length, and once you lock into it, you can prove it by doubling (or halving) the cable length. Even better is moving the TNT into the studio once you have your settings, and using a 4 to 8 foot cable from the mic to the preamp. It seems puny low level mic signals are more fragile to these effects than hotter line level (and robust line driver driven) signals but we don't know why. Give it a try. If this seems a bit inconvenient for level tweaks, add a simple passive variable attenuator (fader or pot) in the control room near your converter (if it has inconvenient input level adjustments) if you are a purist, or use the compressor or EQ gain controls, if you're not. The TNT has quite a bit of headroom (except in 60's / 70's mode) so there won't be much chance of overloading it and the real thing to keep an eagle eye on is the analog to digital converter at the end of the chain.

Away from the extremes, the 2K (2000 Ohms) setting represents the standard impedance of most mic preamps and what most microphones are designed to drive. In other words, its safe, and maybe a bit 'vanilla' and this isn't a bad thing. The 600 ohm setting is also pretty safe and may have some advantages because it gets closer to a 50/50 blend of voltage and current mode preamps.

At very low impedance settings with dynamic and ribbon mics another effect can come into play. Damping is a term normally associated with speakers that refers to the fact that a dynamic speaker comes to rest faster and is better controlled when connected to a very low impedance amp. (It was one of the biggest selling-point features when early solid-state amps with tons of negative feedback first came out.) Damping often has a dramatic effect on a speaker's frequency response and is one reason why some speakers work better with solid state and some better with tube amps (they were designed to, or what they were designed with). A similar effect often happens with dynamic mics and the bottom tightens up at low impedances. Whether this is desirable tends to depend on whether you are aiming for a tight and probably 'truer-to-the-source' sound or the sound of the mic that you and the world is more familiar with and might be viewed as more authentic or traditional. You might even use the tight 300 Ohm settings for tubby instruments and the 2K setting for drier sources and mics.

To complicate matters further, it depends on the mic.

Modern FET condenser direct coupled (transformerless) mics are mostly immune to whatever setting impedance you select, (though you might isolate some of those cable and preamp circuit effects described above) so expect generally very subtle or negligible differences. At worst, with loud sources and lowest impedances you might introduce early clipping with the occasional FET condenser mic. Tube condenser mics are fairly immune, but the impedance may affect the frequency response of the transformer. And with ribbon mics, one concern might be getting enough highs to start with, so you might want to especially watch out for losing highs while you focus on the tighter bottom with low impedances. But the good news is that ribbon mics tend to be famously forgiving when boosting the highs with a good EQ so it may be easy to "get the cake and eat it" this time.

And for those who just don't want to be bogged down by any technical issues and complications: you are in luck again – just turn the knob and pick the setting that sounds best for the track. In fact, this is the best advice for those that love all the technical explanations too, and when you get down to the session, the most important thing is to listen and choose based on the heart and the tapping foot rather than the intellect and some words in a manual or web-site. Remember the music, remember to listen. The old adage remains valid "If it sounds good, it is good".

We touched on using the impedance switch by ear and how the low impedance settings may be tighter in the lows, the medium impedance settings might be the ticket for the advantages of blended preamp whose settings represent typical mic pre impedances, and how the highest impedance may be useful for squeezing the last drop of highs (but not necessarily the most accurate highs).

One way (not the only way) to approach the IMPEDANCE switch is to begin at the middle or "600" setting. Listen to what you get. If the sound strikes you as OK but already a bit bright then try the lower impedance settings. If your first impression is that this instrument/mic sounds good but a little dark, then try the higher settings. Quite likely, your first impression is that it won't sound exactly OK or good enough, so the best advice is that you should really be out in the studio adjusting the mic position and you are not at a point where the subtle effects of adjusting preamp impedance will help enough. Maybe you started off on the wrong foot, or wrong mic in this case. You might try approaching mics the same way as the IMPEDANCE switch. If the instrument is hard sounding, try a softer sounding mic, and vice versa. When you have a bright, stinging instrument, maybe you don't want to use the brightest mic in the brightest position and coupled to a bright preamp in its brightest setting, followed with EQ in maximum "air" settings. The real trick to getting "air" is letting that track and the others "breathe", give it some room to move, rather than add some electronic artifacts. It doesn't take Einstein to suggest sounds or tones in a song are "relative" (and so is volume).

A proven approach is first to listen to the instrument in the studio, walk around, get a handle on where it sounds best and how the tone changes around the instrument. We do that because instrumental projection isn't necessarily obvious and because it gives you a starting point and the information needed to tweak mic positions. Then one might choose a microphone or 3, maybe based on complementary characteristics. We might also suggest experimenting with mic positions by ear rather than by the eyes, or ego. One might say the first task of an engineer is recreating the sound that the musician is hearing and intended. The second task might be understanding the musician intended it to sound better than what he got in the room and that maybe something larger than life (as opposed to squashed and smaller) might be what the sound becomes in the mix. Some of you laugh and say "Not my clients, not my mixes!" One can hope.

The Tube Channel

The left side (T) or channel one, is almost an exact replica of the SLAM! Mic Preamplifier and audio path. The only differences are the addition of the Impedance switch and an additional shunt regulator on the phantom power. The latter reduces noise and provides a softer start & off with the phantom power switching – less thump and less chance of input transformer magnetization.

After being introduced, the SLAM! immediately developed a reputation as one of the best and easiest to use tube mic pres. It seems to particularly shine on the traditionally most difficult sources like sax, brass, raspy vocals, and most percussion. It might not be the first choice for those looking for a dramatically colored preamp or those looking for gobs of tube-warmth (distortion) and it isn't a one-trick-pony that stamps its own personality on every sound. However, it does have both an input transformer (Lundahl) and output transformer (the one we developed for the SLAM!) and it does have two stages using tubes and it is true class-A from input to output, so yeah, it does have some character and tube-magic, and a tasteful amount of warmth. In other words, clean but not sterile, and it is neither flavor-less nor overbearing. Perhaps, the simplest description from a reviewer both describes this tube preamp's 'sound' and the designer's actual intention – "Just plug in a mic and it sounds great".

There are twin tube gain stages based on one of most rave-reviewed hi-fi phono preamps ever, the Manley Steelhead. These gain stages can be described as JFET-Tube cascode amplifiers. The FET is the first stage to keep the noise floor low, and the tube provides the bulk of the voltage gain. The beauty is that the FET and tube are so arranged to cancel the distortion of the other (complementary). The topology and balancing of this circuit has such low distortion that there is no need for negative feedback, (which might be appreciated by audiophiles). The circuit is also set up to compensate for both FET and tube variations and their drift.

This preamp also tends to have a great deal of headroom in most situations. While one can push the first stage hard to get some clipping when desired, one really has to try hard – this preamp wasn't designed to be an expensive fuzz-box. On the other hand, this is why it tends to work so well with difficult and complex wave sources and why it succeeds as a "plug the mic in and hit record" preamp.

Are there any tricks to using it, anything in particular to be concerned about or suggestions about its care and feeding? Practically none. Set the GAIN TRIM knob to near the middle or straight up as a starting point, turn the monitors down when plugging in a mic cable or switching phantom on or off (as usual), adjust the GAIN rotary switch to get an good level, hit record. The 4 LEDs on the far right are intended as a rough starting point to set that "good level" but there is ample headroom and low noise in the preamp that the LEDs mostly serve as 'signal present' and 'overload indicators'.

The IMPEDANCE switch can be set to taste. Probably the 4K setting may be the brightest or hardest, and the 600 setting the softest or phat-est. It affects the instrument ¼" jack that way too when fed by magnetic pick-ups. It probably won't have any effect on guitars or basses that have internal preamps, or after pedals, and probably no effect on synths. And on that note, before you write off a mono preamp as a last resort for synths because they all have stereo outputs – usually you can plug into just its left output and get the full sound, save a track, and not be semi-forced into yet another wide spread left-right synth sound. In the mix, give it a position, and add some good convolution reverb (especially authentic rooms), and maybe it will begin to resemble a real-life instrument and not a 'stereo-type' wash.

There are two outputs, a balanced XLR and an unbalanced ¼" jack. They sound a bit different and you can get a little variation of tone that way too. The XLR has a transformer in the path so it may sound a little "warmer", softer, rounder, and fatter - more 'vintage' or 'classic'. The ¼" output interrupts the feed to the transformer and the XLR, and sonically it might be called a hair brighter, harder, more transparent, or accurate, depending on who is trying to describe subtle little details in sounds, what aspect they are focusing on and what instrument is being fed through it. Of course, life is never quite that simple. There is a variable that can affect the XLR output and cause all the above generalizations to be thrown out the window. Transformers are dependant on the load. For example, if there is appreciable capacitance in the cable because it is a few hundred feet long, it'll probably cause some high frequency resonance (a few dB boost).

Tube Channel Continued

If the transformer is driving some heavy resistive load like some piece of vintage gear still set up for 600 Ohms then you can expect some high frequency roll-off and maybe a shift in the distortion and clipping towards the tubes and away from the transformer. Maybe the best approach is just to listen and compare, pick the best sounding output for this track, and don't worry about trying to label it or defining hard and fast rules for recording music (other than to impress your clients, of course).

Tubes will need to be replaced occasionally. Sometimes they last a few months, sometimes 30 or 40 years so about the only thing we can tell you is replace them when they get noisy, microphonic or the preamp stops passing signal. Generally, there won't be a major advantage with new tubes, broken in tubes, or esoteric and rare expensive tubes and in fact any of these might be worse. Here is the thing – there are 2 trims for the 2 JFET-Tube stages that adjust the bias and thus set the distortion null point. Inserting a fresh tube might require adjusting both of those trims, and that really should be done by a technician with a distortion meter for the best results. And there are another two trimmers that are "fine gain adjustments" for each stage, and while less critical, should be tweaked with fresh tubes. In other words, the performance of this preamp is equally dependent on the tubes and the trimmer tweaks, and the tweaks are there to get ideal performance from a variety of tubes and compensate for drift in a tube over decades. The JFETS used as the first stage relax the requirements for a super-low-noise, low-microphonic, expensive esoteric tube and allow those pesky trimmers that should optimize for a good variety of 12AT7A's.

So the preceding paragraph was aimed at those who like to buy $300 tubes on eBay, and that's OK, but most of us (and Manley Labs) tend to just use the $15 to $30 ones and get as good performance because we follow the procedures. And those who just need to change a 12AT7A and don't have access to a distortion meter or don't have the time, generally, it will work just fine and the difference between tweaked-out and not will be negligible – You see, the circuits are also set up to self-adjust to a large degree, so you can sweat the details or not, and usually be OK. And changing tubes is almost as easy as changing a light bulb, about as easy as changing a 9 volt battery in stomp-box and a whole lot easier than changing a transistor or chip, especially if its surface-mount. Your grandfather probably fixed the family TV any number of times. Relax.

About the only other "tricks" we might add here are more general and apply to most preamps and not just the TNT. Avoid plugging in mics, cables, mic patches, etc., when phantom power is turned on (and especially if the monitors are up). What can and often does happen is that one "leg" of the balanced line (Pin 2 or Pin 3) connects first, which can put a spike of 48 volts through a transformer and magnetize it. This has been known to damage ribbon mics and there are usually transformers in ribbon mics, dynamic mics, many condenser mics and of course many preamps, the TNT Tube side included. There are some engineers that like to demagnetize input transformers on preamps before big sessions. Probably a small, weak cassette tape-head demagnetizer won't be too effective because virtually all mic pre input transformers are mu-metal shielded, but a bigger pro head de-magnetizer, carefully and slowly brought near and away from the transformer is probably a good thing to use as yearly maintenance. One might also feed in a strong (say +25 dB) low frequency tone and slowly decrease its level to zero once in a while and get similar benefits. As far as those dynamic and ribbon mics and their transformers, best not to try because they also need those permanent magnets (in the capsules). It might be worth checking with the manufacturer on transformer coupled condenser mics, but probably they would rather you not take a chance or risk breaking something. So lets repeat, avoid plugging in mics, mic patches, etc., with phantom on so that you never have to worry about it. And lets also take the devil's advocate point of view to balance the issue. Originally phantom power was called "phantom" for a reason and most consoles of the 70's and 80's (before external mic pres) didn't have an on-off switch for phantom power, per channel or even global – it was always on. For 99% of us who were there, we always turned down monitors when changing mics, and we didn't think to de-mag anything except tape heads and we occasionally made great sounding records. Back then, we weren't looking for 'air', 'warmth', or 'loud', we were just having fun capturing first takes, hopefully exciting performances, and experimenting with mic choice and positions.

A real good trick with mic preamps, console channels, etc., that isn't mentioned nearly enough has to do with the PHASE switch. It is relevant here because we expect the TNT will often be a first choice for vocals and many simple overdubs. In a single mic situation, you in the control room probably won't be able to hear any difference with the phase switch in one position or the other so you might write it off as insignificant.

Consider the singer, headphones on, and getting a blend of their voice through bone conduction and those phones. All too often there is something in the chain with a polarity problem and it is usually a vintage mic or the headphones, but can also be caused by a wiring mistake or a power amp. Did you ever get one of those vocalists that continually complained about her voice in the cans? Did you try flipping the phase? One way will be thin and weird and the other will be hopefully better, but only the person in front of the mic can say. Might be worth singing into the mic yourself, phones on, before the session while somebody in the control room flips the switch for you. And while you're there, check out that headphone mix and level, and the room temperature and creature comforts. This will give you a bit more chance to work with the talent in the beginning of the session to also see if there is some choice reverb in the phones that helps her perform and hit those notes. Either that or spend more time auto-tuning later. These 'tricks' are not only limited to vocal sessions because a lot of times the talent is hearing a blend between the live room sound and the headphone feed. The benefits of good sound in the talent's phones can be subtle, if you as the engineer are focused on the 'sounds' in the control room, because the benefits tend to be in the performances. One might also consider that one's skills as an engineer are often more related to the performances and hit records that they have been 'lucky' enough to record than how great the mix was. Makes one wonder about little things like phase switches and using gear versus choosing gear.

As a matter of fact, one can view the TNT as a single piece of relatively simple gear that offers a fair number of tints to explore and use. But that is the key! One has to really dig in and explore the options and approach it like an instrument with many possibilities. Though it may be just another preamp that you try for 1 minute and see if it delivers a sound that you like, it should be approached as an instrument that needs some time to learn. After all there is a variety of settings on the Tube Channel, another bunch of settings to explore on the Cool Channel and maybe the sound that you are looking for is really there with a little coaxing in mic choice and positioning choice and maybe even some coaching of the talent to get that sound. Maybe one of the biggest tricks that we can share is that it isn't just the gear, it is how one uses it. And before that, it is about the source, the musician, the music and the instrument and the room, and you working with all those factors before going crazy with choosing between 8 different preamps. The preamp does represent many tints, but not prime colors.

More "Techniques" from other sources

For a really great source of tricks and techniques like these, there are too few books. An Australian engineer, who worked at Air Studios with Sir George Martin and even has a forward by him, has written one of the best books. The author is Michael Paul Stavrou and it is called "Mixing with your Mind" www.mixingwithyourmind.com, (Flux Research Pty Ltd, P.O. Box 397, Mosman NSW 2088 Australia) The inspiring part is that much of his focus deals with the counter-intuitive and non-geeky approaches that were learned the hard way through 20+ years of experience. For those struggling with technology, it may suggest that some fresh techniques that reduce techno-dependence, and for those that "just go for it" it may enlighten them to very practical acoustic and signal flow thoughts explained in easy visual metaphors. You may be approaching the task of recording in one way, and this book can pull you into a completely opposite alternative, which of course lets you then roam that entire space between your preconceptions and his. And because it really is difficult to describe sounds in ways that everybody understands, some of the labels and categorizations tend to be personalized and this makes for a great read.

Beyond that, it helps to know that "Stav" tends to be a brilliant recordist who truly gets results on tape that most of us would be jealous of. So he knows, and he has taken the time to share in print - very rare.

On the other hand, we might suggest avoiding the dry technical literature that seems to be the majority of the recent texts on recording technology. Those might be handy if you can make a living debating math minutiae or you are curious about those chips in the box you bought. However most of them are almost useless for both the recording engineer and the gear designers, and often cloud the real issues rather than help. When it comes to audio engineering, it mostly comes down to the ears and making tasteful decisions. It may be more an art than a science. It is not "paint-by-numbers".

The point is this: If you are hungry for knowledge and you search books and bulletin boards and magazines, it all helps. What helps most is advice from guys doing what you want to do and are deeply experienced and (importantly) are getting obviously good sounding results. Artistic mentorship is at least as valuable as it ever was, and is a huge advantage if you can get it.

THE COOL CHANNEL
Solid State or Right Channel

The No Tubes side of the TNT gives you a range of colors and might even be considered a contrast to the Tube side. Or maybe not. If one has strong ideas that tubes and transistors sound vastly different then the TNT might be a little unsettling. For what it is worth, both the Tube side and Solid State side of the TNT are pretty clean representatives of their technologies.

On the other hand it is extremely easy to find a variety of “personality preamps” using tubes (we make a few too) that all sound different from each other. Similarly there are many different solid state or discrete sounds. It largely depends on topologies, parts choice (especially transformers), and a variety of other choices a designer might make (or a cloner might copy).

As a “clean” preamp (CLEAN setting and IRON set fully counter-clockwise) we think the TNT Cool Channel may be hard to beat for sheer transparency and lack of electronic artifacts, especially in the 300 and 600 Ohm impedance settings. It is also designed to be useful as a fuzz-box or distortion device (60’s or 70’s setting, INPUT GAIN hot, GAIN TRIM fully counter-clockwise). It won’t simulate a Marshall stack or analog tape with VU’s buried deep in the red, but it shares some of those characteristics and might be an alternative sound, and at least sounds better than most preamps driven hard. Or it can give you a range of subtle flavors just as a basic preamp with the IMPEDANCE and IRON knobs, and/or by using the 60’s & 70’s settings, but driven lightly and before obvious distortion. These colors are set up to be easy and obvious to use, even though they are for the most part subtle. In fact, considering that this is really just a Mic Pre, and not a compressor or EQ, it seemed most appropriate to not go overboard and make it into an full blown processor, but rather give the user more control than is usual yet keeping it within a ‘safe’ realm where it is unlikely one will screw up a recording doing something that can’t be undone. That ‘subtlety’ might be obvious in the IMPEDANCE switch, that unlike other preamps that may have a 6 dB gain change from one setting to the next, and again, the TNT is designed to have maybe 1 dB of gain change from one extreme to the other, with most mics. We consider this important for selecting which impedance to use. Alternatively, comparing settings with radically different volumes makes the process near impossible to do especially when the differences tend to be subtle.

The IRON control may also be subtle in many situations because it uses the actual output transformer (which starts off pretty good) rather than a fake simulation just labelled to suggest a sound. The IRON control practically removes all audible effects of the real output transformer in the “0” setting and that can be easily verified by comparing the ¼” transformerless output to the XLR transformer output.

Just a little note on the subtleties of the Impedance and Iron controls: Maybe other units have more dramatic changes when you adjust similarly labelled controls and maybe what you are hearing then is flaws in the implementations or parts they use. We try not to do that, and we won’t con you and we did put some effort into maintaining constant levels as these controls are adjusted. We also try to maintain similar frequency responses when impedance is changed, where others don’t. So the TNT knobs do what they say they do, and don’t add strange misleading effects (for a change).

The NT preamp starts as 3 gain stages. 2 are identical and symmetrically used for opposite phases (XLR pin 2 and pin 3, i.e., balanced) and are voltage amplifiers like 99% of all mic preamps. Unlike 99%, these voltage amplifiers are very high impedance (2 meg Ohms) and exhibit very low capacitance and inductance, not that microphones need or are designed for that light of a load, but occasionally there are audible benefits. The third initial gain stage is a transformer input current-mode amplifier, which might be considered a rare technology in a mic pre and nearly opposite to the voltage amplifier topologies. These 3 preamps are variously selected and combined with the deceptively simple IMPEDANCE switch. Rather than a resistor that just shorts out the signal for lower Z, that resistor feeds the current mode amplifier in the TNT, and the part of the signal that is normally thrown away is amplified and mixed back in.

On the IMPEDANCE switch there are 5 positions. Furthest counter-clockwise is the “300C” setting , and next is the “300” setting. What is the difference? The “300C” position is only the current mode preamp by itself. The “300” setting uses both the current mode and the voltage mode preamps and is closest to a good 50/50 blend. The next setting, “600”, restricts some of the mic’s signal from going into the current stage and more is available for the voltage stage, and the blend is closer to 30/70. The 2000 Ohm setting and the 2 meg settings only use the voltage preamp.

The Impedance switch also has markings on the right hand side that are for the INSTRUMENT input only. How a magnetic pick-up reacts somewhat depends on the impedance it is driving. This effect is diluted because most guitars and basses have those volume and tone controls that pretty much set the maximum impedance the pickup will 'see'. The pickups also have to drive a length of cable, which often has appreciable capacitance and may roll off some highs. A fun trick for a session instrument is to bypass the volume and tone controls (maybe on a switch??), just leaving the pickup switch and jack, and then using a fairly short low capacitance cable into a variable impedance preamp like the TNT. In that situation the IMPEDANCE switch becomes very dramatic and the high impedance settings like "3 meg" and "10 meg" often get bright enough to sound much more acoustic-like. Typical authentic guitar amp tones are the 300K and 1 meg settings. The 100K setting may be similar (thick and un-bright) whether the volume control bypass mod is there or not. Guitars and basses with "active pickups" (battery included) and guitars feeding stomp boxes before the TNT, should be mostly immune to the IMPEDANCE switch settings and the 100K setting may have marginally less noise.

After the preamps are selected or are combined in a very transparent summing amp, the next stage is the "60's / 70's module" that contains a fairly simple class-A all-discrete circuit, and is inserted when the 60's or 70's switch is selected, or relay-bypassed when that switch is set to "CLEAN". The module is meant to simulate some of the qualities of vintage discrete nonlinearities, tape overload (what some call tape compression), and changes the frequency response slightly. It is not meant as a straight simulation of any particular piece of vintage gear and is meant to evoke some general characteristics of those eras and give the user a few more variations in the ol' tool-kit. The 60's/70's switch also adds DC bias to the output transformer to further simulate some old technologies (and which can be further adjusted with the IRON control). If sufficient interest is shown, Manley may be able to offer alternative "modules", as this block is easily removed and replaced.

Then the signal hits the GAIN TRIM conductive plastic pot, and is routed to the RAPTURE AMPS, which are the final line drivers in this preamp. It drives the ¼" output directly and the output transformer (custom designed and manufactured in house for the TNT) for the XLR output. Two controls are wrapped around the RAPTURE AMP.

One is the IRON knob, which compares the input and output of the transformer, derives an error signal, which is then sent to the IRON pot, that acts much like an EQ boost/cut pot with zero 'effect' at 12:00 or straight up. This way the user can reduce the effect of the transformer to near inaudibility or exaggerate it or just leave it as it is where this additional circuit is essentially bypassed and the transformer output is just pure conventional transformer and the ¼" output is just clean and unaffected. In fact, if you use the ¼" output and adjust the IRON control counter-clockwise, you get an effect that might be called "Anti-Iron" and could practically reduce the effect of a transformer in the next piece of gear following TNT, such as a compressor – kind of like "cleaner than clean".

The other circuit wrapped around the RAPTURE AMP is the OUTPUT MODE switch on the back panel. Rather than use the conventional cross-fed feedback output that many use to make a balanced output a bit fool-proof and to simulate a transformer and how it accommodates balanced and unbalanced inputs, we have this switch. Why? For one thing, we have the real transformer, for another, those circuits are inherently usually within .5 dB of being called an oscillator, and thirdly, the switch is 3 position, so we can properly accommodate +4 unbalanced, +4 balanced, and –10 unbalanced – all driven low impedance, high current (full headroom down to 50 Ohms) and unconditionally stable. One might also note that these two outputs are isolated and can serve to drive two separate distant destinations easily.

The RAPTURE AMP itself was the result of months of auditioning almost every discrete op-amp and gain stage and every chip op-amp known to us to ever be used for an audio product or published DIY project. And we experimented with most of the tricks used to further enhance all of these op-amps. In the end, we found a circuit that was practically unique in its lack of artifacts and sonic purity. After all those months of R&D, it was decided that it should go in a block of epoxy.

A similar routine of comparing the whole TNT preamp to a lot of known expensive reference preamps was performed while also continually comparing the raw source to the preamps attenuated. Lets just say we are confident that it is a winner and particularly true to the source. Hope you like it!

Any particular tips using the NT channel?

Techniques specific to the solid-state side of the TNT are not daunting. Most likely the first thing to get your attention will be the LED metering because it is a bit unusual. The two bottom LEDs show the first few stages of the preamps or what is going on with the stepped gain switch, but BEFORE the Gain Trim knob. The top two LEDs show that Gain Trim Knob and the final output.

The bottom GREEN LED shows “SIGNAL PRESENT”. The RED LED above it shows when one is beginning to distort the preamp or the 60’s - 70’s circuit. So use this RED LED to help set how much “OVERDRIVE” and a typical setting makes that LED flash about 50% of the time - but trust your ears - too much overdrive may be hazardous to one’s career as an engineer. You can’t undo mic pre distortion.

The top two LEDs are for the final output and the Green LED is intended to show a good level to your converter and the RED LED is intended to indicate probable A to D converter overload. Originally we had that RED LED indicate when the Preamps began to clip, but they tended to stay dark, so now they just help set a level for the next piece in the chain.

A good starting point for the 11 position GAIN switch is “counter-clockwise” and the GAIN TRIM should be set at roughly 12:00 or straight up. Turn up the GAIN switch until you are getting a good level. You might use the 4 LEDs on the TNT. The first or bottom is a “signal present indicator” and lights up about 20 dB below optimum levels. The second from the bottom LED shows clipping in the first stages and follows the GAIN switch, but is before the GAIN TRIM pot. In normal clean operation this LED shouldn’t light up and is typically ‘skipped’. Where it will come in particularly handy is if you are operating the preamp to deliberately over-drive it by turning down the GAIN TRIM pot and turning up the GAIN switch. It should be kept in mind that this won’t be easy in CLEAN mode, but it becomes quite easy in 60’s and 70’s modes. An interesting sound is when that second LED is just occasionally flashing in the 60’s/70’s modes, and the distortion is subtle, and creates a psycho-acoustic effect of “character”, “richness”, and a 3D effect.

When that LED is glowing steadily, the distortion should be pretty obvious. The LEDs are designed to flash on both positive and negative peaks, and slowly fade. A lot of peaks are too fast to see when the LED directly displays peaks and the fade gives some clues to the duration and musicality.

The top two LEDs are set to show output levels at the output jacks. These are factory set up for +14 dB and +20 dB over our standard +4 dBm, so they are technically at +18 dBm and +24 dBm. Most A to D converters are set up for +16, +18 or +20, so those two LEDs should get you to a reasonable starting point and give you a pretty good idea when clipping is likely. Of course, you really do need to watch those A to D meters, or tape machine VU meters as the final judge, especially when you have processing between the preamp and recording device. However, if need be, there are internal trims for the LED thresholds, so one can set up the LEDs for particular needs. It might be noted that there is no LED to indicate when the TNT clips – because it clips at over +30 dBm which is probably higher than 98% of the gear it might be driving. It seemed more useful to indicate where the next device is likely to run out of headroom, besides using the TNT for its own overdrive effects is covered by LED #2.

If you tend to want the Cool Channel to be generally your “clean preamp” and intend to use it as a reference especially for acoustic instruments, then you might want to consider using the 1/4” output as the main output because the output transformer is not in that path. And keeping it clean, you may want to leave the “IRON” knob set at 12:00 or straight up. Turning the knob either counter clockwise or clockwise introduces some subtle transformer color. Now if there is another transformer down the chain in another piece of gear (or even a magnetic tape recorder), the counter clockwise settings may help minimize it’s contribution. You might be able to use the 60’s - 70’s switch if you keep the Input OL (overload) LED off by keeping the Gain Switch lower and the Gain Trim higher or near its max of +10 dB.

If you want the Cool Channel to be multi-tinted then maybe the XLR output should be your choice. Then experiment with all the preamp’s controls keeping in mind that moving a microphone by a few inches might affect the color more, maybe a different mic might be more significant a variation, and once you hit “record” the player’s tone and volume usually change anyway.

Like most mic preamps, the TNT has high-pass filters to remove unwanted low frequencies. The most common situations to use it is for vocals to minimize “pops”, wind noise, and air conditioning rumble or leakage, or for sounds that have little low frequency information anyway, like acoustic guitar or high hats. The basic idea is removing the garbage before it gets recorded. You just might want to compare the effect of a good analog high pass filter compared to the standard digital ones and also check out what happens when both are used, from time to time. There are some engineers that use a combination of a high pass filter and a low frequency boost EQ to get a tight fat bottom. For that situation, we might suggest auditioning the combination, but recording only the high pass filter, saving the boost EQ for playback and refining in the mix.

There is a 3 way toggle labelled “CLEAN”, “70’s”, “60’s”. The CLEAN setting has the TNT operate in a standard clean mode. In fact, we probably could have labelled it “damn stunningly clean” if there was enough panel space. Telltale things to listen for if you want to compare clarity are ‘harshness in the highs’, ‘smoothness and liquidity in the mids’, but also ‘snap and punch’ and ‘dynamics in the deep lows’. It is hard to overdrive this mode. We should point out that for “absolute clean” use the ¼” output and the IRON knob set to 1 (12 o’clock, straight up) to avoid any transformer coloration. And “virtually absolute clean” is using the XLR transformer output along with the IRON knob set to –1 or fully counter-clockwise. This method compensates for the transformer. Often the 300 setting on the IMPEDANCE switch sounds slightly clearer as well. And while we are at it, if pure-clean is the goal, choose your mic carefully and avoid processing or choose it extra carefully because it all adds up, and nothing is perfect. In other words, maybe use nothing, except a great passive ribbon mic, TNT, and a great converter running at 192K.

So if you are comparing preamps someday, your natural tendency may be to set everything as flat and level as possible on each one, use the same mic and try to judge which basic sound you prefer. If it seems you tend to pick expensive transformerless discrete preamps, the TNT should do well that way. However, if you tend to prefer color boxes, then maybe you should be working with the TNT controls, setting up modes and gains appropriately. Otherwise, it would be like comparing a several different cars without adjusting the seats or mirrors. Even if the range of colors isn’t drastic, one is still expected to adjust to taste.

While the CLEAN mode is a relay bypass of an additional module, simply selecting “**60’s**” or “**70’s**” engages this special module. It is a discrete class-A circuit meant to simulate some of the stronger characteristics of circuits and analog tape of those eras. The most important thing that we must point out is that it is purposely very level dependent and that it is between the two gain controls so that the user can drive it as hard or soft as they want to get a pretty wide variation in tones. Driving it very lightly by keeping the GAIN switch lower and the GAIN TRIM way up mostly affects the EQ and introduces a faster roll-off in the deep lows and some shaping in the highs, so that there is a subtle “presence” boost. Driving it a little harder is particularly interesting and one begins to hear typical vintage sounds including 3D depth, richness and edge, and a little further becomes a bit ballsy, aggressive, and forward. The trick is careful adjustments of those gain controls and mic technique to land in the “sweet spot”. This module can also be driven quite hard and be used for some obvious dirt and you may find that the character of the distortion is unlike most clipped electronics and perhaps smoother, like analog tape or a guitar amp. For the best overdriven guitar effects, you may want to combine TNT with some good stomp boxes driving into the instrument input, and some EQ at the TNT output, and you might try feeding a real speaker/room/mic or convolution reverb so that it doesn’t end up too dry and clinical.

There isn’t much that can be said about the difference between the 60’s and 70’s settings. “60’s” has subtler shaping in the distortion and should sound brighter when pushed hard. “70’s” uses more drastic shaping and forces the output transformer to be biased, much like some famous old British console electronics of that period. One might say the “60’s” setting is more like tubes and tape, and the “70’s” more like discrete and transformer, depending how one associates tones. In both cases, the distortion starts off mostly even order and becomes more odd order as it is pushed harder. Both tend to help get hotter levels due to softness of the clipping character. Both are pretty easy to overdrive and require care to avoid effects that cannot be undone. Keep in mind that usually an instrument starts softer and gets louder during the song and during the session and that our ears become less sensitive & critical as the session wears on. When in doubt, back down the GAIN switch one notch – better safe than sorry.

The IRON knob may disappoint some who expect a radical and obvious effect. It can be subtle, especially for mid-range dominant sounds like vocals and guitars because, for the most part, a transformer affects the extreme lows, extreme highs, and to some degree the dynamics. This control is build around the actual output transformer and, lets face it, a good audio transformer shouldn't be particularly colored and messy. On the other hand, it has always been the transformers in vintage gear that helped them sound warm, smooth, and round, yet were quite OK at passing audio pleasantly and without major damage. The IRON control allows you to adjust the "Transformer Contribution" from near zero audible effect, through to "typical" at 12:00, and further to where the effect is exaggerated and practically tripled which, by the way, is still not always obvious.

At very low frequencies a transformer can be said to have 3 significant effects: 1) it rolls off the subsonic frequencies starting around 10 Hz, which is audible due to the phase shift that happens mostly below 50 Hz, which acts to slightly delay the extreme lows. 2) Hysteresis shapes the waveform for low level, low frequency signals, putting a couple of bends around the zero-crossing, which simply adds harmonics. 3) Hot low frequency signals can cause the transformer to saturate or overload, again causing harmonics and may cause earlier roll-off and more phase shift. It is in the transformer design details that set the amount and balance between those effects, and is mostly a function of the core lamination material and size (weight or bulk), but also a function of the number of turns of wire and how the transformer is driven, and the expected load. We design and manufacture these transformers in house at Manley Labs.

At very high frequencies, the biggest effect is an ultra-sonic roll-off. This can be as gentle as 6 dB per octave for several octaves, but typically becomes 18 dB per octave. In fact, the way that capacitance between windings and leakage inductance interact, it usually creates a significant bump in the frequency response somewhere between 30 kHz and 100 kHz, which is then tamed, as standard procedure, by adding a resistor and capacitor at the output or secondary, leaving a smooth predictable roll-off usually around 80k – 120 kHz. This still leaves some phase shift in the 5k - 20kHz range which is audible and gives one the impression that the highs are softer and don't extent to infinity.

This may surprise some people, but some designers and researchers, including yours truly, have experimented and determined for themselves, at least, that we might need a frequency response out to 500kHz or 1 megaHz to completely avoid any audible phase shift in analog audio circuits and where absolute transparency is the goal. Then again, our ears adjust rapidly to rather drastic roll-offs even as low as 10kHz and we often choose a rolled-off high frequency response as being more pleasant and comfortable. And some might say, "no microphone or speaker can reproduce those frequencies, so why bother?" but these effects are relative and additive, so every little bit may be audible for those that can compare. And considering the best audio systems only approximate full range live performances, then maybe one of the significant reasons is these infra-sonic and ultra-sonic roll-offs happen too early, causing phase shift in the audible spectrum and audible time smear. Dynamic performance is another reason, but lets not dwell on that now. Given the capabilities of modern audio and the demands of the audience, much of the goal is creating illusions and emotional impact. This is how we should approach the controls on the TNT, and maybe what should become the criteria for choosing it against other preamps.

By now, you have probably correctly guessed that turning the IRON control counter-clockwise results in a cleaner, tighter bottom and extension in the highs. Turning the IRON control clockwise results in a rounder, smoother, warmer, and softer tone. The mids won't be affected much, and mostly only in a relative way compared to the lows and highs, but if your ears are great, you might notice a subtle effect with IRON maxed where the mids arrive slightly before the lows and highs and that this is a big part of the vintage sound. And if you have been reading carefully, the transformer is biased in "70's" mode so the IRON control will have a slightly different (and greater) effect. You may have also picked up that the IRON knob acts like an EQ knob, where it has near zero effect as a circuit when the knob is straight up at 12:00.

We also mentioned that the ¼" output does not use the transformer but is affected by the IRON knob. As before, the IRON circuit is essentially bypassed when the knob is straight up. Turning it counter-clockwise introduces the transformer correction circuit, and with this output creates an Anti-IRON effect. Extreme highs and lows are boosted slightly. As the Iron knob is turned up, you introduce the effects of a transformer into a transformerless output.

The ¼" transformerless output has a switch on the back panel that allows it to PROPERLY drive +4 dBm balanced inputs, +4 dBm (pin 2 hot) unbalanced inputs (rare enough), and –10 dBu unbalanced inputs. In each case, a very stable and very low output impedance is given that does not suffer and cause common mode rejection problems when driving less than perfect cables and inputs. Typical cross-coupled line drivers magnify their source resistors while attempting to deal with imbalances, which only magnifies common mode rejection problems. For example, a great input stage may have a CMRR spec of 60 dB at 1kHz, (and not in the specs is that this typically drops to 30 dB at 20K, may or may not be 80 dB at 60 Hz, and may be 30 dB at 10 Hz), but CMRR is more dependent on the output impedance match of the device driving it, and that 60 dB spec can drop to 20 dB when there is a 10% difference between source resistance from Pin1 & Pin2. Cross-coupled line drivers create CMRR problems for the sake of idiot-proofing and people using funky cables and adapters. The TNT just uses a switch.

Transformers have traditionally been great at the idiot-proofing aspect and are very "forgiving" and solve more practical connection issues than they create. However, the back panel switch does have an effect on the XLR transformer output. For both ¼" and XLR outputs, in the "+4 balanced" mode, headroom reaches +30 dBm comfortably, but drops to +24 dBm when the switch is set to "+4 unbalanced", which probably close to maximum level of most A to D converters. This might be used to choose which box clips first and best. The impedance on pin2 and pin3 remains equal and constant and low, so it will work better than most balanced outputs into a balanced input even though it is labelled "unbalanced". In other words, when the switch is set for "unbalanced", it will drive both balanced and unbalanced inputs fine, and unbalanced inputs at the right level (no 6 dB drop), but does have 6dB less headroom, which should still be good enough in most situations (balanced into balanced is the best choice generally). In the "-10 unbalanced" mode the ¼" jack drops the appropriate 12 dB (not 14) and the XLR drops 6 dB. Much semi-pro and most consumer gear is happiest with –10 dBu signals, but you may need a ¼" to RCA adapter. Again, this can be used in some situations other than "standard procedures". One can use one output to drive an input in one room, the other to a different room (like an amp in the studio or a remote truck) and avoid a lot of hassle with splitter boxes or ground loops and interaction.

Please note that the tube (T) channel has a different output arrangement, uses different technology, and has a ¼" output that is only unbalanced and doesn't have the back panel switch, yet the ¼" output will still be OK to drive balanced inputs, especially if one is trying to avoid transformers today.

TROUBLESHOOTING

There are a number of possible symptoms of something not quite right; some may be interfacing, others we will touch on as well. If you suspect a problem, the following paragraphs should help.

NO POWER, NO INDICATORS, NADA - Probably something to do with AC power. Is it plugged in? Check the fuse on the back panel. A blown fuse often looks blackened inside or the little wire inside looks broken or it's resistance measures higher than 2 ohms. A very blackened fuse is a big hint that a short occurred. Try replacing the fuse with a good one of the same value and size. If it blows too, then prepare to send the unit back to the dealer or factory for repair. The fuse is a protection device and it should blow if there is a problem. If the unit works with a new fuse, fine, it works. Sometimes fuses just blow for unknown reasons.

LIGHTS BUT NO SOUND - First try plugging the in and out cables into each other or some other piece of gear to verify that your wires are OK. If not fix them or replace them. Assuming that cables passed sound - it probably is still a wiring thing. The output XLRs are transformer balanced which require both PIN 2 and PIN 3 to be connected somewhere. When driving an unbalanced input (inserts on some consoles) PIN 3 needs to be grounded or connected to PIN 1. Same with the unbalanced 1/4 inch jacks - if driving a balanced input you can't ignore the negative side. It needs to be connected to the sleeve of the phone plug. Another way to do basically the same thing is join PIN 1 and PIN 3 on the XLR male at the destination. Easiest way - Use the balanced with balanced, unbalanced with unbalanced. That is why the options are there.

LEVELS SEEM TO BE WRONG, NO BOTTOM - Several possible scenarios. Manley uses the professional standard of +4 dBm = Zero VU = 1.23 volts AC RMS. A lot of semi-pro gear uses the hi-fi reference of -10 dBm = Zero VU. This is a 14 dB difference that will certainly look goofy and may tend to distort. Often there are switches on the semi-pro gear to choose the pro reference level. If the loss looks close to 6 dB and it sounds thin then one half of the signal is lost. The cause is probably wiring again. One of the two signal carrying wires (the third is ground / shield on pin 1) is not happening. Check the cables carefully because occasionally a cable gets modified to work with a certain unit and it seems to work but its wrong in other situations. If only one side of the TNT exhibits this problem, it may be a problem in the TNT.

HUM - Once again - several possibilities - several cures. Most likely it is a ground loop. Ideally each piece of gear should have one ground connection and only one. However, the short list of grounds include the AC mains plug, the chassis bolted to a rack with other gear, each input and each output. The two most common procedures are: try a 3 pin to 2 pin AC adapter (about a dollar at the hardware store).This while legal in many countries can be dangerous. Method two - cutting the shield on one end of each cable. This is done by some studios at every female XLR to "break" all ground loops. All the other gear in the rack is "dumping" ground noise onto the ground. Try removing the EQ from the rack so that it is not touching any metal. You just may have cured a non-loop hum. Some gear radiates a magnetic field and some gear (especially if it has audio transformers or inductors) might receive that hum. A little distance was all it took. It is worth a few placement experiments if you notice hum.

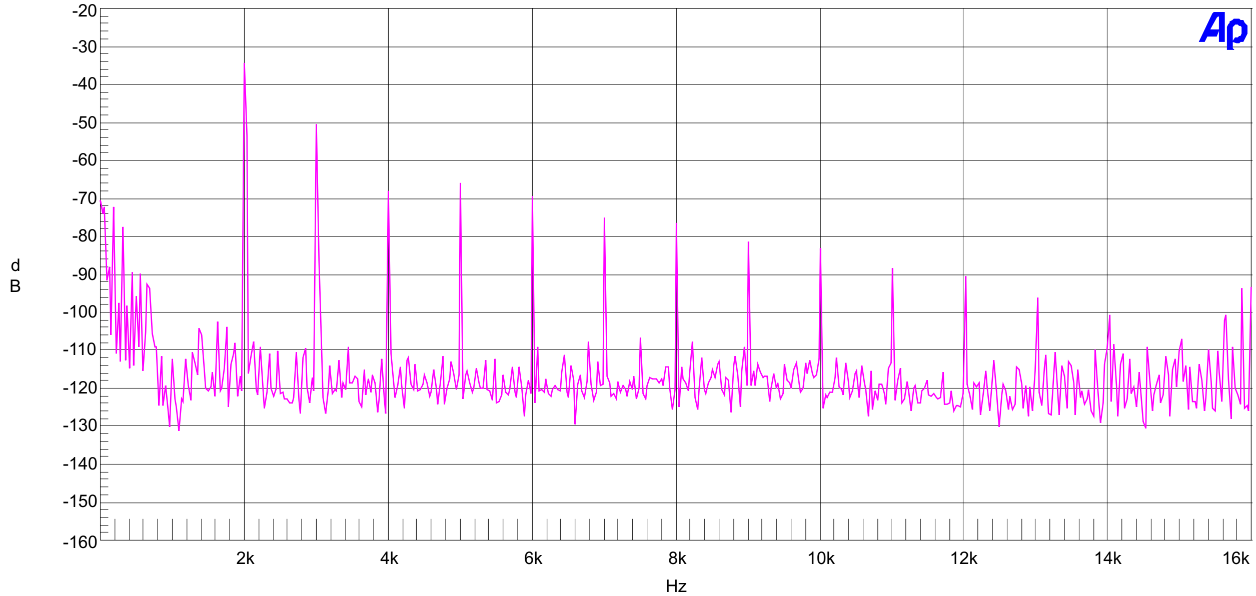
IT MAKES NOISES WHEN THE FRONT PANEL IS TAPPED - An easy one. Some tubes become microphonic over time. That means they start acting like a bad microphone. Vibration has caused the supports for the little parts in the tube to loosen and now the tube is sensitive to vibration. Easy - Replace the tube. Which one? The one that makes the most noise when you tap it. Usually this will be the smaller tube closest to the front. The TNT will have to be on, connected and speakers up but not too loud for the sake of your speakers.

Further Questions? Please contact our Service Department:

+1 (909) 627-4256 x325

Audio Precision 06/26/06 18:06:03

TUBE CHANNEL DISTORTION FFT



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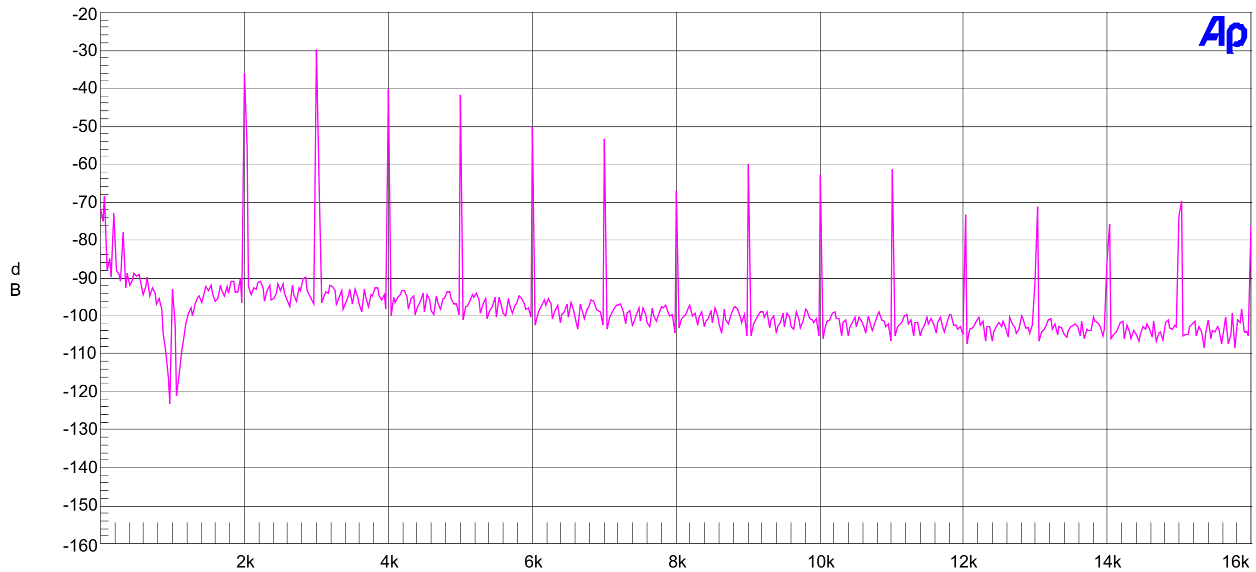
FFT SPECTRUM of THD Residual @ 1kHz

TUBE CHANNEL DRIVEN JUST 2 dB HOTTER



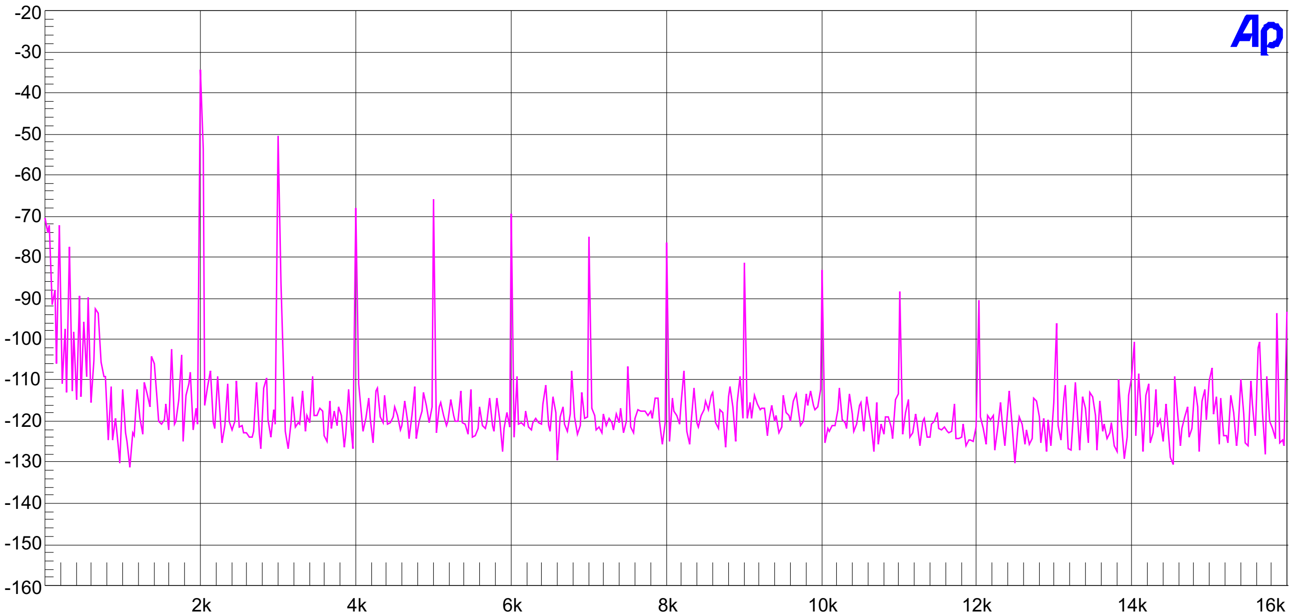
The TOP graph shows the distortion harmonics just prior to full clipping. Notice both even and odd harmonics and a smooth decay of upper harmonics.

The BOTTOM graph shows just slightly hotter signal brings the unit into deeper clipping and the symmetrical clip brings in more odd harmonics and involves more upper harmonics.

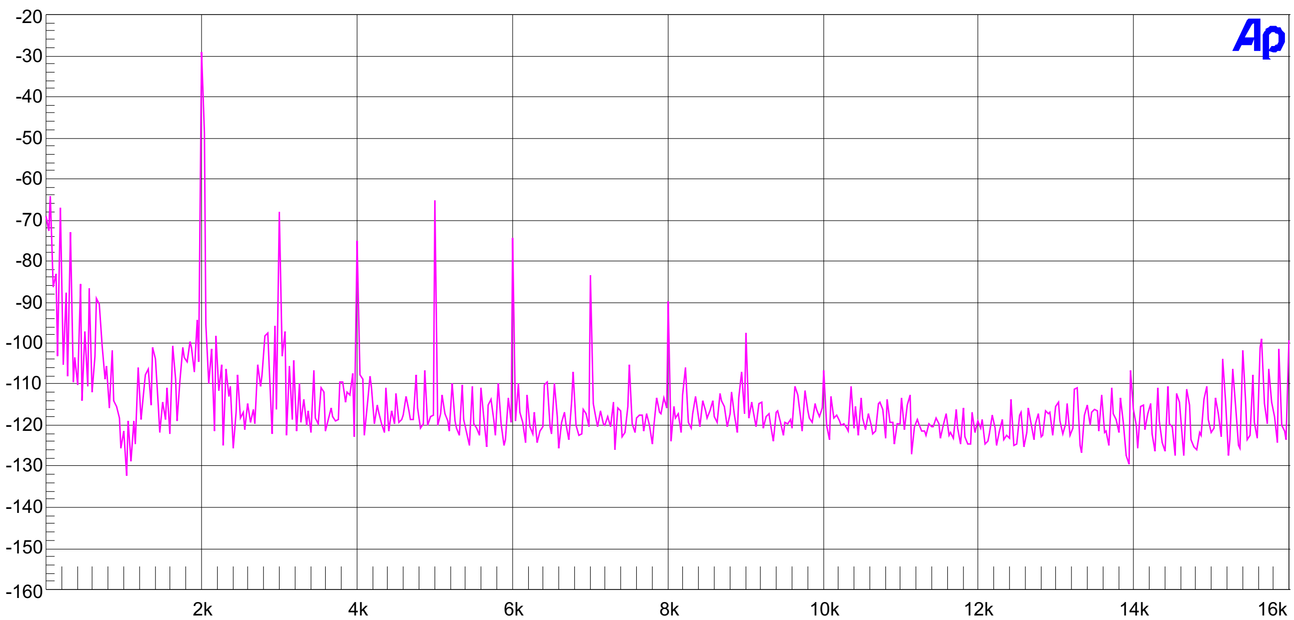
These graphs illustrate that tube circuits do not just create even harmonics and that they don't just create low order harmonics, nor does clipping just produce odd harmonics. It is a little more complex than that. It depends on the topology, circuit details, levels and measurement techniques.

What matters in this case is the lack of harmonics before clipping (low THD) which is not graphed here because it is just a boring noise floor

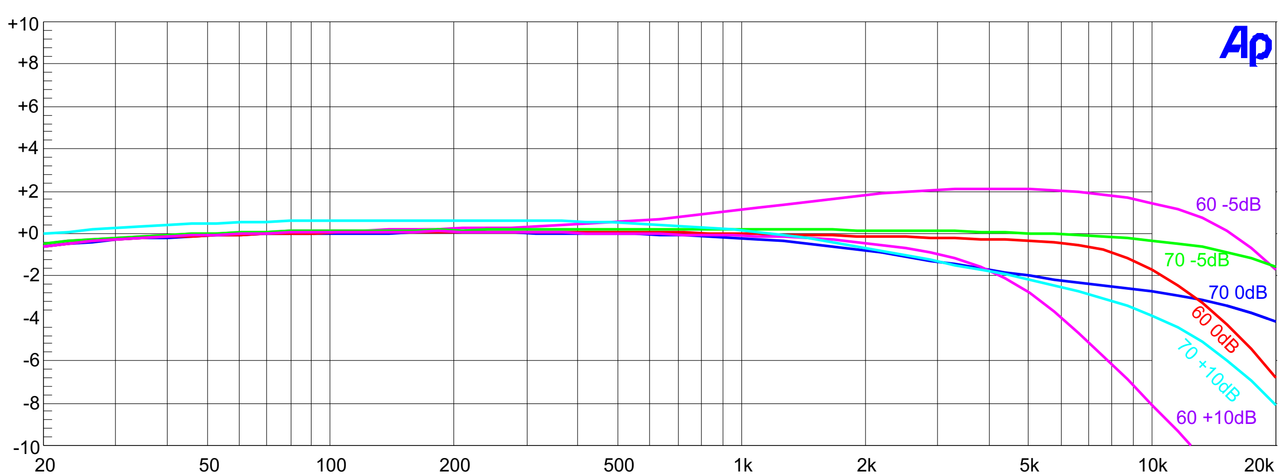
60s Switch ONSET OF DISTORTION SPECTRUM



70s Switch ONSET OF DISTORTION SPECTRUM

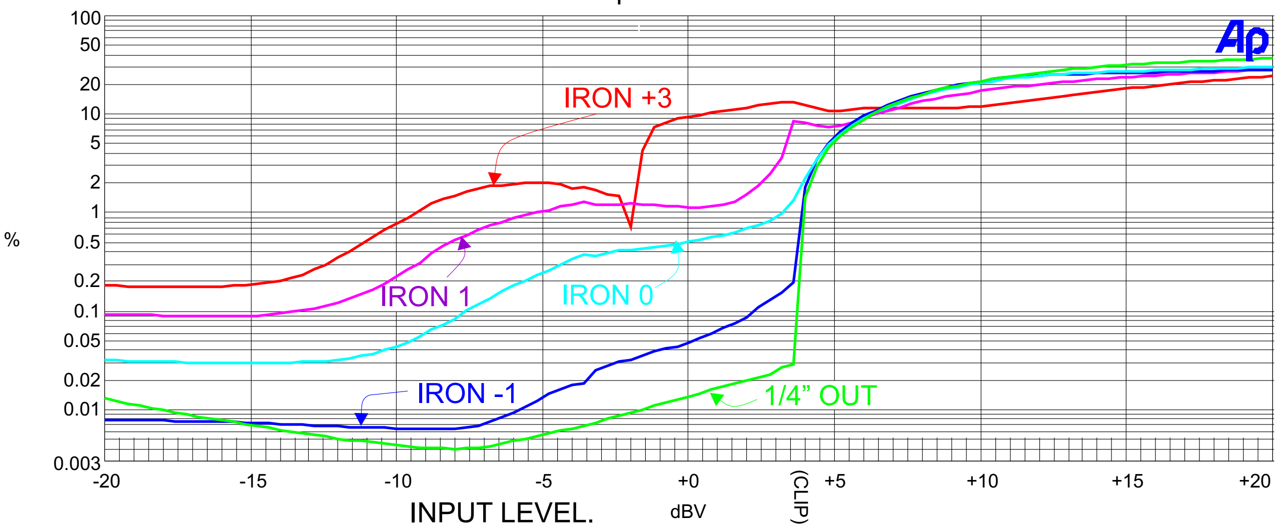


60s - 70s SWITCH FREQUENCY RESPONSE AT VARIOUS GAINS & COMPENSATED with TRIM POT

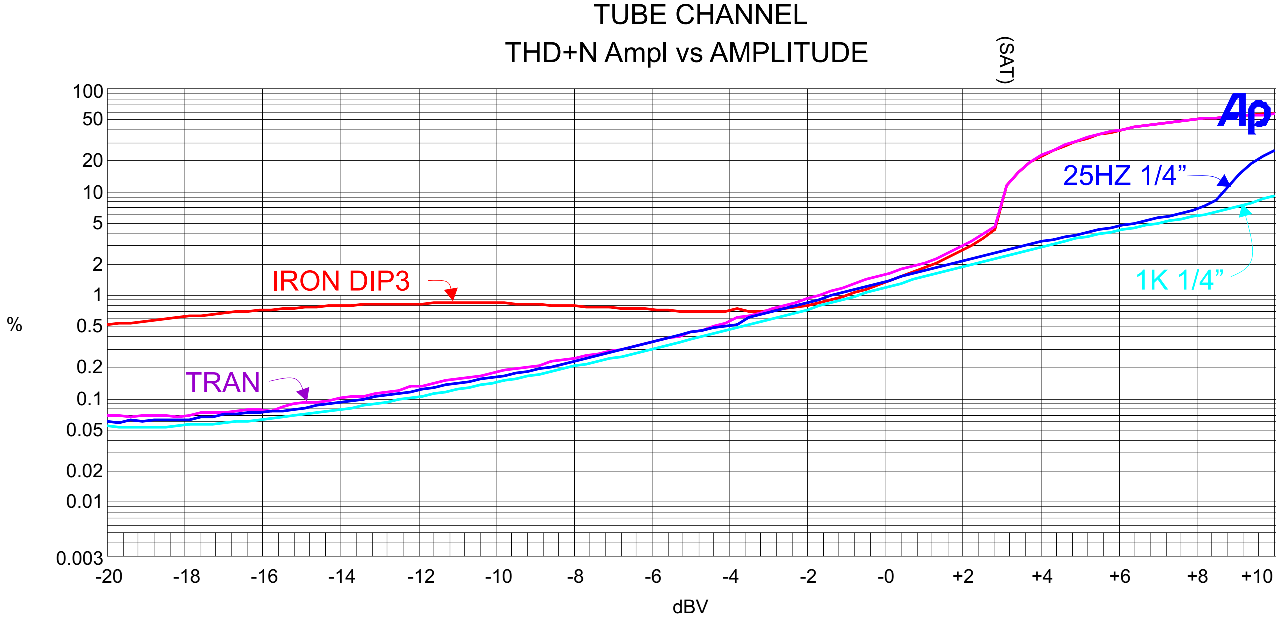


0 dB REFERENCE IS ROUGHLY AT ONSET OF DISTORTION AND WHEN OL LED BEGINS TO LIGHT

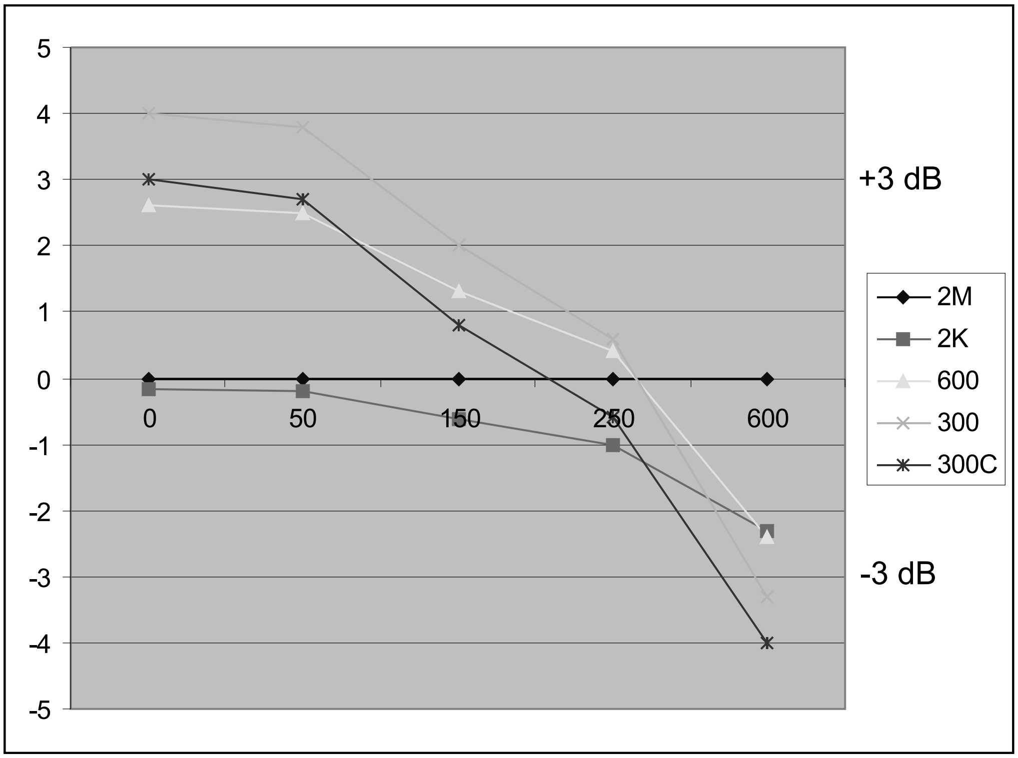
COOL CHANNEL
Gain Switch @ 20, Gain Trim @ 0. 4x IRON SETTINGS + BYPASS
THD+N Ampl vs AMPLITUDE at 25 Hz



TUBE CHANNEL
THD+N Ampl vs AMPLITUDE



COOL CHANNEL IMPEDANCE SWITCH RELATIVE LEVELS



Microphone Output Impedances

The middle horizontal line suggests a range of typical microphone output impedances. For example, many mics are approximately 150 ohms, so for those, changing the TNT Cool Channel Impedance switch alters the level by 2.5dB going from the highest setting to the lowest. If you have other preamps with an impedance control, you might be used to much bigger level changes plus strange tonality changes. You might be accustomed to that preamp's design flaws.

With the TNT, changing the Impedance switch tends to be subtle. Worst case would be very low source impedance which corresponds with 4 dB gain change across the five Impedance switch settings. Best case would be 200 to 600 ohms mic output impedance, which corresponds with 1-2 dB gain change across the five switch settings.

With some mics like FET condenser types, expect very little tonality change. With ribbon mics or dynamic mics, one can alter some aspects of their damping. However, it will be the mic itself changing and sometimes mis-loading its output transformer rather than switching the preamp's input transformer into the technical twilight zone...

MAINS CONNECTIONS

Your TNT has been factory set to the correct mains voltage for your country. The voltage setting is marked on the serial badge, located on the rear panel. Check that this complies with your local supply.

Export units for certain markets have a moulded mains plug fitted to comply with local requirements. If your unit does not have a plug fitted the colored wires should be connected to the appropriate plug terminals in accordance with the following code.

GREEN/YELLOW	EARTH
BLUE	NEUTRAL
BROWN	LIVE

As the colors of the wires in the mains lead may not correspond with the colored marking identifying the terminals in your plug proceed as follows;

The wire which is colored GREEN/YELLOW must be connected to the terminal in the plug which is marked by the letter E or by the safety earth symbol or colored GREEN or GREEN and YELLOW.

The wire which is colored BLUE must be connected to the terminal in the plug which is marked by the letter N or colored BLACK.

The wire which is colored BROWN must be connected to the terminal in the plug which is marked by the letter L or colored RED.

DO NOT CONNECT/SWITCH ON THE MAINS SUPPLY UNTIL ALL OTHER CONNECTIONS HAVE BEEN MADE.

Note: This unit has been factory wired for your country. If you plan to take the unit to countries with a different mains voltage you will need to send the unit to Manley Labs for the correct power transformer - or use AC voltage converters.

SPECIFICATIONS

TUBE CHANNEL (Left Channel)

GAIN RANGE		+10 dB to +80 dB (20-70dB in 5 dB steps)
MIC INPUT IMPEDANCE		600, 2400, 10,000 Ohms
INSTRUMENT INPUT IMPEDANCE		300K,1Meg, 3 Meg Ohms
FREQUENCY RESPONSE	(-1 dB)	10 Hz to 30 kHz
	(-3dB)	5 Hz to 45 kHz
THD & NOISE (1 kHz @ 20 dB gain)		.035% typical (-68 dB)
NOISE FLOOR		-60 dB typical referenced to +4dBm
EIN		-120 dB
SIGNAL TO NOISE @ 70 dB Gain		95dB (TRIM has major effect)
MAXIMUM OUTPUT (1% THD) BALANCED +4		31 dBu

COOL CHANNEL (Right or Solid State Channel)

GAIN RANGE		+10 dB to +80 dB (20-70dB in 5 dB steps)
MIC INPUT IMPEDANCE		300, 600, 2000, 2 Meg Ohms
INSTRUMENT INPUT IMPEDANCE		100K, 300K,1Meg, 3 Meg, 10 Meg Ohms
FREQUENCY RESPONSE	(-1 dB)	15 Hz to 30 kHz
	(-3dB)	5 Hz to 50 kHz
THD & NOISE		.01% typical (-80 dB)
	(no distortion products measurable,	noise dominates measurement)
NOISE FLOOR		-66 dB typical referenced to +4dBm
EIN		-126 dB
SIGNAL TO NOISE @ 70 dB Gain		85dB (TRIM has little effect)
MAXIMUM OUTPUT (1% THD)	BALANCED +4	30 dBu
	UNBALANCED +4	24 dBu
	UNBALANCED -10	16 dBu
DISTORTION @ 1K WITH 60' / 70's SWITCH		1.5% typical with LED onset (+4 dBu out)
		3% typical with LED fully lit (+7 dBu out)

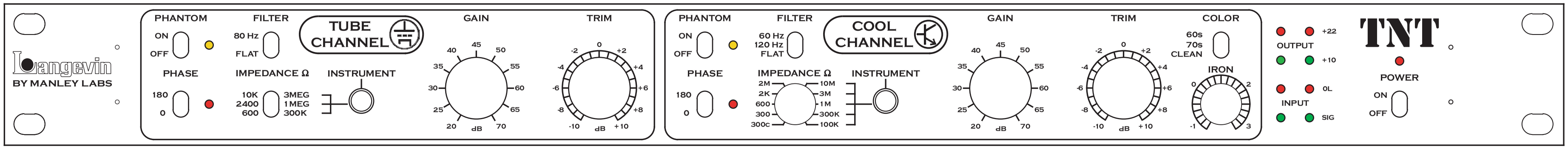
GENERAL

POWER REQUIREMENTS		30 WATTS
FUSE	*Serial #MTNT142 and above:*	For 120 VAC - 0.5 AMP SLO-BLO
		For 220 VAC - 0.250 AMP SLO-BLO
	Serial #MTNT141 and below:	For 120 VAC - 1 AMP SLO-BLO
		For 220 VAC - 0.5 AMP SLO-BLO

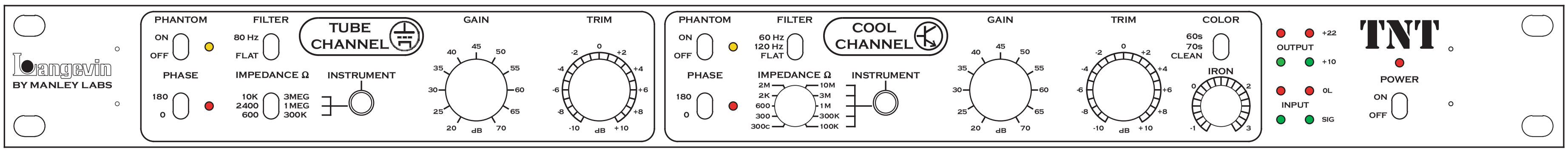
PHYSICAL

SIZE	19" x 11" x 1.75"
WEIGHT	11 POUNDS
SHIPPING BOX	22.5" x 14.5" x 8"
SHIPPING WEIGHT	16 POUNDS

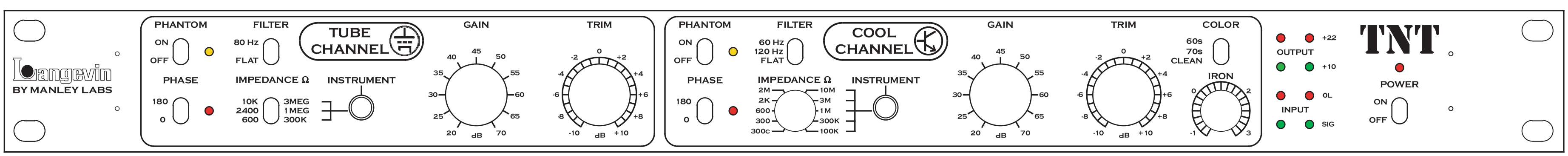
MANLEY TNT TEMPLATE FOR STORING SESSION NOTES



ARTIST	SONG	PRODUCER	ENGINEER	DATE
INSTRUMENT	TRACK	INSTRUMENT	TRACK	
MICROPHONE	NOTES	MICROPHONE	NOTES	
POSITION		POSITION	BACK PANEL OP MODE	UNBAL +4 **BAL +4** UNBAL -10



ARTIST	SONG	PRODUCER	ENGINEER	DATE
INSTRUMENT	TRACK	INSTRUMENT	TRACK	
MICROPHONE	NOTES	MICROPHONE	NOTES	
POSITION		POSITION	BACK PANEL OP MODE	UNBAL +4 **BAL +4** UNBAL -10



ARTIST	SONG	PRODUCER	ENGINEER	DATE
INSTRUMENT	TRACK	INSTRUMENT	TRACK	
MICROPHONE	NOTES	MICROPHONE	NOTES	
POSITION		POSITION	BACK PANEL OP MODE	UNBAL +4 **BAL +4** UNBAL -10