THE THERMIONIC VALVE AND ITS USES FOR LOW POWERED GUITAR AMPLIFICATION

MICHAEL JOHN KEITH AITCHISON

Supervising Tutors: Steve Fenton and Dr. Jonathan Wakefield.

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Abstract

In accordance with modern day demands for reduced sound pressure levels in recording and live performance situations, this thesis describes researched methods for accurately reproducing the full power 'valve sound' at low power. The output harmonic weighting of the valve sound is examined via an array of plots depicting individual harmonic amplitude throughout the dynamic range of an ECC83 dual triode. Comparative subjective analysis of a referenced high-voltage solid-state preamplifier, with acclaims of large signal capabilities comparable to a valve amplifier, and the ECC83 topology, demonstrates that the weighting of high order harmonics is not only audible but can ultimately determine preference of certain distortion characteristics. Using a reduction in triode anode voltage in parallel with careful management of input signal amplitude, a harmonically accurate reproduction of the 'valve sound' is achieved at a power reduction factor of 9.796. Dynamic data of the ECC83s overload region is collated and included in the thesis as a reference for future digital dynamic modelling. The use of ECC83 triodes at anode voltages below $110V_{DC}$ is shown to have perceivable detrimental effect upon output harmonic content and consequently for accurate production of the valve sound below this threshold the thesis concludes that valve modelling is required.

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Chapter 1 - Introduction

Many would suggest that the thermionic valve or vacuum tube with its regions of space charge, glowing filaments, miller capacitances and non linearity has become a thing of the past, far surpassed in all aspects of performance, quality, tolerance, and applicable uses by solid-state technologies. Indeed in practically all boundaries of modern day research and development this is the case, but for one exception, the field of audio. There remains a devout camp of audio engineers and enthusiasts in high fidelity reproduction of audio, music production, and particularly guitar amplification who still stand to be convinced that the infamous tonal characteristics of a valve amplifier will become obsolete, replaced by modern day analogue electronics or digital signal processors.

Whilst there can be seen to be definite advantages of leaving things in the past, this thesis is more sympathetic towards the philosophy of looking backwards in order to move forwards. Much in the way Noetic Sciences are currently re-inventing the understandings of long proven fundamentals of science, it is the authors belief that using the more advanced objective analysis techniques available today may expound theories from research previously carried out in to the characteristics of the 'valve sound'.

A particular area of current prevalence for valve electronics in guitar amplification is that of power scaling. With the vast advancements in public announcement (PA) system technology and the onset of the 'home' recording studio, there has become a large increase in demand for the application of low sound pressure levels at source in both professional and consumer usage. As such, the high output valve topologies of old, designed with stadium levels of amplification in mind, are becoming scrutinised for their lack of diversity in their range of output. Methods are being introduced to reduce the power output of said topologies but mainly at the risk of losing some of the subjective favourability of the 'valve sound'. Consequently this thesis researches directly in to what the effects of power scaling in valve amplification are and how the objective reductions in power translate subjectively to the human ear.

Despite valve technology being the focus of this disquisition, research has been carried out in to solid-state and digital modelling to provide a balanced argument between the two fields of thought.

1.1 - Research Perspective:

It must be clearly identified from the onset of this body of work that high-fidelity reproduction of signals is not the ultimate goal of this research. This thesis, as its title entails, is aimed at guitarists and guitar amplifier engineers with the understanding that a guitar amplifier is an instrument in its own right.

"The classic works treat distortion as a detrimental quality to be minimized as much as possible. For guitar amplifiers, however, distortion is something to be embraced and optimally exploited." [1]

This is not to say that this thesis dismisses all high-fidelity research, on the contrary it embraces it and its quest to eliminate distortions with the understanding that in order to eliminate a particular audio property one must have first isolated the cause of the effect.

1.2 - Thesis Objectives:

- Use modern day objective analysis to expound current researched theories about what the 'valve sound' is, and what valve properties produce it. This involves conducting a literary review of all relevant and to-date research, defining what characteristics are synonymous with the valve sound, and highlighting the objective measurements needed to display these properties. Once defined and highlighted the modern test equipment available at the University of Huddersfield will be utilised to produce an altogether more accurate picture of the 'valve sound'.

- Produce a wealth of data detailing the dynamic operation of the triode valve and its response to a range of increasing amplitude input signals. The digital test equipment available today allows all measured data to be recorded, collated and organised into large data sheets, very precisely detailing all the significant and not so significant values from each test set.

- Extend this data accumulation to examine the effect on the dynamic operation of a valve when reducing the High Tension (HT) supply to the anode of a triode. Furthering the second objective, and using the same methodology, the data accumulation for increasing amplitude input signals is expanded to include varying HT supplies from 300-50V_{DC}. All the data is then organised into a format accessible by all and any interested in the dynamic operation of a triode.

- Determine if the thermionic valve can be used in low power guitar amplification. By manipulating the data obtained, plots can be derived displaying the individual harmonics and their reactance to set independent variables. The correlations between these harmonic relationships can then be discussed and conclusions can be drawn as to whether or not the 'valve sound' is achievable at reduced power levels.

- Determine the required advancements in valve modelling needed to produce accurate valve simulation. The above objectives will be completed for a valve simulator as well as an actual valve. The resultant data sets and plots can be compared to determine what factors of the 'valve sound' are already accurately modelled, and which areas, if any, need improvement. The data accumulation should allow educated theories to be drawn as to which direction valve electronics needs to take in order to continue as a commercially viable option for all aspects of audio.

1.3 - Thesis Methodology:

This thesis presents a literature review of existing research into valve electronics. Using a wide range of references the current perceptions of valve tonal characteristics are determined, and explanation is provided as to what valve operating parameters are responsible for these distinctive traits.

Methods of power reduction of the 'valve sound' are discussed throughout the text, and the alternative of valve modelling is deemed to be the most likely candidate to accurately simulate a valve topology with further acquisition of dynamic data.

The effects of anode reduction on output harmonic content are closely scrutinised, and theories are postulated as to achievable power scaling factors using the researched technique of controlling input signal amplitudes in parallel with anode voltage dropping.

Conclusions are drawn and the thesis objectives are satisfied.

Chapter 2 - The Valve Sound

This chapter will introduce the fundamental areas of valve electronics in order to ascertain an understanding of what the valve sound is and why many audio enthusiasts perceive the valve as producing a favourable and pleasant reproduction of signals. It will describe the principles of what is currently thought to be the key characteristics of the valve sound, describe how the operation of the valve helps introduce these characteristics, and ultimately demonstrate what to test for to objectively display the properties of a valve sound and help qualify the commonly associated subjective terms.

2.1 - Valve Fundamentals:

Prior to explaining what is thought to be the characteristics of the 'valve sound' it seems prudent to examine the construction of the valve and analyse its dynamic properties. In understanding the valves inner workings the author hopes to explain what causes said characteristics, and highlight areas to be researched into in relation to possible power reduction of the overall system.

The valves origins can be traced back as far as 1880 to Thomas Edisons' research into increasing the longevity of the incandescent light bulb. [2] The phenomenon known to date as the 'Edison' effect, the release of electrons from a metal surface when heated, provided the fundamental physics behind the development of the modern world of electronics. John Flemming exploited this principle in 1904 duly noting that when placing two carbon filaments within an evacuated envelope, heating one, leaving the other cold, current flowed in only one direction when an AC source was connected across the two electrodes. Described in his Patent No. 24850 as "a two-electrode valve for the rectification of high-frequency alternating currents", [3] the British named 'valve', as opposed to the American equivalent 'tube' was born.

Valves, as we know them today, have evolved greatly from their diode heritage. Indeed it took only until 1908 before Lee de Forest inserted a "grid shaped member α" [4] between the two electrodes now named as the filament (the heated one), and the plate (the cold one). By altering the voltage on the grid de Forest could control the rate of flow of electrons between the filament and plate, the fundamental principle behind the amplification of signals. Originally documented as the audion, this three electrode valve is the direct predecessor of perhaps the most commonly used audio amplifier valve, the triode, which ultimately led to the applied principles in the development of transistors, from which practically all modern day consumer technology owes its existence to.

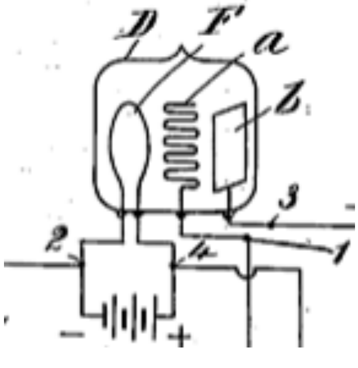


Figure 2.1 – De Forests Audion. [4]

2.1.1- The Triode:

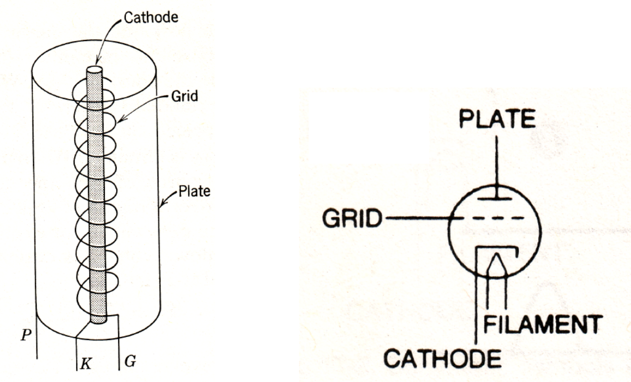


Figure 2.2 – Triode construction and circuit symbol. [20] [21]

The triode, as the name suggests, consists of three electrodes in an evacuated envelope; the anode (plate), the grid, and the cathode. A large positive High Tension (HT) DC voltage supply is connected to the anode via an anode resistor creating a large potential difference between it and the cathode; held at either ground or significantly lower potential dependant upon bias configuration. The cathode element is heated to incite electron emission and the negatively charged particles are attracted to the anode. The grid is inserted between the two electrodes and held at a certain DC bias voltage which controls the flow of electrons through the vacuum, restricting the flow with a negative cathode-grid voltage, or occasionally increasing it with a slight positive cathode-grid voltage.

Due to their simplicity in design triodes provide relatively low noise amplification of small signals making them a good candidate for pre-amplifier stages, increasing the low levels of instruments up to useable voltages for amplifier systems. Note the use of the phrase “a good candidate” and not the ideal. The fact of the matter is that triodes are indeed the most commonly used valve type for audio pre-amplifiers, particularly guitar pre-amplifiers, where the ECC83/12AX7 dual triode design is the most prevalent [6] [13], not necessarily because they produce the best performance but because they are the cheapest and most widely available valve. Either way the triode provides the backbone for most valve topologies and as such cannot be ignored.

As previously stated the triode has three electrodes, each of which has specific physical properties and individual characteristics that need to be understood. The defining properties of these three electrodes ultimately determine the operating performance limitations and primary parameters for all valve characteristics and so must be researched and explained.

2.1.2- The Cathode:

The cathode is the element within a valve that determines how much current is available for the process of amplification and the aforementioned Edison effect lies at the core of its operation. In order for the Edison effect to take place one essential element must be provided, heat.

Heat is applied to the cathode either directly or indirectly, though more commonly indirectly as will be explained. The heat provides electrons within the cathode material with thermal energy. Each electron requires enough energy to reach a certain escape velocity at which point the cathode releases or emits one negatively charged electron from its surface into the surrounding vacuum. Should this release take place in anything other than a vacuum then the electron would immediately be in contact with air particles and ions negating its charge. The *emissivity* of a cathode is defined as "the number of amperes produced per square meter of emitting surface per watt of applied filament power." [7] The larger the emissivity of a material; the more electrons it can release.

The minimum amount of energy needed for an electron to be able to escape is known as the *work function*, and differs dependant upon material type. [8] Appendix 8.1 shows a table of elements and their known emissivity. Tungsten, chemical symbol W, was the original choice for filament material due to it being the electrical conductor with the highest melting point, 3695K, [9] when directly heated. Research is still continued to date on the emissivity of Tungsten for use in filament light bulbs [10]. The value of emissivity is "difficult to predict" and filaments of the same specification can have "very different emissivity behaviour," [10] two properties that do not lend themselves well to controlled environments such as audio amplification.

$$I_{th} = A\, D_0\, T^2 exp(-b_0 / T)$$

The Richardson-Dushman equation. (1)

Where:

A = surface (emitting) area of the filament, m^2.

D_0 = material constant for a given filament type, $A/m^2 \cdot K^2$.

T = absolute temperature, K.

Exp = base of natural logarithms

b_0 = 11, 600 E_w, K.

E_w = the work function of the emitting surface, eV. [7]

The Richardson-Dushman equation (1) allows calculations of thermionic emission current I_{th} and defines its dependence upon the operating temperature of the filament and the emissivity of the material. Research into this area, by Irving Langmuir in 1923 [11], resulted in a large power reduction of valve electronics when thorium was added to the tungsten to give thoriated-tungsten. This increased the emissivity of the filament as well as reducing the materials work function (necessary thermal energy). Efficiency of the thermionic valve was increased by a factor of ten by the introduction of this chemical compound. [9]

Research continues to be carried out into this area for both industrial and scientific advancement such as welding rod efficiency, and NASAs development of high-power microwave communication systems [12], another commercial area where valve electronics are still utilised. There has supposedly been "a 10-percent-per-year growth in demand for tubes used in MI amplifiers and high end audio devices since the late 1980s" [13] with "vacuum-thermionic devices holding sway over the US $100 million worldwide guitar amp business." There could well be a new breed of developed audio valves that take advantage of current research into thermionic emission. As yet there are none, and unfortunately the resources are not available to the author to allow research along these lines at Masters level.

Heated filaments tend only to be used in high-power applications, where large voltages and operating temperatures are needed. They prove detrimental to low powered audio devices due to inherent noise caused by three factors:

- Thermal instabilities causing hum at twice the mains frequency.
- Electrostatic imperfections from the filaments positive and negative peaks of its AC supply producing hum at twice the mains frequency.
- Electrons missing the anode due to a curved flight as a consequence of the electromagnetic field created by the changing polarity within the filament causing hum at mains frequency.

[9]

Lower power valve topologies take advantage of another development, indirectly heated cathodes. Essentially this involves taking the heated element of the filament and coating it in "a ceramic insulating material to isolate the heater electrically from the cathode." [7] Electrical isolation immediately resolved the issues of induced hum, ideal for low signal amplification. It also allowed further exploitation of chemistry.

By combining elements with oxygen to create oxides, lower work functions were achievable, again increasing efficiency and lowering power consumption. The cathode became split in to two elements, the cathode itself, and the heater. The heater material was originally changed from tungsten to the cheaper and more durable nickel, though in more recent times nichrome, an alloy of 80% nickel, and 20% chromium, has become the common heating element because of its improved operating parameters of higher melting point and greater resistance, requiring less power to achieve operating temperatures.

A great deal of power reduction can be achieved with experimentation of the cathode and heater materials, unfortunately cost is a major preventing factor to the use of these exotic materials in the majority of commercially available audio valves. As such, the relevance of researching further into this area is negated. Only with a vast increase in demand for new valves would such research become viable.

2.1.3- The Anode:

The negatively charged electrons emitted form the cathode build up to form a cloud in the evacuated envelope called a space charge. Without the addition of a positive attraction this space charge would remain and current would be unable to flow. The anode provides such an attraction, held at high positive voltages in relation to the cathode its positive charge attracts the negative electrons creating a flow of current through the evacuated chamber, accelerating electrons from the space cloud towards it.

$$\textbf{\textit{Electron velocity}} = \sqrt{(2V(e / m_e))}$$

Electron velocity equation. (2)

Where:

V = accelerating voltage, anode voltage in this case.

e / m_e = electron charge/mass ratio ≈ 1.7588×10^{11} C/Kg.

[9]

The speed at which electrons are accelerated is directly proportional to the voltage applied to the anode. This speed can be calculated using the electron velocity equation (2), which highlights the incredible amounts of energy at work within the system. An average anode voltage for a preamplifier ECC83 valve would be approximately $200V_{DC}$. Using equation (2) shows that electrons within the vacuum will reach speeds of:

$$\text{Electron velocity} = \sqrt{(400(1.7588 \times 10^{11})}$$
$$\text{Electron velocity} = \underline{8.388 \times 10^{6} \text{ m/s}}$$

This is approximately eighteen million miles per hour, and with the anode being continuously bombarded by electrons the transference from kinetic energy to thermal energy has a huge effect upon the anodes operating characteristics. Indeed one parameter that must never be exceeded within a valve is the anodes power dissipation. The transfer of kinetic energy to electrical energy produces thermal energy, one of the large factors in valve inefficiency. An anode must be able to dissipate the heat effectively to prevent overheating. An overheated anode causes outgassing; the release of gas into the evacuated chamber interferes with the electron flow as electrons collide with the gas molecules to form positively charged ions. This leads to an unwanted sensation known as ionisation noise.

“Many effects within valves can be understood by having an appreciation of the collision velocity of the electrons as they hit the anode.” [9]

Although useful for demonstrating the high kinetic energy levels of the accelerated electrons, the equation (2) has its flaw. Voltages used in large power valves for distribution grids will average 512kV, which according to the electron velocity equation would accelerate electrons to faster than the speed of light, a known physical impossibility. The error in the equation is in the electron charge/mass ratio, which only accounts for the *rest* mass of an electron.

Gallileo and Newton define mass as "that property of a body that governs its acceleration when acted on by a force." [14] Thus as acceleration increases so does a particles mass, never allowing enough force, in this case anode voltage, to be available to reach the speed of light. In order to calculate true electron velocity, taking in to consideration the increasing electron mass, the Alley-Atwood equation (3) must be used:

$$\text{velocity} = c \cdot \sqrt{1 - \frac{1}{\left(1 + \frac{e}{m_e} \cdot \frac{V}{c^2}\right)^2}}$$

Alley-Atwood equation. (3)

Where:

c = velocity of light in a vacuum ≈ 2.998 x 10^8 m/s.

[15]

So at $200V_{DC}$ anode voltage the actual electron velocity is equal to:

Electron velocity = 2.998 x 10^8 (√(1-(1/1.00078)))

Electron velocity = (2.998 x 10^8) x 0.0279

Electron velocity = <u>8.385 x 10^6 m/s =100v</u>

"Note that the distance between the anode and cathode does not feature in either equation because an infinite distance would also allow an infinite time for acceleration, and even if the rate of acceleration was very low, the collision velocity would still be reached." [9]

To relate these equations to the thesis title of power reduction we can use simple kinetic energy theory, in tandem with the Alley-Atwood equation to determine how a change in anode voltage will affect the output power of the valve. Remember that the Alley-Atwood equation has already taken into consideration the reduction in mass of an accelerated electron; therefore the calculated velocity already considers the rest mass energy, which is significantly larger than the electrons kinetic energy. [16]

$$\mathbf{KE = \tfrac{1}{2}.mv^2}$$

Kinetic Energy Equation (4)

Upon collision with the anode surface the kinetic energy is converted to potential energy, calculated in Joules. Audio applications most commonly use Watts as a measure of power, Watts being the energy usage per unit time, or Joules per second.

The ratio between the two units of measurement is therefore directly proportional and as such the percentage reduction of power can be postulated by the percentage reduction in kinetic energy of an electron accelerated by reducing anode voltages.

Kinetic Energy of electrons at 200V_{DC} HT supply:

m = rest mass of an electron = 9.109×10^{-31} kg.

v = electron velocity in a vacuum using Alley-Atwood equation (mass reduction considered).

$$KE = \tfrac{1}{2}.mv^2$$
$$KE = \tfrac{1}{2} \times (9.109 \times 10^{-31}) \times (8.385 \times 10^{6})^2$$
$$KE = \underline{3.202301767 \times 10^{-17}\ J}$$

Kinetic Energy of electrons at 100V_{DC} HT supply:

$$\text{Electron velocity} = 2.998 \times 10^{8} (\sqrt{(1-(1/1.00039))})$$
$$\text{Electron velocity} = (2.998 \times 10^{8}) \times 0.0198$$
$$\text{Electron velocity} = \underline{5.930 \times 10^{6}\ m/s}$$

$$KE = \tfrac{1}{2}.mv^2$$
$$KE = \tfrac{1}{2} \times (9.109 \times 10^{-31}) \times (4.193 \times 10^{6})^2$$
$$KE = \underline{1.601620735 \times 10^{-17}\ J}$$

Percentage reduction of halving the supply voltage:

$$\text{Percentage} = 100 \times ((1.601620735 \times 10^{-17}) / (3.202301767 \times 10^{-17}))$$
$$= 50.01467231\%$$

$$\text{Percentage reduction} = 100 - 50.01467231$$

<u>**Percentage reduction = 49.9853%**</u>

Kinetic Energy of electrons at $50V_{DC}$ HT supply:

$$\text{Electron velocity} = 2.998 \times 10^{8} (\sqrt{(1-(1/1.000196))})$$
$$\text{Electron velocity} = (2.998 \times 10^{8}) \times 0.0139$$
$$\text{Electron velocity} = \underline{4.193 \times 10^{6}\ \text{m/s}}$$

$$KE = \tfrac{1}{2}.mv^{2}$$
$$KE = \tfrac{1}{2} \times (9.109 \times 10^{-31}) \times (4.193 \times 10^{6})^{2}$$
$$KE = \underline{8.009278759 \times 10^{-18}\ \text{J}}$$

Percentage reduction of further halving the supply voltage:

$$\text{Percentage} = 100 \times ((8.009278759 \times 10^{-18}) / (1.601620735 \times 10^{-17})$$
$$= 50.00733684\%$$

$$\text{Percentage reduction} = 100 - 50.00733684$$

<u>Percentage reduction = 49.9927%</u>

This result shows an *instantaneous* value of power reduction by 50% of a *single* electrons conversion from kinetic energy to electrical energy in Joules. However, as calculated, the electron velocity decreases with anode voltage, and as such the number of electrons bombarding the anode per second, ie. the current flow, is also reduced. So despite the *instantaneous* drop in power of 50% the *actual* power is reduced exponentially. Simple analysis of the fundamental law of Physics P=VI demonstrates this, in that an instantaneous halving of voltage will result in an instantaneous halving in power, but as the voltage drops, the current drops proportionately, and the power is reduced further.

Fortunately to save on calculation of this exponential curve, the relationship between anode voltage and anode current has been plotted by valve manufacturers to produce arguably the most useful set of curves for valve operation analysis, the anode characteristics (2.1.5). There is however one other variable plotted on the anode characteristics that must be taken into consideration, possibly the most important electrode of the three, the grid and its bias voltage.

2.1.4- The Grid:

The grid is the controlling electrode of the amplification process. Input ac signals fluctuate about a set grid voltage bias point controlling the flow of electrons through the evacuated envelope. Without the grid a valve would be constantly 'turned on', the direct opposite of transistor operation, with electrons flowing unimpeded from cathode to anode. Under normal operation the grid is held at a negative potential in relation to the cathode and is placed between anode and cathode, usually closer to the cathode where the electron velocity is lower.

As the negatively charged space cloud electrons, emitted from the cathode, are accelerated towards the anode they encounter the grids slight negative charge. Although the negative charge on the grid is not enough to overpower the anodes greatly significant positive charge, it has the effect of slowing the amount of charge passing through the valve and thus reducing the flow of current; Current being the flow of charge passing a certain point per second, Coulombs (Q) per second, measured in Amperes (A).

The more negative the grid voltage swings the more the flow of charge is impeded and less current flows. In contrast, should the grid swing into positive voltages the electrons are aided in their acceleration towards the anode and current flow increases. However, should the grid voltage become too positive it can begin to attract and collect electrons from the cathode causing grid leak current, a not all together unwanted phenomena as indicated by Rutt [17], but if not controlled can ultimately lead to the valves destruction. Grid current will be discussed in more detail in the ensuing sections.

Like the cathode and anode the grid has a parameter that can significantly alter the power output of a valve, it's positioning. A very small change in the positioning of the grid in relation to the cathode can have a significant affect upon the flow of electrons; Increased grid/cathode distance gives greater output, decreased grid/cathode distance gives reduced output. This, like the cathode material, is another parameter which researching into would prove futile because of the large costs involved in constructing one off valves.

The grid bias voltage is of utmost importance in valve amplification because it sets the operating point for the whole system. The following section 2.1.5 will describe the effects of changing bias voltages, but for now it is important to understand there are two ways of achieving the bias voltage.

The first of these methods is named *fixed* bias. It is achieved by taking a tapping off the mains transformer and rectifying the AC voltage using diodes, then smoothing the resulting half AC waveform using capacitors to produce a negative DC voltage supply. This supply, set using potential dividers, is applied directly to the grid of the valve. It is not uncommon to use a potentiometer in the bias circuitry to allow small adjustments of the voltage to be made.

Fixed bias can be seen as a disadvantage due to its unsympathetic nature to valve deterioration with time, constantly driving the valve at its *specified* optimum operating point as opposed to its *actual* optimum. However when valve deterioration is not a problem, for example when using new valves, fixed bias has the advantage of allowing maximum cathode current to be drawn and as such is often used in output stages of amplifiers. The imperfections in the supply can however "cause noise and instability problems with small-signal stages." [9]

The unimpeded current flow produces less distortion of a signal when approaching clipping by maintaining a constant potential difference between the cathode and the grid. The alternate method, *cathode* biasing, may not have this advantage but does indeed have others. Cathode bias is achieved by placing a resistor in series with the cathode, figure 2.3 demonstrates:

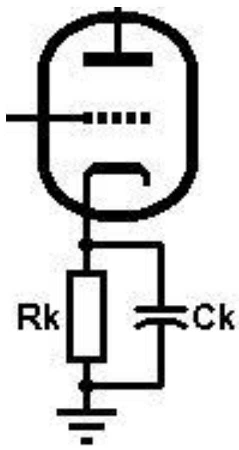


Figure 2.3 – Cathode bias using a cathode resistor and decoupling capacitor.

Initially no current flows through the valve, and so by definition (Ohms law) no voltage is dropped across Rk and the cathode voltage remains at 0V. In this configuration the grid is no longer supplied with a bias voltage, it is tied to ground, 0V, meaning that the grid to cathode voltage V_{gk} must be 0V. As stated previously the grid must have a negative voltage in relation to the cathode to 'turn off' the flow of current. Therefore the flow of current through the valve increases, drawing current through the cathode resistor causing a voltage drop across it. "This voltage drop causes the cathode voltage to rise, V_{gk} falls, and an equilibrium anode current is reached." [9] This is commonly known as self-bias and is advantageous in being sympathetic to valve deterioration. Intrinsically the value of the cathode resistor will determine the bias point for valve operation. Anode voltage and current are known parameters defined by the anode characteristics plots (Section 2.1.5) and simple manipulation of Ohms law (V=IR) allows the value to be calculated to achieve specific values of bias voltage.

The biggest appeal of cathode biasing comes in stark contrast to the advantage of fixed biasing. As the amplifier reaches the point of clipping the added resistance in the cathode line limits the amount of available current. This results in the largest peaks of the input signal being restricted in amplification by the current limitation, effectively compressing the output signal, an effect often thought to be subjectively pleasing.

$$\mathbf{C_k = 1 / (2\pi f R_k)} \tag{5}$$

For 'normal' or 'full power' operation the cathode capacitor Ck in figure 2.3 should have low enough reactance to provide a short circuit to all ac frequencies of interest. This cut off frequency can be calculated using the well-known filter formula (5), but has no real relevance to this thesis and so no further explanation is included. The interest in this component lies in its affect on the power of the valve when removed from the circuit.

Output signal from a valve is derived from the changing current flow through the anode and its load resistor. The changing current is also drawn through the cathode resistor and therefore a signal voltage must develop across it. This signal voltage will be in phase with the input signal voltage, effectively reducing the voltage difference between the grid and the cathode, the bias voltage, and ultimately the driving voltage of the valve.

This effect is known as *negative feedback*, and applied negative feedback is known to reduce open loop gain to a considerably lower closed loop gain value [18] resulting in a power output loss that can be calculated using the universal feedback equation (6).

$$\boldsymbol{A_{fbk} = A_0 / (1 + \beta . A_0)}$$

Universal feedback equation. (6)

Where,

β = feedback fraction = R_k/R_L [9]

"The feedback is series derived, and series applied, so it raises the input and output resistances." [9] This increase is negligible compared to the valves virtually infinite input resistance, but has a significant affect upon the internal anode resistance which, in series with the anode load resistor, increases its overall output resistance.

An increase in output resistance of the valve can be seen as an increase in source resistance to the load of the next stage. The transference of power between stages is affected by the ratio of transfer (η), calculated using equation (7):

$$\eta = \frac{R_{\text{load}}}{R_{\text{load}} + R_{\text{source}}} = \frac{1}{1 + \frac{R_{\text{source}}}{R_{\text{load}}}}$$

Ratio of power transfer. (7)

[19]

By increasing the source resistance we reduce the ratio of power transfer between stages and ultimately the output power of the system.

The cathode capacitor is included in almost all audio amplifier stages utilising cathode bias because it opens the closed loop created by the cathode resistor, allowing an unimpeded route to ground for any ac signals. This prevents the voltage across the cathode resistor from varying with input signal and consequently prevents the power reduction, usually an unwanted factor.

It would be possible to place a single pole single throw (SPST) switch in series with this capacitor in a valve topology to allow reductions in gain and power at the flick of a switch. However, switching out said decoupling capacitor would also impede all audio frequencies through the cathode resistor as opposed to just the ones below the set cut off frequency, ultimately resulting in a perceived 'dulling' of the tonal characteristics; not something satisfactory for guitar or audio amplification, and essentially the reason this capacitor is included in the first place. As such this method of power reduction is deemed too detrimental to output sound to warrant further research.

Looking at the three electrodes present in a triode has highlighted four directions in which research could be conducted with an ultimate objective of reducing amplifier power output:

- Cathode material emissivity properties.
- Anode voltage reductions.
- Grid to cathode distance.
- Cathode capacitor switching.

Of these four areas reducing the anode voltage can be seen to have the largest power reduction. It is also the most feasible direction to research at Masters level, and as such the effects of anode voltage reduction will be analysed in greater detail. The theoretical implications of anode voltage dropping can be indicated using the anode characteristics.

2.1.5- Anode characteristics:

As articulated, the anode characteristics are perhaps the most useful plots for determining the operating parameters of a valve. Depicting the relationship between anode voltage and anode current for a range of varying bias voltages in graphical form allows optimum bias voltage, maximum operating conditions, and the three primary parameters of a valve to be calculated with relative ease.

To fully appreciate the anode characteristics one further line must be plotted. This line, named the *load line*, is a user variable parameter and is determined by the value of load resistor chosen to place in series with the anode. The load resistor is the key component in the amplification stage, converting the changes in current flow through the valve, controlled by the signal on the grid, into an alternating voltage drop across said resistor. In this way a small voltage swing on the resistor is amplified to a large, 180° out of phase, voltage swing across the anode. All single stage valve amplifiers are inverting amplifiers.

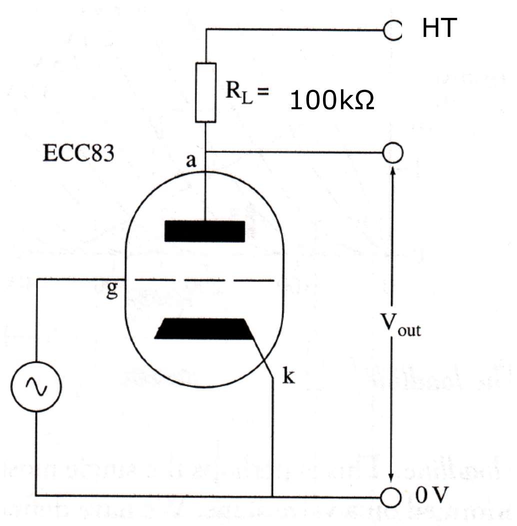


Figure 2.4 – Triode connected in circuit with anode load resistor. [9]

Using the load resistor value of 100kΩ in figure 2.4 as an example we can determine our load line for diminishing anode voltages of $300V_{DC}$, $200V_{DC}$, and $100V_{DC}$. With no anode current no current flows through R_L and so the full HT supply is dropped across the valve. With no voltage on the anode the full HT is dropped across R_L and the current can be calculated using Ohms law (I = V/R) to be 3mA. Voltage and current are directly proportional so these two points can be connected by a straight line with equation y = mx + b, with b directly affected by anode voltage as shown in figure 2.5.

Figure 2.5 – Anode characteristics of 12AX7 with load lines for 100V, 200V and 300V HT.

With the load line drawn an operating or bias point needs to be selected at one of the intersections between the load line and the gird voltage curves. This point ultimately governs the overall output performance of the valve stage, with particular impact on the amplifiers linearity. As the load line converges upon the maximum HT the distance between grid lines becomes less, demonstrating non-linearity. Contrastingly as the grid voltage becomes more positive the intervals between the intersections become more evenly spaced, more linear.

Distortion in a guitar amplifier is generally accepted as being favourable. “In this application the amplifier becomes part of the musical instrument, and is frequently used to radically alter the signal from the guitar.” [23] For this reason the bias point in guitar preamplifiers is often found towards the non-linear region of the bias curves. The bias point however is governed by a few operating parameters outside of which the valve will produce unwanted distortions as opposed to the distortions exploited for their pleasing characteristics.

Should the engineer attempt to introduce too much non-linearity by biasing the grid at too negative a potential, *over-biasing*, the valve will reach *cut off* as the grid becomes more negative than the electron space-cloud and negates the positive attraction of the anode. The electrons begin to return to the cathode; the flow of current is cut off.

As the grid voltage is increased to positive voltages, *under-biasing*, it turns the valve on harder until no voltage is dropped across it and the full HT is dropped across the load resistor. In reality another problem arises well before this, *positive grid current*. The electrons begin to be attracted to the grid, flowing out through it to ground. The increase in grid current reduces the input impedance of the valve significantly enough to start loading the previous stage, (the input to the amplifier in a preamp stage) resulting in an attenuation of positive input signal peaks.

Figures 2.6-2.8 demonstrate the effects of under bias and over bias on a signal using plots of grid voltage on anode current, the transfer characteristics of a valve, for a "general purpose triode biased for class-A operation." [24] The linear region of operation is depicted by the straight-line segment of the plot.

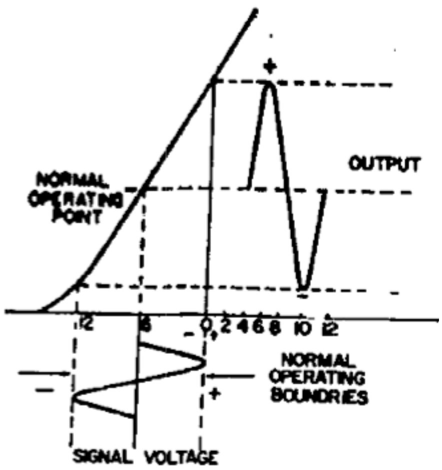


Figure 2.6 – 'Normal' / linear operating point. [24]

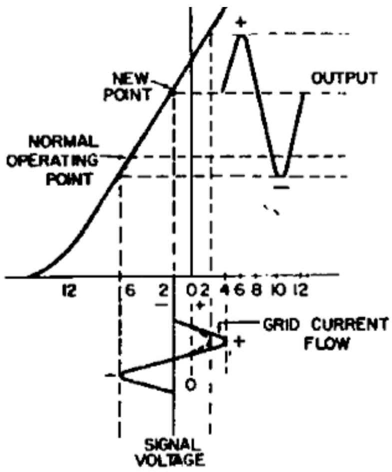


Figure 2.7 – Distortion of the anode circuit waveform caused by under-biasing.

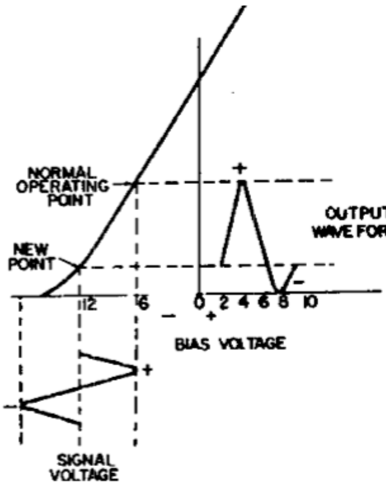
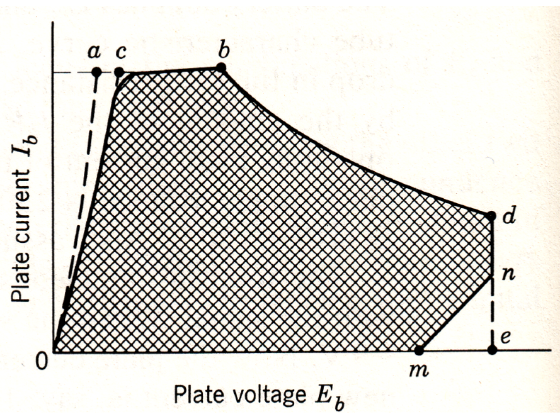


Figure 2.8 – Distortion of the anode circuit waveform caused by over-biasing.

Over biasing, figure 2.8, produces an asymmetric output waveform. This is one key factor in the origin of the pleasing nature of the 'valve sound' as will be discussed.

As explained earlier, the bombardment of the anode by electrons produces thermal energy that must be dissipated to prevent the heating of the other elements within the vacuum. Heating of the grid can cause grid emission, producing a negative current which can increase the grids potential, in turn increasing the anode current, further heating the valve, and further heating the grid. The effect is cyclic and known as *thermal runaway*, producing unwanted noise and resulting in the destruction of the valve.

Consequently three key datasheet parameters, maximum anode current, maximum anode voltage, and maximum anode dissipation must be respected at all times. All five of the discussed limitations can be superimposed upon the anode characteristics to visibly define the acceptable region of operation in which the valve may be biased.



- **0c** = maximum positive grid current.
- **cb** = maximum anode current.
- **bd** = maximum anode dissipation.
- **mn** = maximum negative grid voltage.
- **de** = maximum anode voltage.

Figure 2.9 – Operating region for a triode. [25]

Operation outside of the shaded area in Figure 2.9 will result in damage to a valve, however any point within and the valve will perform adequately. The region encompasses anode voltages right down to the 0V level further adding to the case to study the effect of reducing the anode voltage. In fact it is well documented that "cool" running of a valve vastly increases its longevity, [1] [6] [7] [9] [20] [25] so the argument for reducing anode voltages is strong.

Because the valve manufacturers themselves wrote the majority of literature originally available for valve electronics, all the suggested operating parameters were tailored to achieve optimum linear output. As such the practice of under running valves became somewhat frowned upon by the audio engineers of the 1950s and 60s. [13] Further to that the development of guitar amplifiers now widely considered as some of the 'classic' designs, such as the Marshall Bluesbreaker, the Fender Bandmaster or the Vox AC30, were particularly orientated towards achieving large power output capable of providing a solid back line for live music using large anode voltages to achieve such levels of gain.

It is only recently that a reduction in power has become a wanted trait. Unfortunately even to date some designers still disrepute the anode voltage drop, particularly those using the 12AX7/ECC83 run "at very low plate voltage—as little as 12V." [13]

"This so-called starved-plate operation delivers high distortion, which some equipment designers consider the only useful characteristic of a vacuum tube in audio." [13] However the percentage levels of distortion produced [26] in this manner introduce so many high order harmonics, figure 2.10, that the overall perceived sound is often described as "glassy". This would suggest that there is a limit to just how low the anode voltage can be reduced before becoming destructive to the tonal characteristics of a guitar amplifier.

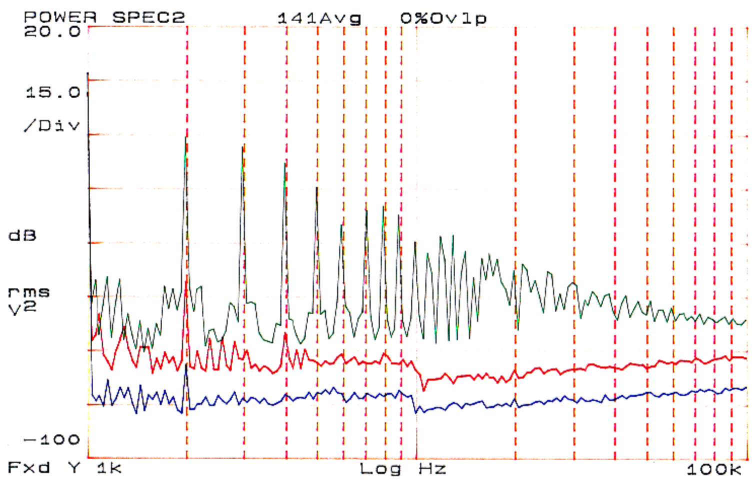


Figure 2.10 – High order harmonics produced by plate-starved 12V 12AX7.

2.1.6- The Primary Valve Parameters:

The final use for the anode characteristics comes in their manipulation to define three primary valve parameters.

AMPLIFICATION FACTOR – $\boldsymbol{\mu = \Delta v_p / \Delta v_g}$

MUTUAL CONDUCTANCE – $\boldsymbol{g_m = \Delta i_p / \Delta v_g}$

DYNAMIC ANODE RESISTANCE – $\boldsymbol{r_a = \Delta v_p / \Delta i_p}$

[7]

The three primary parameters for valve operation. (8)

Where:

Δ = a change in value.

v_p = plate voltage.

v_g = grid voltage.

i_p = plate current. [7]

(NB: lowercase letters are used to denote signal parameters, uppercase letters denote DC values. Hence the above three parameters are all defined at a fixed Q point.)

Each of these parameters is directly affected by the other, highlighting the dynamic nature of the valve, and can be simplified into one very important equation:

$$\boldsymbol{\mu = g_m r_p}$$

Amplification factor for valve amplifiers. (9)

These three AC parameters "define completely the characteristics of a valve, provided that they are evaluated at the operating point." [9] The Amplification factor (μ) is determined by taking a constant quiescent current through the bias point to a 0.5V grid swing either side, see figure 2.11. We can then show the anode voltage swing achieved from a 1V grid change, in other words the gain (Vout / Vin). This gain is however a theoretical one because it assumes that load resistance is infinite, providing constant current. So to achieve a more accurate gain measurement we use the gain formula (10), visibly displayed in figure 2.12.

$$\boldsymbol{A_V = \mu \cdot R_L / (R_L + R_a)}$$

Valve gain formula. (10)

Figure 2.11 – Anode characteristics used to demonstrate the amplification factor.

The amplification factor, despite not giving an accurate gain for a stage is one of the most stable measurable parameters of a valve because it is a "geometric factor dependant upon distances between electrodes." [7] Inclusion of the anode load resistance in equation (10) decreases the *theoretical* gain (μ) to a smaller *actual* gain (Av). Valves are often classified as low μ, medium μ, and high μ, as such the available gain from said valves is often over estimated.

Figure 2.12 – Anode characteristics used to demonstrate the *actual* gain.

Internal anode resistance is another stable measurable parameter determined from the anode characteristics by taking the tangent of the grid curve at the bias point, shown in figure 2.13.

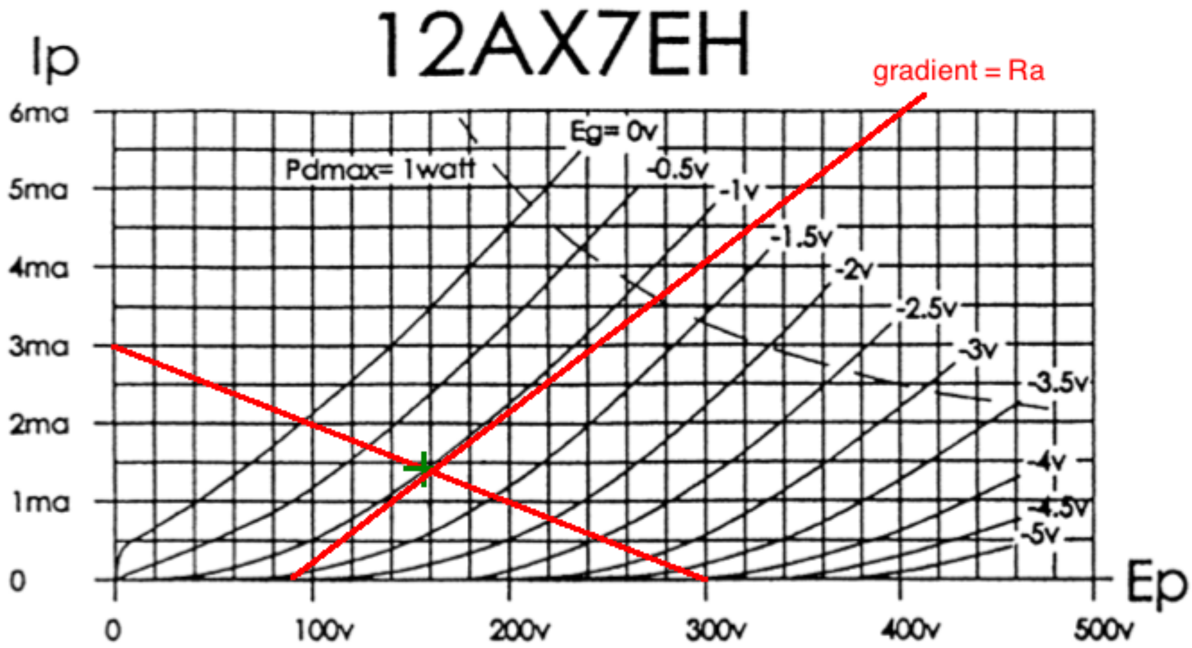
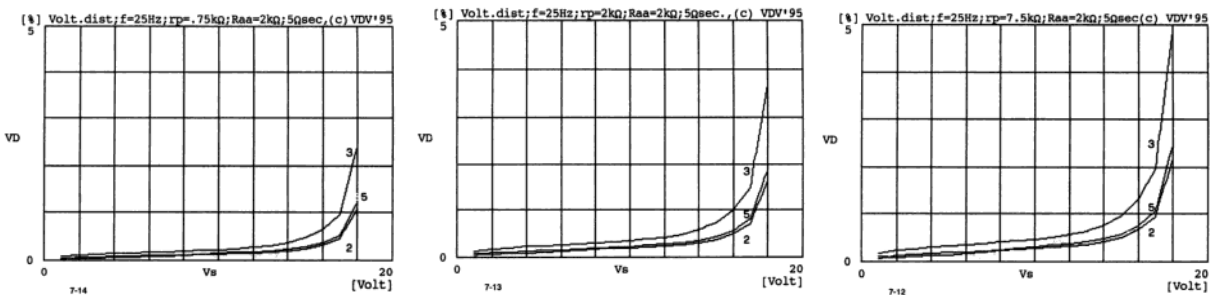


Figure 2.13 - Anode characteristics used to demonstrate calculation of anode resistance.

Menno van der Veens' research demonstrates that an increase in anode resistance actually results in an increase in "unpleasant output distortion," [27] as shown by figure 2.14. The fact that the gradient of the grid curve, the anode resistance, decreases as anode voltage decreases would allude to the fact that in accordance with van der Veens findings lower anode voltage can result in lower distortion. However the effects of anode voltage drop are not exclusive to anode resistance, it also changes the other parameters, mutual conductance being one of the more affected.



R_a = 750Ω R_a = 2kΩ R_a =7.5kΩ

Figure 2.14 – Increased distortion V_D due to increased anode resistance. [27]

Figure 2.15 - Anode characteristics used to demonstrate calculation of mutual conductance.

Mutual conductance (dynamic transconductance being the American terminology) can be determined using the anode characteristics by holding the anode voltage constant and measuring the change in anode current. However, mutual conductance is the ratio between the change in current in the anode and the change in voltage in the grid, two factors that can change through the lifetime of a working valve. Therefore, as opposed to calculating its theoretical value in the described manner, it is often preferred to measure its actual value using a valve tester and is quite often described as a valves measure of its 'goodness'.

Reducing the anode voltage reduces the mutual conductance, as shown in figures 13.2 – 13.5 in Appendix 8.2. The affect of anode voltage reduction on the value of gm is much greater than its affect on anode resistance. Formula (9) demonstrates that if the value of gm is reduced, with μ remaining constant, then r_a actually increases, and so too does distortion. So it could be hypothesised that reducing the anode voltage will result in greater measurable distortion percentages at output, possibly a good effect for guitar amplification.

2.1.7- Miller Capacitance:

One final valve fundamental that must be discussed is 'Miller' capacitance. First reported by John Miller [28] in his paper "Dependence of the input impedance of a three-electrode vacuum tube upon the load in the plate circuit", it defines what he believed to be an "unwanted" internal component present in all valves.

A capacitor is defined in general as "a component consisting essentially of two plates or electrodes separated by a dielectric." [29] A dielectric being "an insulator in which an electric field persists." [29] These two definitions quite clearly depict the parallels between a capacitor and two electrodes within a valve, essentially what a Miller capacitance is; An inherent capacitance between the anode and the control grid which is "reflected into the grid circuit and forms a low-pass filter in conjunction with the output resistance of the preceding stage." [9]

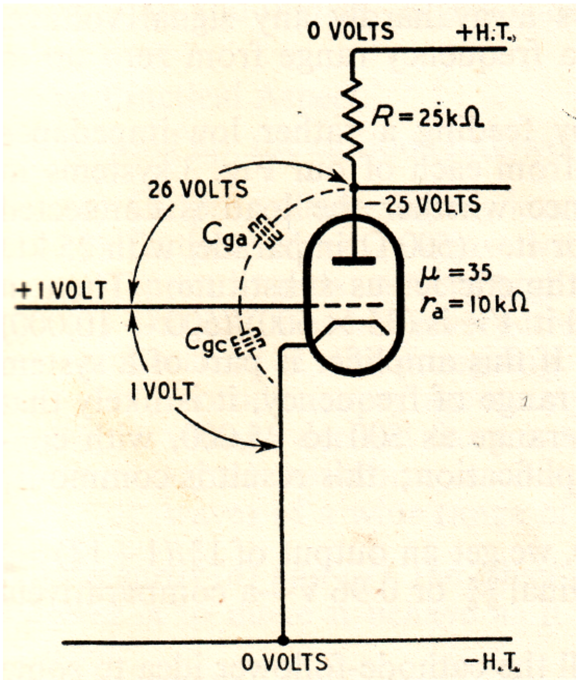


Figure 2.16 – The Miller effect in a triode. [30]

The Miller effect is only inherent when valve stages are cascaded. A simple study of any guitar amplifier schematic will quickly reveal that cascaded stages are always used in guitar preamplifiers. When the anode of the second stage valve amplifies, its changing voltage charges and discharges the anode to grid capacitance, C_{ga} as shown in figure 2.16. Because the grid is high resistance, the charging current must be sunk or sourced by the previous stage. The increased current in the previous stage effectively reduces the following stages 'seen' input impedance, and increases the inter-electrode capacitance.

To use Lundbergs example [31]:

"Consider the following amplifier with voltage gain –*A*, with an impedance of *Z* connected from input to output.

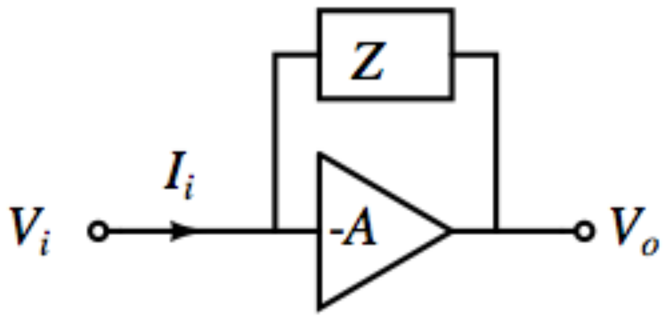


Figure 2.17 – Lundbergs simplified amplifier stage. [31]

Calculating the input current,

$$I_i = \frac{V_i - V_o}{Z} = V_i\left(\frac{1+A}{Z}\right)$$

(11)

The Thévenin* input impedance is,

$$Z_{in} = \frac{V_i}{I_i} = \frac{Z}{1+A}$$

(12)

Thus a resistor or inductor connected from input to output will look a factor of (1 + *A*) times smaller (as seen from the input terminal), and a capacitor will look (1 + *A*) times larger."

The static anode to grid capacitance is often given in data sheets, however these values do not take in to consideration the dynamic Miller effect. The operating capacitance values are therefore significantly larger ((1 + *A*) times larger), having a perceivable effect on the stages frequency response.

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* Thévenin equivalent circuits calculate the total resistance looking into the output terminals, treating whole topologies as a black box. It is a concept used to "break down complex networks quickly and efficiently." [9]

The induced increased capacitance, in series with the reduced input impedance, creates a low-pass filter described as having "an alarming effect on the high frequency response of an amplifier" [9]. Take for example the datasheet value of C_{ag} = 1.6 pF, and average amplification factor (A) = 78 for an ECC83/12AX7, (Datasheet shown in Appendix 8.3) which when placed into the Miller equation (13) results in an actual capacitance of:

$$\mathbf{C_{Miller} = C_{ag}.(1 + \mathit{A})}$$

The Miller Equation (13)

$$C_{Miller} = 1.6 \times 10^{-12} (1 + 78) = \underline{126 \text{ pF}}$$

Taking this value and the datasheet nominal anode resistance of 54.1kΩ, and using equation (5) we get a cut off frequency/-3dB point of 23.34kHz. Any output signal frequencies above this cut off will therefore be attenuated at a roll off rate of 6dB/octave (roll off rate for a first order filter). Although this frequency is above the typical audio frequency range it will result in a reduction of the higher order harmonic content subjectively described as "glassy" in plate-starved operation [26]. This could have a perceivable effect on the resulting tonal quality of the amplifier stage.

Therefore, it can be postulated that the dynamic Miller capacitance has an appreciable effect on the tonal characteristics of a valve stage. It can also be postulated that because the Miller effect is reduced with reduced anode voltages, a reduction in HT supply should result in a higher frequency cut off point, going some way to help explain why starved-plate operation results in such large amounts of high order harmonics being present at the output.

To return to Millers description of an "unwanted" internal component, it could be suggested that the inclusion of more high frequency content at the output, as a result of reduced anode voltage causing reduced Miller capacitance, may well be perceived as a "wanted" phenomena in accordance with Hamms research into the significance of musical harmonics, [32] discussed further in section 2.1. Maybe a point can be found before the "edge harmonics" become too objectionable, at which the reduced anode voltage produces high frequency harmonic amplitudes that results in a "wanted" increased perceived loudness.

The key valve fundamentals have been highlighted and discussed, and the links between power reduction and its affect upon valve electrodes, operating parameters, and inter-electrode capacitances have been analysed. The ensuing section will discuss existing research explaining what the 'valve sound' is, and try to determine what valve operating parameters might be responsible for the so often described subjectively 'warm' pleasing sound that a valve amplifier produces.

2.2 - Existing Research into 'The Valve Sound':

'The valve sound' has become a somewhat infamous terminology in the world of guitar amplification, many guitar enthusiasts having the idea that a valve amplifier is the pinnacle of the guitar sound. Whether this is through personal experience with valve amplifiers, or simply falling into the vast marketing ploy associated with valve electronics it cannot be dismissed that general opinion has the valve sound marked as "warm".

This is perhaps abetted by the fact that "since transistors did not enter wide usage until about 1960, all of the originating styles of rock and blues guitar were developed on tube amplifiers." [13] Some think this could attribute towards "a cognitive bias towards that sound," [33] a sound which undoubtedly has something about it that makes it subjectively pleasing to the human ear. As Hamm notes; "Consider the possibility that the ear's response may be quite different to that of an oscilloscope's." [32]

In actuality the human ear is known to "create harmonics within the cochlea" [34] as such "colouring" perceived sound in a way objective measurements are not capable of showing, figure 2.18. However, objective measurement can be used to try and determine where the subjective "warmth" yields, and knowledge of the complex psycho-acoustical problem of the ear and brains perception of audio lends a hint towards the key lying in the weighting of harmonics within an output signals spectrum.

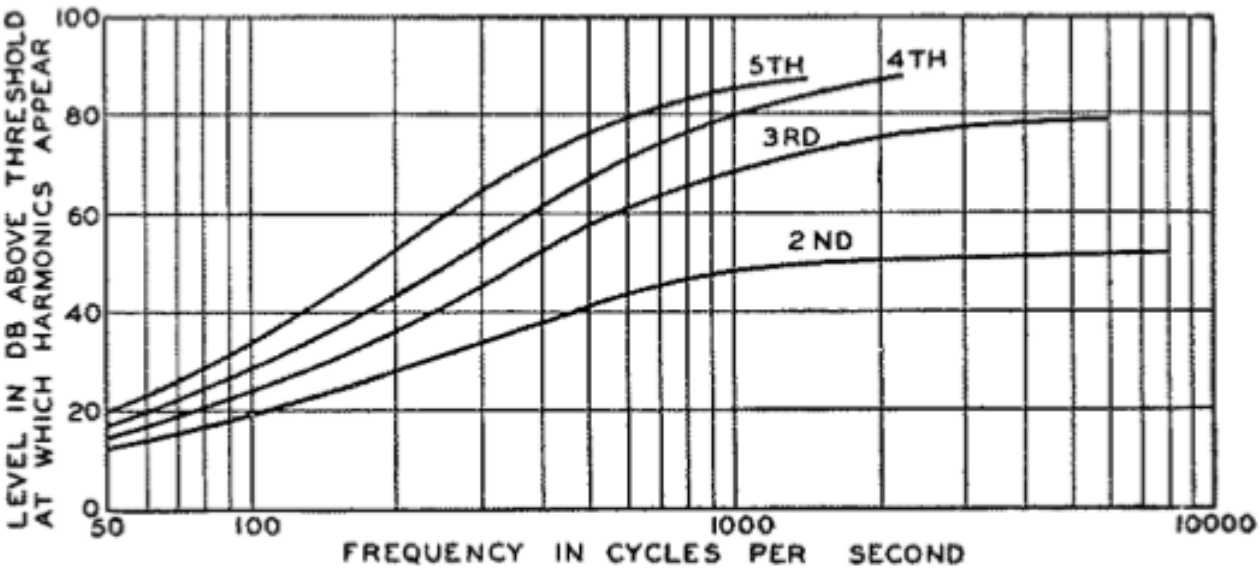

Figure 2.18 – Fletchers plot; Ear self-generated harmonics, frequency versus level. [34]

2.2.1- Determination of the Valve Sound:

Guitar amplifiers can be distinctly split into two sub categories, solid-state, and valve-state. Only with the introduction of transistor based audio electronics was it noticed that the two states yield differing tonal qualities. Russell O. Hamm was the first engineer to measure in "meaningful terms" the differences between the two types by making the key observation that "previous attempts to measure the differences have always assumed linear operation of the test amplifier." [32]

His paper, directed at vacuum tube use in music production, looked at the generated harmonic distortion components of an amplifiers output signal when operated in the non-linear region, concluding that there was an objectively measurable difference when signal transients severely overloaded the amplifiers under test.

To show this he first attempted to plot total harmonic distortion (THD) percentages against relative input level, the results of which can be seen in figure 2.19. Deciding "the lack of a wide variation between the curves indicated that THD plots are not very relevant to what the ear hears," a different approach was needed.

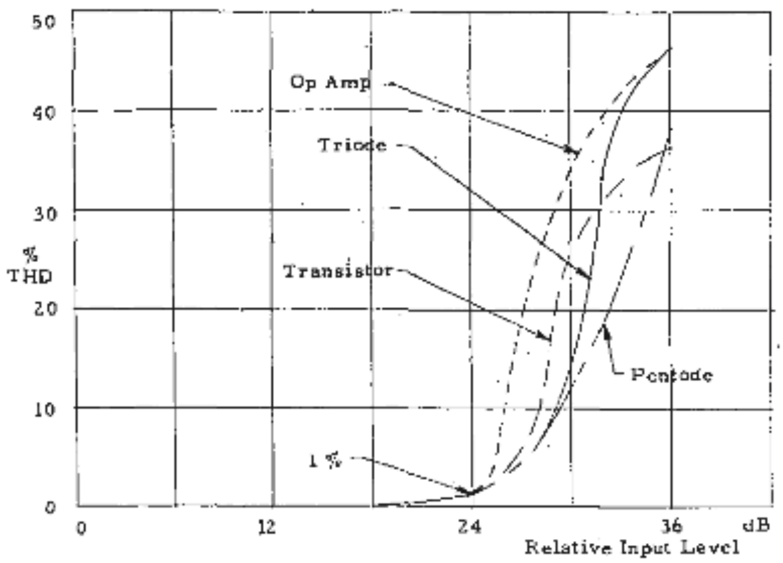

Figure 2.19 – Multi-stage amplifier comparison of total harmonic distortion. [32]

As shown by figure 2.18, the auditory system weights different harmonics. With this in mind the next step was to spectrally analyse the output waveform of a valve preamplifier, a decision that unveiled two of the now wider known traits of the 'valve sound':

- "The warmth is created by a large component of second-order distortion." [13]
- The output signal has asymmetrical clipping with a shifted duty cycle.

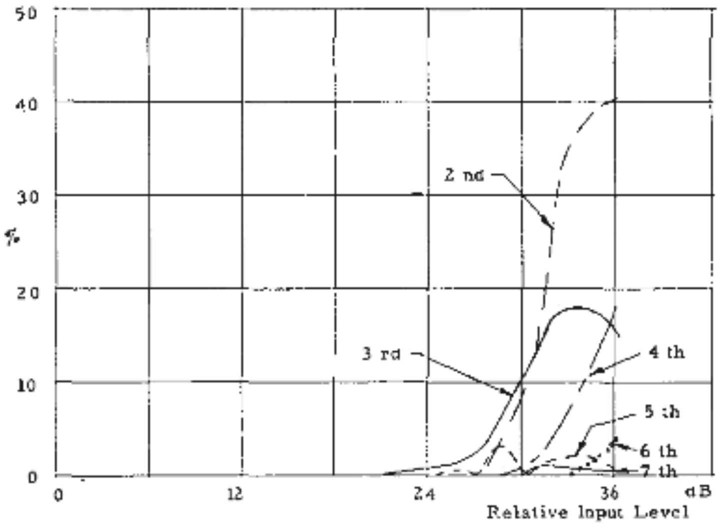

Figure 2.20 – Distortion components for two-stage triode amplifier. [32]

The resultant plots also took into consideration the weighting of the higher order harmonics in the output signal spectrum allowing a more detailed comparison to be drawn between the two states. It can be seen from figure 2.20 and 2.21 that the triode stage produces predominantly more even order harmonics, and the transistor stage odd order harmonics.

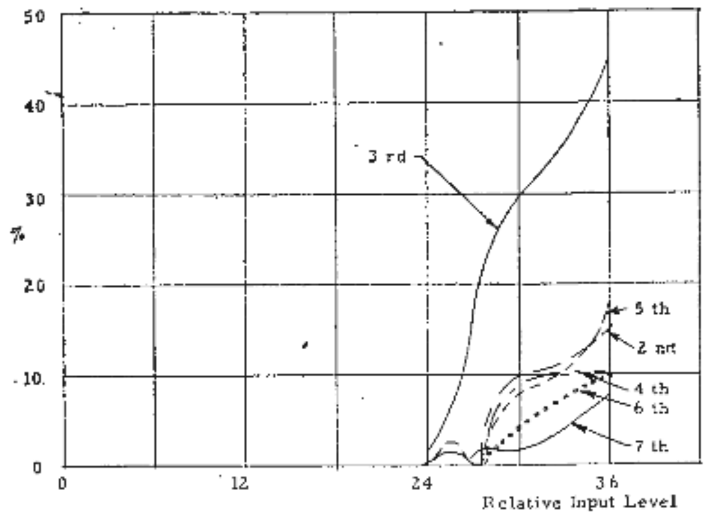

Figure 2.21 – Distortion components for multistage transistor amplifier. [32]

Not satisfied with just objectively showing a difference, Hamm wanted to use "the significance of musical harmonics to determine how the harmonics relate to hearing," believing that "there is a close parallel here between electronic distortion and musical tone coloration that is the real key to why tubes and transistors sound different." Using references from organ manufacturers experienced in determining how various harmonics relate to the coloration of an instrument's tonal quality, he related the measured harmonic amplitudes to make four paramount observations:

- "Odd harmonics (third and fifth) produce a 'stopped' or 'covered' sound. Even harmonics (second, fourth and sixth) produce 'choral' or 'singing' sounds."
- "Musically the second harmonic is an octave above the fundamental and is almost inaudible; yet it adds body to the sound making it fuller."
- "The third is termed a quint or musical twelfth. It produces a sound many musicians refer to as 'blanketed'. Instead of making the tone fuller a third actually makes the tone softer."
- "The higher harmonics, above the seventh, give the tone 'edge' or 'bite'. Too much 'edge' can produce a raspy dissonant quality. Since the ear seems very sensitive to the edge harmonics, controlling their amplitude is of paramount importance"

[32]

Relating these observations to Hamm's objective findings helps shed light on the perceived "warmth" synonymous with the 'valve sound', particularly because, unlike the use of valves for microphone preamplifiers in a recording environment, guitar amplifiers spend the majority of their lives being operated in the overload region; A region in which valve electronics can operate "without adding objectionable distortion," [32] and even adding consonance to the guitars signal by virtue of even harmonic generation.

Furthering Hamm's work, a white paper was published by Bussey and Haigler [35] looking specifically at the perceivable subjective differences between transistors and valves in electric guitar amplifiers. They balanced the frequency response and gain of the two systems under test to eradicate the effect caused "by the reactive speaker impedance and the output impedance of the amplifiers" particularly because "the output of a tube amp increases as speaker impedance increases," figure 2.22. [35]

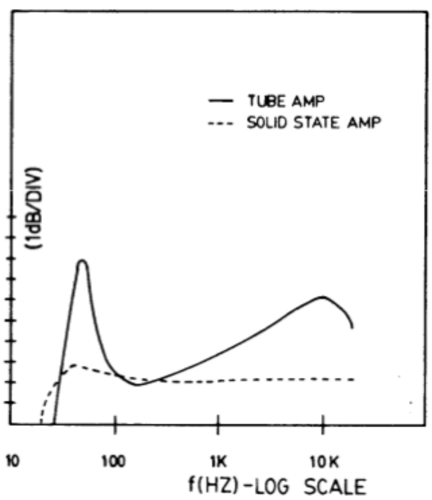

Figure 2.22 – Frequency response differences into a speaker load. [35]

With the gains set "to give identical output clipping levels for each amp," an objective measurement was made to show the differences in harmonic amplitude, figure 2.23. Guitar players with professional or semi-professional experience were then invited in to use "a specially designed A/B box which allowed the test subject to select either amp A or amp B via a footswitch."

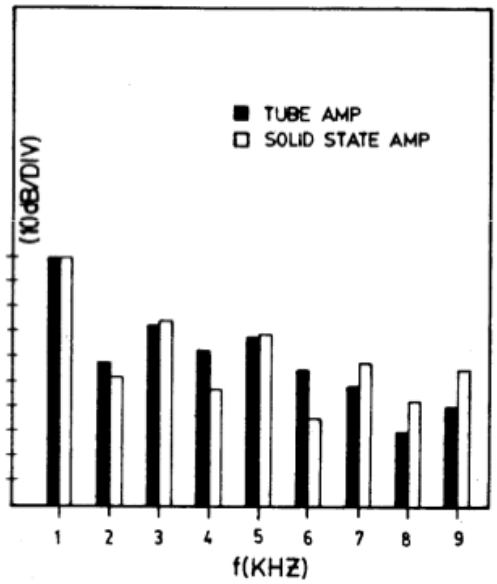

Figure 2.23 – Harmonic distortion spectrum with balanced frequency response and gain. [35]

The results of these subjective tests helped provide some evidence to support Hamm's theories. In the first set of experiments carried out the amplifiers "did not reach clipping," and although one subject "correctly identified the difference" he only did so on two out of three trials. The remaining eleven subjects were unable to reliably detect a difference.

However, when driven into overload half of the subjects could reliably detect a difference, one describing "the tube amp as sounding 'fuller' two out of three times." Interestingly the subject who identified a difference in the first test set correctly identified the difference 100% of the time in this test set, but did so due to a "buzzing" noise he felt was more pronounced in the valve stage, demonstrating just how perceptible the human ear is to noise.

This noise was investigated and subsequently determined as being "ripple intermodulation distortion" as a result of power supply ripple causing sidebands at 120Hz intervals either side of an input signal. "To verify this hypothesis, the filter capacitors on the power supply were doubled in value. This resulted in a 6 dB reduction in the relative amplitude of the sidebands." [35]

Ultimately, Bussey and Haigler's paper determined that perceivable differences could be heard when the two differing amplifier states are driven in to overload.

Rutt [17] further studied into the characteristics of valves for guitar amplification, paying particular attention to the preamplifier stage, almost exclusively triode nonlinearity, reiterating that the common ECC83 triode stage was perhaps the most commonly used stage in guitar preamplifier design. Rutt indicated, "Bussey and Haigler discuss the perceptive difference of tube and solid state guitar amplification at sub-overload signal levels." He made the distinction that his paper would "examine the effects of the vacuum-tube guitar amplifier distortion at levels of overload much greater than" previous works.

Prior to analysing harmonic content, Rutt initially drew attention to Hamm's findings that "the vacuum-tube amplifier under overload conditions causes a shift in the duty cycle of the output signal," as shown in figure 2.24. Identifying duty cycle shift as causing "more even harmonic distortion components than in solid-state amplifiers" he derived that grid current was a likely cause of said effect.

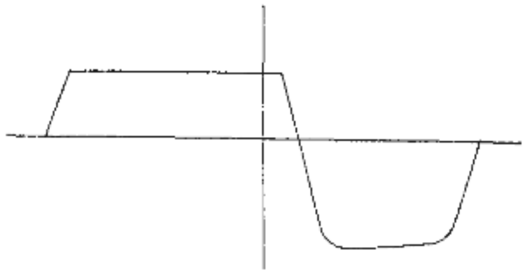

Figure 2.24 – Output waveform of a triode amplifier 12dB into overload. [32]

As figure 2.7 demonstrated, under-biasing causes the positive swing of an input signal to increase the grid voltage to the point of attracting electrons unable to be collected by the anode because it is too saturated. When this happens the current leaks out through the grid, "the grid circuit becomes forward biased." [17] This reduces the input impedance of the valve, loading the previous stage, resulting in an attenuation of the positive swings of input signal. Note the similarity in output waveform between figures 2.24 and 2.25.

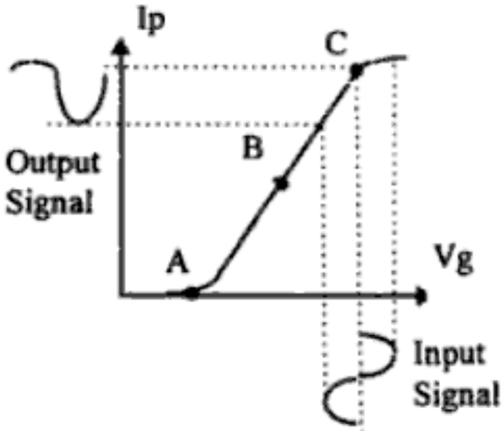

Figure 2.25 – An input signal overloading the positive swing at the point of saturation. [36]

Rutt introduced the concept of 'soft limiting' and 'hard limiting' showing that "hard limiting causes an undesirable type of distortion which adversely effects the output sound quality." He defined tubes as being soft limiting, and transistors as hard limiting. Barbour, however, believed that "the clipping characteristic of tubes is actually not much softer than transistors, but feedback tends to 'square-up' the clipping. Thus, the heavy feedback required in most solid-state designs gives them worse overload performance." [13]

Either way, the author agrees with Rutts opinion that "the most pleasant sounding amplifiers tend to have a very smooth and continuous transition from linear operation into overdrive." This is a trait that valves invariably possess due to the smooth curvature either end of their transfer characteristics, figure 2.25.

The two extremities of the transfer characteristics are known as *saturation* and *cut off*. For a valve the plate cut off, when I_p equals zero, is technically a hard limit because "any change in grid voltage will not change the plate current." [17] However, as said, the transition in to cut off is relatively smooth due to a gradual decrease in the triodes mutual conductance with reduced anode current. "Triode plate cut off is similar to transistor cut off, although a transistor has a less gradual transition into the hard limit." [17]

Saturation, by definition is the point at which "further increase in input signal to a device gives no increase in output," [29] and is the main factor that produces the 'soft limit' in valves because of its relationship with grid limiting. As a valve anode approaches its point of saturation the grid bias also reaches its most positive potential resulting in grid current, as described previously, also known as grid limiting. "Grid limiting is a soft limiting effect, since at all times the grid voltage will continue to change in response to changing input voltage, no matter how far into overdrive the grid is driven." [17]

In transistors, collector saturation, with both its junctions forward biased, determines the saturation point. Because its output current in this state is controlled purely by the supply voltage and load-line resistance, as opposed to the dynamic grid-to-cathode voltage in a valve, the effect is a hard limit as soon as the input signal amplifies to levels greater than the supply rails. Thus the transistor has hard limiting at either end of its transfer characteristics.

"Transistor amplifiers are usually biased for symmetric distortion to maximize output level, and as a result produce predominantly odd harmonic distortion components when driven by a sine wave. Transistor amplifiers can be designed to produce even harmonic distortion by shifting their bias points, but they still will not sound like vacuum-tube amplifiers. Transistor amplifiers in overdrive always have hard limiting on both sides of the modulating waveform." [17]

Rutt believed that ultimately the preference of valve amplifiers by guitarists is not due to the aforementioned duty cycle shift but due to the soft limiting of input signals. He noted that duty cycle shift only occurs when anode cut off begins to take effect, and that the subsequent grid-leak current causing "the shift in input DC level is responsible for the duty-cycle shift of the output waveform." [17]

Bussey and Haigler pointed out the difference in the way an amplifier responds to single plucked strings and chords specifically mentioning, "one subject felt that the difference between amps was an order of magnitude below the difference in striking the strings." [23] Rutt concurred with this idea, breaking down a guitar waveform into "riding waveforms" and "modulating waveforms":

He further qualified his belief of soft limiting as the preferential effect of valves by suggesting that soft limiting gives valve amplifiers "the ability to pass riding waveforms through during more portions of the overloaded modulating signal." [17] In other words, as the amplifier approaches saturation the small higher frequency transient nuances produced by guitar strings, which are "riding" on the "modulating" waveform peaks, are still present at the output giving greater harmonic detail to the sound.

It can be stated that the soft limit effect of the valve, in conjunction with the even order harmonic generation as a result of asymmetrical clipping caused by DC and duty cycle shift are the fundamental reasons for the perceived preference of the valve sound. Although as Cohen states, even in valve topologies, once both negative and positive modulating peaks reach the extremities of the transfer characteristics "symmetrical saturation will reduce second order products and increase third order products." [37] So it could be postulated that transistors and valves will sound alike when driven far enough into overload to clip both peaks of a waveform.

2.2.2- Developing Methods of Visualising Valve Characteristics:

In accordance with the trend of researching valve topologies at ever increasing levels of overdrive, Scott and Voss [38] published a paper in 1996, describing research that takes advantage of modern technological advancements to thoroughly test differing amplifier stages through the available gain range.

They introduced a new notion of "gain derivative surfaces" (GDS) to present surface plots of solid-state and valve amplifiers in the "hope that a solid-state amplifier design may be realized whose sound is characteristic of that of valve designs." Scott and Voss visually surmised the linearity and frequency response of an amplifier using frequency and level as independent variables, and second and third harmonics as dependant (z-axis) variables.

Taking ten spectral measurements per decade produced frequency responses over an input range of 40dB across the audio band of 20 – 20,000 hertz. Each data set was split into four plots, one for overall gain (frequency response), one for THD percentages, and most interestingly two plots for second and third harmonic amplitudes at each input frequency. Ten amplifiers were tested, the most relative to this thesis being a vacuum state preamplifier, a NAD 1020 stereo preamplifier, a Fender 1968 "Bandmaster" and "an all-solid-state circuit built by the authors using JFET and MOSFET transistors in the signal path, but with the circuit topology of amplifier 9 (the Bandmaster), adapted suitably in order to retain the transformer-coupling and overall level of feedback desensitivity."

It was duly noted that some sets required 10dB of attenuation before or after the device under test (DUT) that was not allowed for in the graphs, however the authors suggest, "it is the shape, not the absolute value (gain or output level of the amplifier) that is important in the comparisons." A further note to point out was their observation that "the thin 'ridges' that appear for low levels and low frequencies are caused by mains-frequency hum entering the circuits form unregulated supply rails." This is a common occurrence in valve amplifiers, particularly due to the expensive nature of smoothing and regulating high voltage supplies, and was first audibly detected by Bussey and Haigler as discussed previously.

Each amplifier was operated into a constant resistive load, and a reactive speaker driver load, it was the reactive load results that have more relevance because, again as Bussey and Haigler reported, "the output of a tube amp increases as speaker impedance increases." [35] This causes a frequency response that plays its part in the character of the "valve sound". Indeed the observed differences of the same reactive load has on differing circuitry and orders of distortion, "particularly of the second-order surface", provides evidence that "there is a complex interaction occurring with the load." [38]

Scott and Voss' observations of linearity and its relation to the weighting of harmonics showed that despite simple theory suggesting that "a 3dB reduction in fundamental level should result in 6dB fall in second-order distortion and 9dB in third," this in fact was not the relationship demonstrated by the resulting plots. As such the authors postulate "such behavior as observed here denies the possibility of a simple, low-order-dominant model for the nonlinear transfer function." This insinuates that there are more complexities in the valves harmonic components that contribute towards its characteristic sound.

As discussed, the particular plots of interest were those of the preamplifier stages and the "Bandmaster" and "pseudo-valve " amplifiers. The preamplifier stages, not having enough current to generate the power needed to drive a loudspeaker, could only be operated into a constant resistive load.

The paper draws attention to two of the previously discussed characteristics of the vacuum tube topology; "The extraordinarily high input and output levels that are needed to reach clipping, and its very gradual onset."

It also alludes to the fact that ECC83/12AX7 pre amplifier stages may well lend themselves to modeling because; "The distortion that is visible falls away with decreasing level at the rates expected from simple theory, assuming dominant low-order terms in the model."

Contrastingly, and as could be expected, the NAD 1020 stereo preamplifier demonstrates one of the defining characteristics of transistors, "the sharp and relatively early onset of overload." The paper also makes reference to this stages "slightly flatter THD surface", but speculates that this is likely to be a result of its differing circuit, class B topology, as opposed to the tubes class A operation.

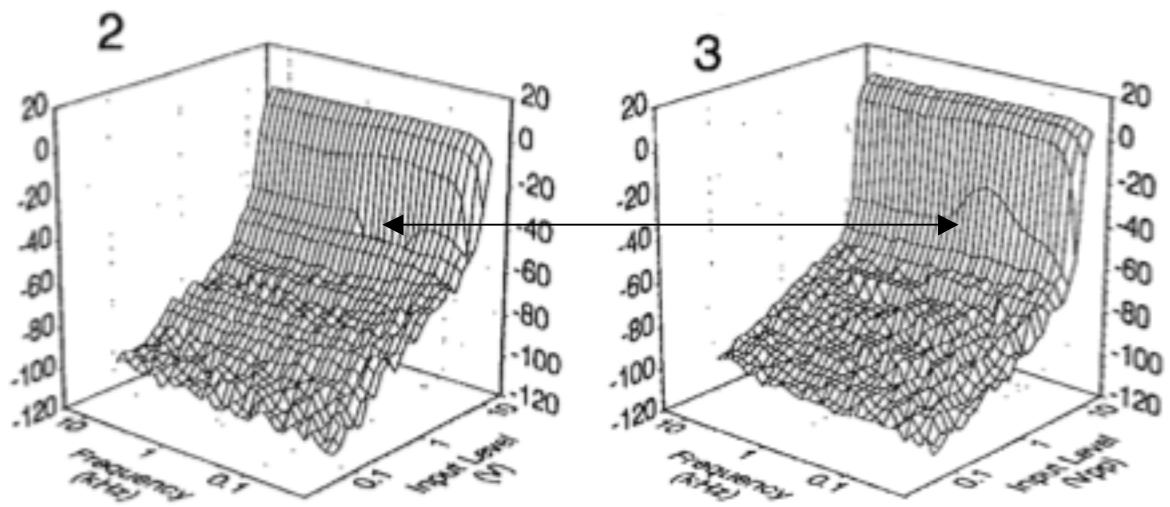

Figure 2.26 – 2nd and 3rd order surfaces for the NAD 2010 preamp (10kΩ resistive load).

To further relate the findings of Scott and Voss' paper to others in the related field there is another visible characteristic in the resultant surfaces. The arrows in figure 2.27 demonstrate that in the mid-frequency range, centred on approximately 1kHz, there is a visible attenuation in second order content, and a correlating increase in third order at the point of clipping.

Again contrastingly, comparison between the two states (valve and solid) also highlights the valve-state overall dominance of second order distortion amplitude throughout the gain derivative surface in relation to its third order distortion, figure 2.28.

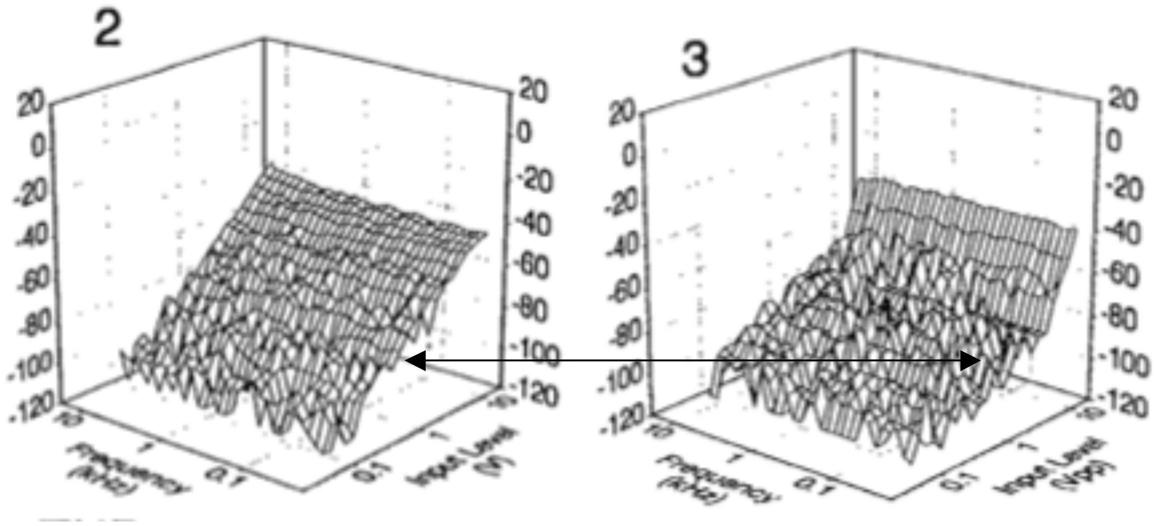

Figure 2.27 - 2nd and 3rd order surfaces for the valve state preamp (10kΩ resistive load).

The Bandmaster plots, figure 2.29, whose topology is "designed for guitar rather than high fidelity application," markedly exhibits the distortion preference of guitar amplification, demonstrating high distortion levels due to minimal local feedback. The interesting observation made is that "the distortion surfaces do not have sharp convoluted portions," possibly relating to the subjective "smooth" distortion and objective "soft limiting" associated with the 'valve sound'.

Interestingly the construction of a "pseudo-valve" stage, substituting MOSFETs and JFETs in place of the valves in the bandmaster circuit, also produced smooth gain derivative surfaces, figure 2.30. It still has the prevalent third harmonic expected from solid-state topologies, although the substitution does embody a large amount of second order distortion, which has greater affiliation with the vacuum tube plots as opposed to the BJT plots. This possibly alludes to the fact that MOSFET and JFET amplification might allow more 'valve-like' emulating circuits to be produced, something to be discussed further in Chapter Three, though subjective analysis was unfortunately not conducted.

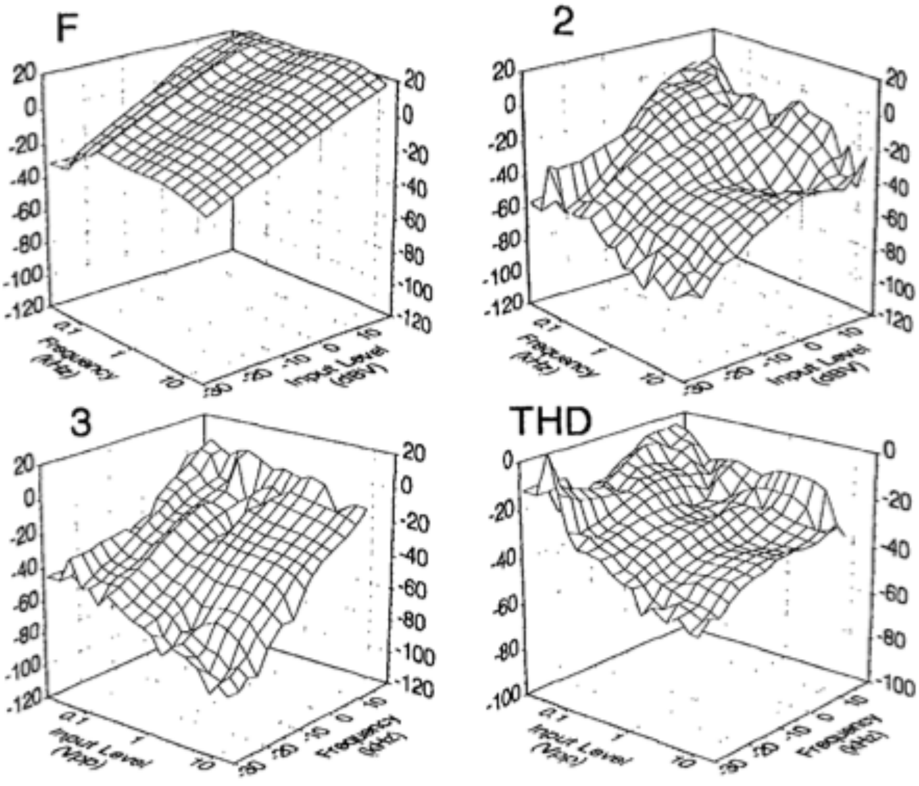

Figure 2.28 – Gain derivative surfaces for the "Bandmaster" preamp (10kΩ resistive load).

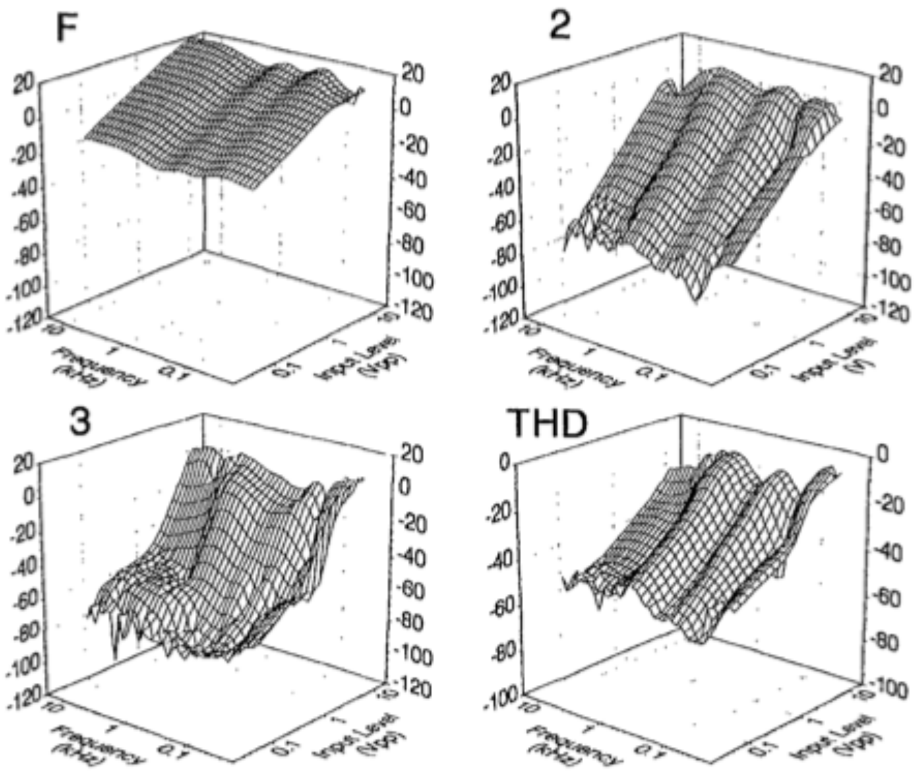

Figure 2.29 - Gain derivative surfaces for the "pseudo-valve" preamp (10kΩ resistive load).

2.3 - Chapter Summary:

This chapter has introduced all the valve fundamentals and discussed how they are affected by the implementation of power reduction. As stated, further research into the effects of anode voltage reduction is necessary to fully understand the resulting changes, if any, in tonal characteristics.

A literary study has highlighted the main traits of the 'valve sound' and determined from which part of the valve topology the favourable characteristics yield. These traits are:

- Asymmetrical clipping, and an output duty cycle shift, as a result of smooth curvature of the transfer characteristics caused at saturation by grid current loading input stages, and at cut off by a gradual reduction in mutual conductance.

- Even order harmonic dominance as a result of the asymmetrical clipping.

- Controlled high order "edge" harmonics due to inherent Miller capacitance between the grid and anode creating a low-pass filter.

These three factors can be objectively shown either through spectral analysis of output signals to produce plots of harmonic amplitude against output level, or through the more modern approach of gain derivative surfaces. As such any analysis to be carried out in to anode voltage reduction should utilise either one or both of these plotting techniques.

Power reduction can be achieved either through analogue electronics, or through digital implementation of the valve characteristics. As such the ensuing chapter will examine both existing methods. Ultimately it must be stated that in order to achieve accurate power reduction, without the loss of the described valve tonal characteristics, it is of great importance that the transfer characteristic plot must be reduced in size without changing the shape of the curves approaching saturation and cut off.

Chapter 3 - Digital and Analogue Power Reduction

Power reduction for guitar amplification can be achieved via two distinctly different methods. The first method uses analogue devices either to reduce the power by directly attenuating the output, or to reduce the overall power consumption of the amplifier. The second method takes the characteristics of the 'valve sound' and uses either analogue or digital modelling to reproduce said sound, requiring less power to operate.

In the current global trend of increasing energy efficiency it is deemed ethical to reduce power consumption of an amplifier as opposed to wasting energy by losing output power through an attenuator. Consequently this thesis does not examine output power attenuators, or hot plates, which reduce amplifier output power using dummy loads and variable wattage-splitters strapped across amplifier outputs to inefficiently convert large amounts of electrical energy into heat.

This chapter will discuss the most current methodology of amplifier power reduction, Kevin O'Conner's "London Power Scaling" [39]. It will then present the alternate method of amplifier modelling, analysing both digital and analogue derivations, and determining which may prove to ultimately be the most practical and subjectively pleasing method of power reduction.

3.1 - London Power Scaling:

"London Power" is a company, which although has been registered since 1995, has only recently become a commonly heard name in the world of guitar amplification. Keeping all its secrets closely guarded there is not a great deal of unbiased literature discussing the registered trademark of "power scaling". In fact London Power has published *all* the available literature, and no schematics of the power scaling circuit have been released.

Two things are known about London Power Scaling as a result of their website [39], and engineer Bruce Clement [40]. The first is that they have reached the same conclusion as this thesis stating that in order to 'power scale', "in technical terms, all that must be accomplished is to keep the "transfer curve" of the amplifier the same." [39] The second being that the 'power scaling' circuit is split into two sections, one "adjusts B+ from about 450VDC down to less than 20VDC."[40] (B+ is another term for anode voltage) The other known as "Drive Compensation lets you maintain the correct mix of preamp and power amp overdrive at any volume level as the power amp is effectively resized via Power Scale." [40] This reduces the input signal to the stage whose anode voltage has been reduced.

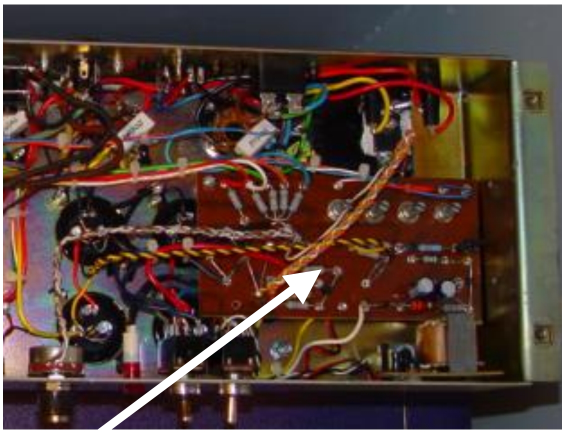

Figure 3.1 – The "power scaling" circuit inserted into a 100W Marshall Plexi reissue. [40]

We also know from various respected audio electronics forum discussions of London Powers "Modding Kits" that the overall 'power scaling' circuit works by switching increasing values of resistance in series with the screen grids of tetrode and pentode output valves*. This effectively holds the screen grid at a more positive potential as opposed to its usual lower potential, which negates the effect of the anode, making the screen grid a temporary anode with a reduced voltage.

---

*London Powers 'power scaling' circuit is designed specifically to reduce the power of the output stage of an amplifier. Output stages in valve amplifiers most commonly use either tetrodes (6L6s) or pentodes (El34s).

Essentially the circuit allows the amplifier to operate under normal conditions until the power scaling circuit is switched in, making the screen grid the new anode, making the whole valve operate in triode mode, and so ultimately achieving power reduction by reducing the anode voltage of a triode. As was determined at the end of Section 2.1.4, the largest power reduction is achieved by dropping the anode voltage of an amplifier stage, something which is obviously being utilised by London Power and as such warrants further investigation.

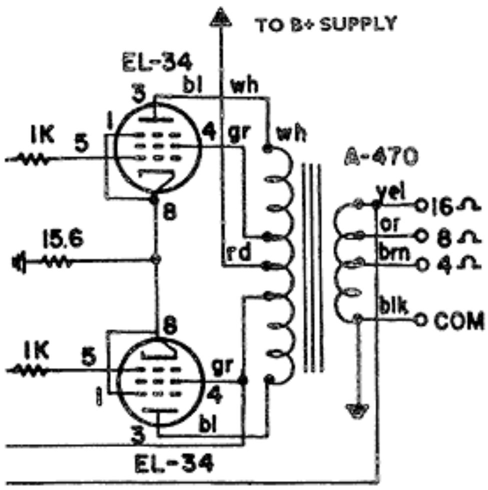

Figure 3.2 – Triode connection of a pentode. [41]

It is documented [41] that using pentodes as triodes reduces the amount of global feedback in an amplifier, figure 3.3, resulting in "the gain of the 'new' tube being less than one tenth of the gain of the 'old' tube." This also has the effect that "output power is cut in half," confirming that London Powers circuit will reduce power output in the output stage.

SPECIFICATION	PENTODE	TRIODE CONNECTED
Open loop gain	175 (44dB)	50 (34dB)
Closed loop gain	20 (26dB)	15.6 (23dB)
Feedback	18.8 dB	10 dB
Distortion (TH+IM)	1% @ 30 watts	2.7% @ 15 watts
Feedback factor (1-BM)	8.7	3.2
Sensitivity	1.3v for 30 watts	1.6v for 15 watts

Figure 3.3 - effect of triode connection of a typical EL34 pentode. [41]

Due to the general lack of reliable sources in the subject area of London Power this thesis does not speculate any further upon the workings of Kevin O'Conner's "power scaling" circuit, instead preferring to take an empirical approach to the effects achievable using anode voltage reduction in triodes. The notion of controlling input signal amplitudes is definitely something to bear in mind.

3.2 - Valve Modelling:

Naturally the progression of technology is a forwards moving mass with most avenues gathering inertia in an exponential manner. Transistor technology was and still is such a movement, rapidly accelerating away from the use of valve technology.

"The past 50 years have seen the demise of the vacuum tube, the development of the transistor, and the development of the integrated circuit. There has been an explosive development of analogue and digital circuits and systems." [58]

Perhaps one of the more significant signs of this technological movement is the lack of white papers looking specifically at analogue valve electronics in recent times. The majority of to-date research into valve amplification has focussed upon modelling, though more recently this movement has split into two factions;

- Simulation – copying and reproducing the output characteristics of a valve.
- Emulation - digitally producing the dynamic inner workings of the valve.

This shift in thinking appears to encompass the acceptance of the valve as being a dynamic system not capable of accurately being statically modelled, requiring "the solution of a nonlinear system of implicit equations," [46] utilising complex and evolving programs of values and coefficients from look up tables of 'real world' measurements of valve parameters.

This section will discuss currently used methods of digital and analogue modelling of the valve, entertaining some of the success of certain theorems such as Rushman's, and Milman's, used to gain differential algebraic equations for modelling electrical circuits [42] [43]. It will also evaluate some of the commonly referenced 'standard' triode models such as Leach's [44] and Norman Koren's [45].

Valve modelling is a topic well worth researching in to because many believe it is the future of the valve amplifier. It relates well to this thesis by being the largest power reduction of the valve sound achievable for guitar amplification. To Quote Pakarinen and Yeh;

"The ultimate goal of amplifier emulation is to convincingly reproduce all the fine detail and nuances of the vacuum-tube sound because despite their acclaimed tone, vacuum-tube amplifiers have certain shortcomings: large size and weight, poor durability, high power consumption, high price, and often poor availability of spare parts. Thus, it is not surprising that many attempts have been made to emulate guitar tube amplifiers." [46]

3.2.1- Digital Valve Simulation:

Computer aided design (CAD) has become "almost universally used in the design of electronic circuits." [44] Arguably the most common of these CAD programs is SPICE (Simulation Program With Integrated Circuit Emphasis) and as such researchers in the field of valve electronics have been developing valve models for use with SPICE because "simulation of vacuum tubes is now a necessary technique in the design of a modern tube amplifier." [47]

The fundamental premise of valve modelling is the Child-Langmuire or Three-Halves Power Law (14), which governs the relationship between anode current and anode voltage as shown:

$$I_a \propto (V_a)^{3/2}$$

Child-Langmuire Law (14)

[9]

For a triode, approximation of this relationship using a binomial series (x^2, x^3, x^4, x^5…) demonstrates the generation of harmonics, and because "the terms die away very rapidly" [9] a triode can be expected to produce predominantly second harmonic distortion as shown by Danyuk's [48] graphical representation of triode distortion products:

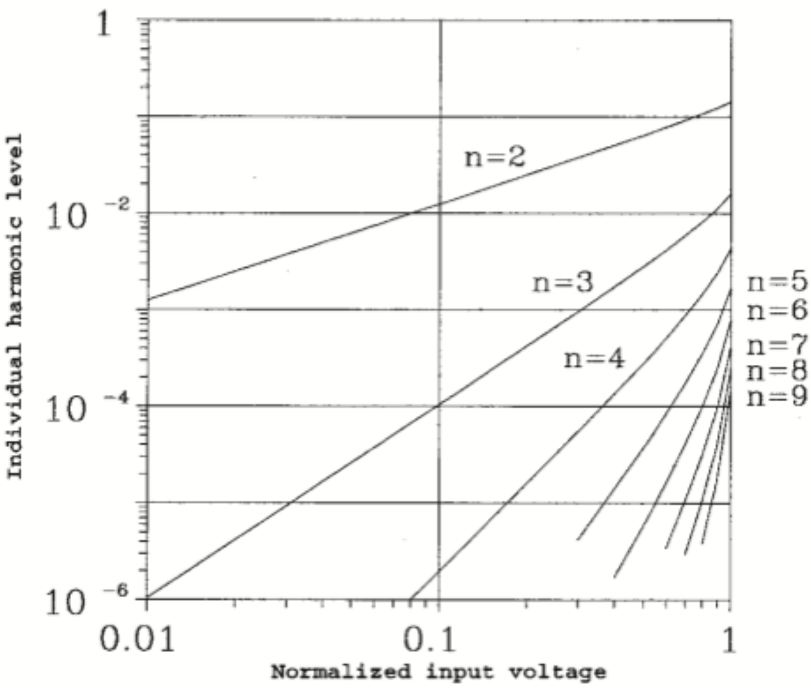

Figure 3.4 – Calculated distortion products for triode anode current, assuming an ideal three-halves power law transfer characteristic. [48]

In 1995 Rydel published the first paper orientated towards producing some SPICE models of valves for use with CAD. Taking Spagenberg's [49] original equation of triode anode characteristics (15) as the starting point for a mathematical model of the anode current and grid voltage curves, Rydel duly noted that "the expression leaves out two parasitic phenomenon existing in triodes: the change in gm (mutual conductance) and the change in the internal resistance when Vg becomes more and more negative."[47]

$$\mathbf{I_a = g.(V_g + (V_a/\mu))^{2/3}}$$

Triode anode current equation (15)

Where,

Ia = Anode current.

g = Perveance of the effective diode between grid and anode.

Vg = Grid voltage.

Va = Anode voltage.

μ = Amplification factor. [49]

Plotting the resultant curves for Spagenberg's equation, Rydel calculated that there was a 9.476% "squared residuals coefficient error" in the fit of the mathematically generated curves when referenced to "well defined" data values taken from "the Telefunken tube 6463." [47] However, by including the "parasitic phenomena" as described in a "new expression for a triode," equation (16), he reduced the error to 3.2%, "which is a three times factor improvement." The findings are graphically demonstrated in figures 3.5 and 3.6.

$$Ip = g.\left(1+\frac{Vg}{B}\right).\left[Vg + \frac{Va+Vc}{\mu}\right]^{3/2}.\left[\frac{Va}{Va+c}\right]$$

Rydels new expression for a Triode. (16)

Where,

B = a coefficient allowing g to change as a function of Vg.

Va+c = **Va+Vc** = Rydels coefficient to represent positive grid voltage. [47]

Despite the improvement in the error of Rydel's equations for SPICE modelling valves he ultimately stated that in his opinion "grid current for positive potential is a highly non reproducible phenomena." [47] He also deduced that although he defined "a complete set of new and more precise expressions than in classical textbooks," there is a "big variability in characteristics," and as such further work was needed to formulate more accurate equations.

$$I_a(V_a, V_g) := 1.625 \cdot \left(V_g + \frac{V_a}{18.13}\right)^{1.5} \qquad ERR = 9.476$$

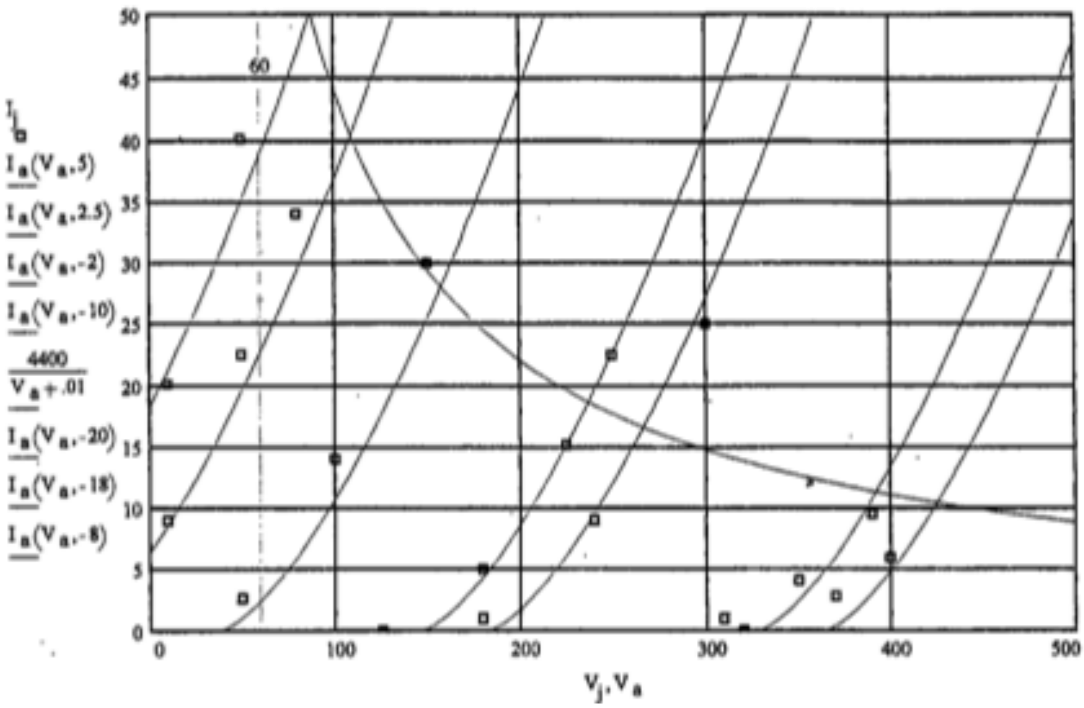

Figure 3.5 – Comparison of Spagenberg's equation with measured parameters. [47]

$$I_a(V_a, V_g) := \left(1 + \frac{V_g}{34.8}\right) \cdot 1.51 \cdot \left(V_g + \frac{V_a + 11.9}{17}\right)^{1.5} \cdot \frac{V_a}{V_a + 3.76} \qquad ERR = 3.253$$

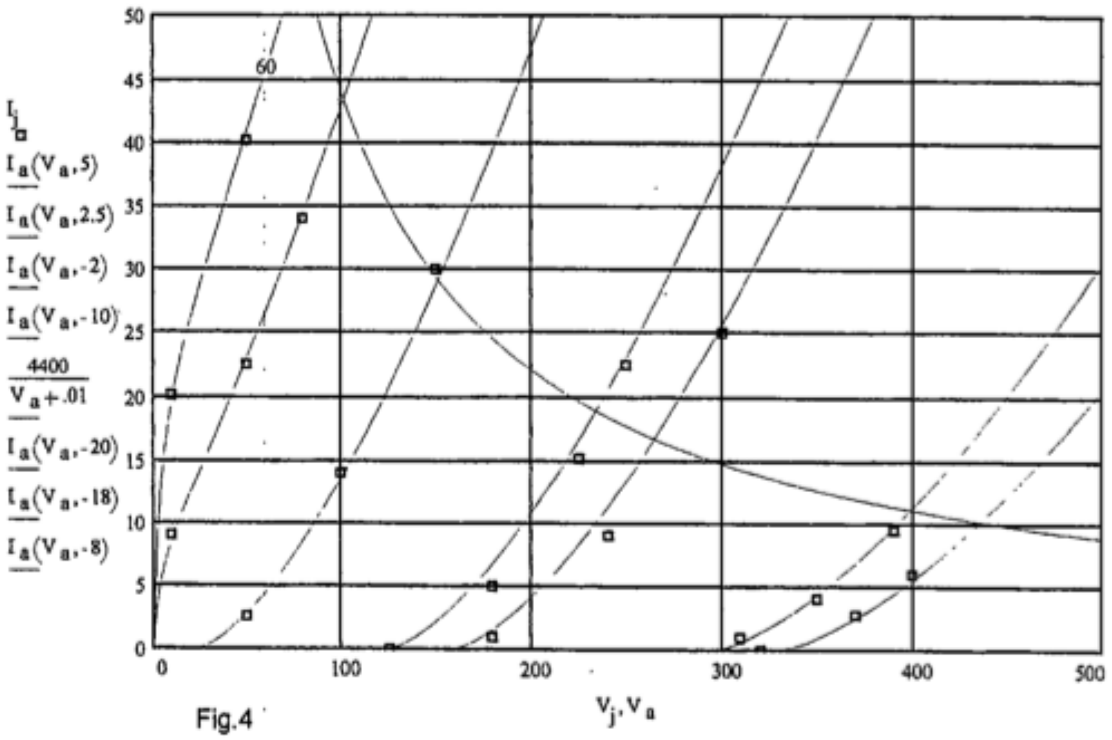

Figure 3.6 – Comparison of Rydel's equation with measured parameters [47]

In the same year Marshall and Leach [44] took a different approach to Rydel's mathematical model, attempting to produce an equivalent circuit model using *PSpice* functions to "implement the nonlinear equation for the plate current," (15). They used SPICE to calculate values from the "subcircuit" shown in figure 3.7, to produce plots of anode current versus anode voltage for a typical 12AX7 triode.

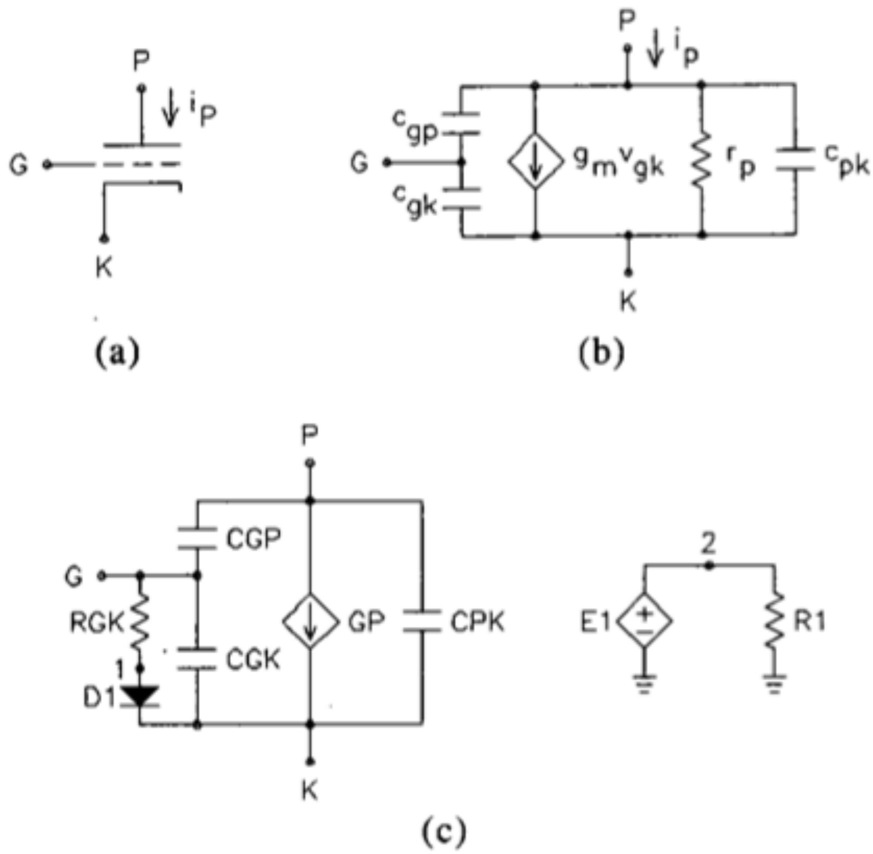

Figure 3.7 – (a) Circuit symbol for triode. (b) Small-signal model of triode. (c) *PSpice* model for triode. [44]

"Parameters used for the 12AX7 were derived from curves of tube characteristics given in the RCA Receiving Tube Manual," and perveance and amplification factor values were determined to produce anode characteristic curves that matched RCA recorded values at points (Vg = 0, Va = 150V, Ia = 3.2mA) and (Vg = -5V, Va = 450V, and Ia = 0.32mA). Their resultant calculated curves showed "excellent agreement with those given in the RCA manual."

One fundamental problem with their approach however was that the *PSpice* model "approximated" the grid circuit using RGK and D1. As Norman Koren pointed out, "they assumed that the grid has perfect control over the plate current, i.e., that there is no leakage current," [45] which as discussed in section 2.2 is quite an important parameter in the 'valve sound', ultimately creating a flaw in the model.

Despite this flaw, and because Marshall and Leach used an equivalent "subcircuit," they were able to produce two interesting sets of data. Using "the parametric analysis feature of *PSpice,*" they determined a family of curves showing "the transient response for a 1kHz sine wave applied to the amplifier input," figure 3.9. Doing this also allowed the use of Fourier analysis to "calculate the distortion component in the output waveform," which indicated, "That the principle distortion created by the amplifier was second order." This shows good agreement with the findings presented in section 2.2 and is perhaps one of the main determining factors as to why the "Leach" model is now referred to as a 'standard' model.

Peak Input Voltage	% 2nd HD	% THD
0.2 V	0.791	0.862
0.4 V	1.51	1.54
0.6 V	2.37	2.39
0.8 V	3.56	3.62
1.0 V	6.93	7.66

Figure 3.8 – *Pspice* calculated distortion percentages. [44]

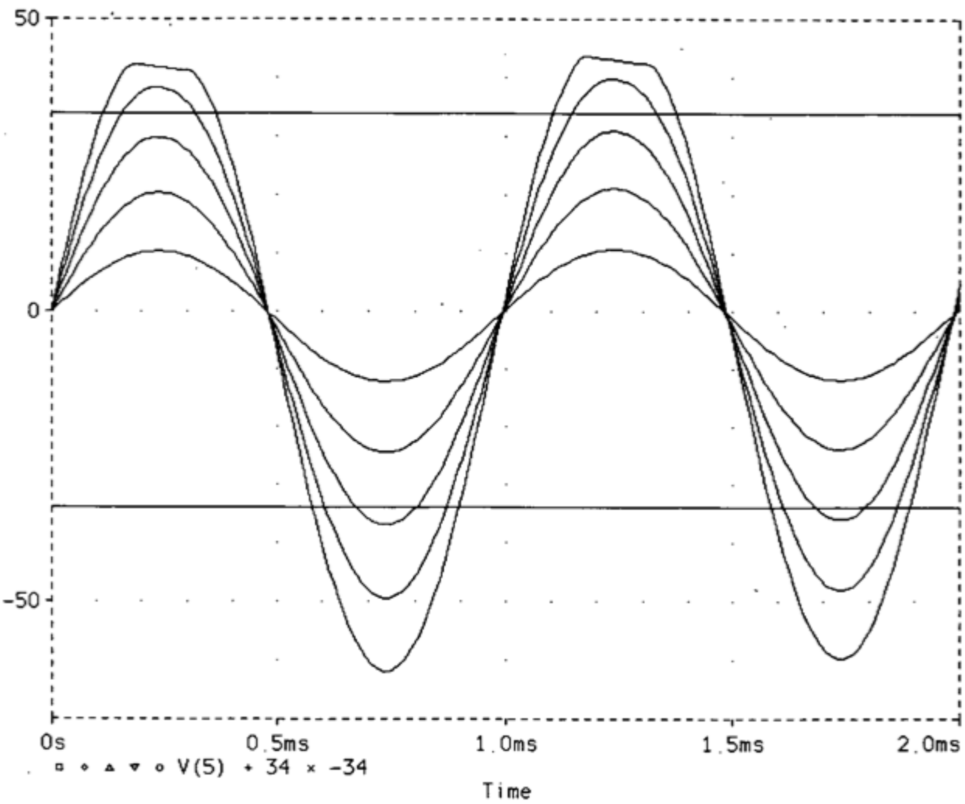

Figure 3.9 – *Pspice* transient response of a triode model peak sine-wave input voltages of 0.2, 0.4, 0.6, 0.8, and 1.0 V. [44]

Much like Rydel, the conclusion of Marshall and Leach was that the model was a good approximation, but that ultimately "the accuracy of the simulation depends on the accuracy of the models." Unfortunately this 'standard' model could not be deemed as "accurate" because it failed to include both grid leakage current and Miller capacitances.

In the following year, 1996, Scott and Parker [50] graphically demonstrated (figure 3.10) the inherent inaccuracy of the Leach model by comparing the 'standard' model with measured values. They calculated the error in the curvatures fit to be 0.0046%, a notable improvement on Rydel's, 3.253%. They then further reduced this error to 0.009% (figure 3.11) using experience from their work in the world of modeling GaAs FETs, "in particular MESFETs and HEMTs", to derive an equation with four model parameters as opposed to the 'standard' two; perveance and amplification factor.

Despite the improvement in the curvature fit, "particularly the close fit in the region approaching cutoff," they, like the others, had to conclude; "That simple physics used to derive equations misses out on much that is going on inside the valve."

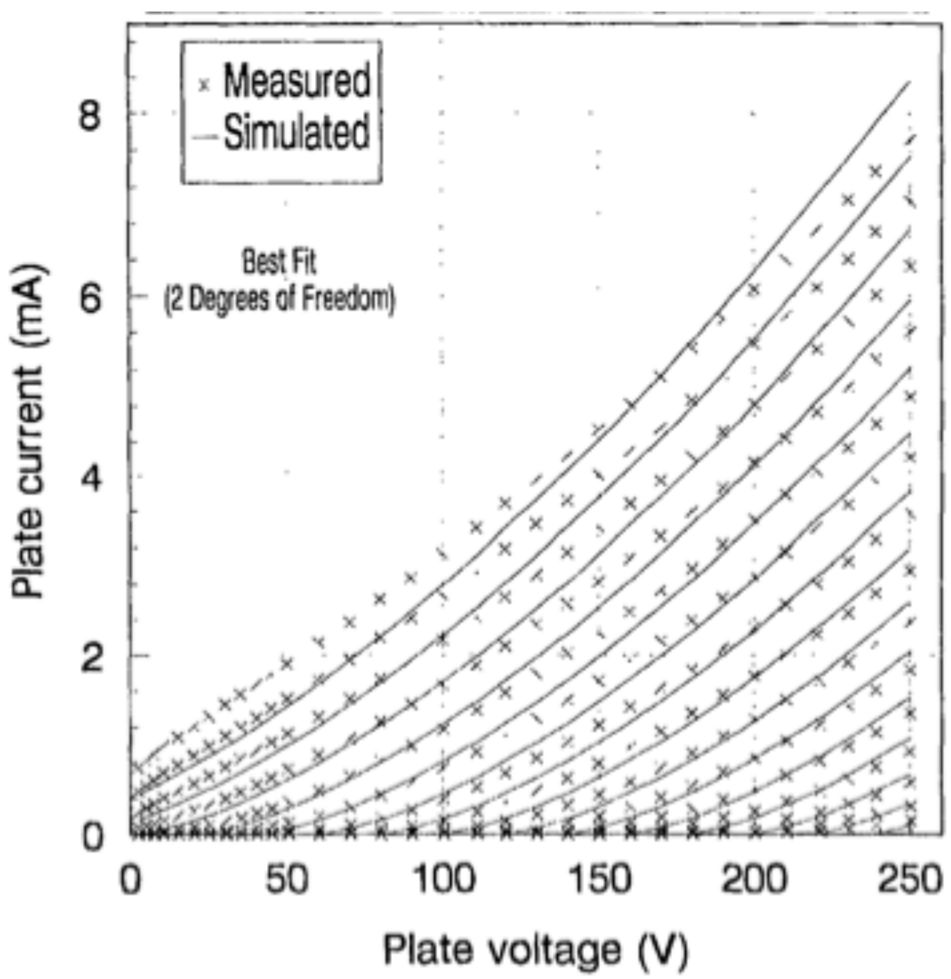

Figure 3.10 – Comparison of the Leach triode model with measured values. [50]

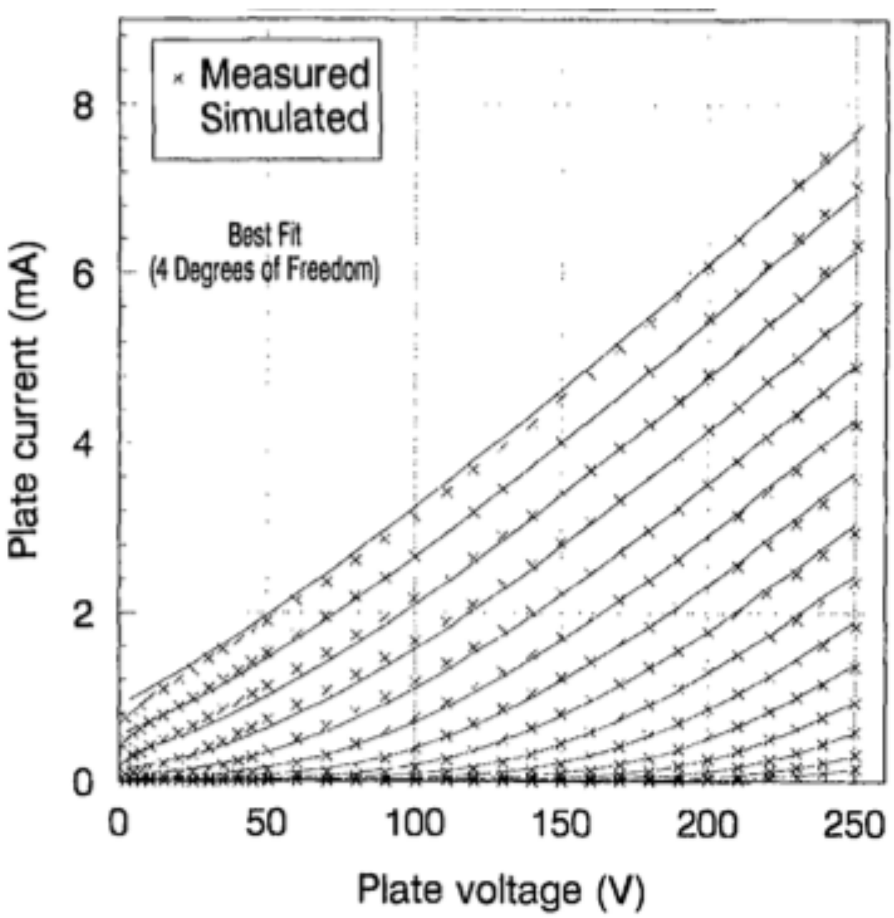

Figure 3.11 – Comparison of the Scott and Parker improved triode model with measured values. [50]

Taking into consideration the shortcomings of these previous works Norman Koren [45] introduced the idea of modelling tube characteristics using "*phenomenological*" equations, defined as "equations that model the behaviour of physical phenomena using a reasonable number of parameters, but are not derived from fundamental physics." [45] He derived an equation from the Leach model designed specifically so that anode current is always greater than zero whenever anode voltage is greater than zero. In doing so he corrected Leach's flawed assumption that there is no grid leakage current, which resulted in "a poor estimate of plate current for large positive plate voltage and large negative grid voltage." [45]

Koren stated that this "modelling error would not be serious if tubes did not operate in the region of greatest error," the aforementioned regions of large anode voltages and negative gird voltages. However, as section 2.2 detailed, a valve is commonly pushed into said regions of operation when stages are driven into overload, and as such accurate modelling of these regions is imperative, particularly for guitar amplification where the overload characteristics are embraced. Norman Koren's resulting equation (17) dramatically decreased the error in the anode curves for these regions, and thus took over from Leach as the new 'standard' model.

$$I_p = \frac{E_1^{E_x}}{K_g}(1 + \mathrm{sgn}(E_1))$$

Norman Korens "*phenomenological*" equation. (17)

Where,

$$E_1 = \frac{V_{pk}}{K_p} \ln \left[1 + \exp\left(K_p \left(\frac{1}{\mu} + \frac{V_{gk} + V_{ct}}{\sqrt{K_{vb} + V_{pk}^2}} \right) \right) \right]$$

K_g , K_p , and K_{vb} = Values determined by trial-and-error using Koren's appropriate plate curve program, which achieve the most accurate fit in the regions of; K_g - relatively low negative grid voltages; K_p - large positive plate voltage and large negative grid voltage; and K_{vb} - at the "knee" of the characteristic curves when the tube is operated with positive grid voltage. [45]

It was this equation that provided the building block for the most up to date (2009 and 2010) published research in to valve modelling, Cohen and Helies two papers, "Simulation of a guitar amplifier stage for several triode models," [42] and "Measures and parameter estimation of triodes, for the real time simulation of a multi-stage guitar preamplifier." [43]

Their first paper initially focused on "the static triode models of Leach and Norman Koren," comparing the two and determining, as discussed previously, that the Koren model "is more realistic than the Leach one, modelling the behaviour of the triode better for high grid and plate voltages." [42] They then made the progressive step forward by moving away from static modelling to dynamic modelling, taking Miller capacitance in to consideration;

"The accuracy of the triode model can be increased if its dynamic behaviour is considered," this was achieved by taking Norman Koren's static model and including the three parasitic capacitances between its poles. "Although these capacitances are low, they significantly change the frequency response of the triode stage, due to the Miller effect. C_{gp} is multiplied by a gain, contrary to the two other capacitances, and acts in combination with R_g as a low pass filter." [44] They presented two frequency response plots to demonstrate the measured difference as a result of including the Miller effect, shown in figure 3.12:

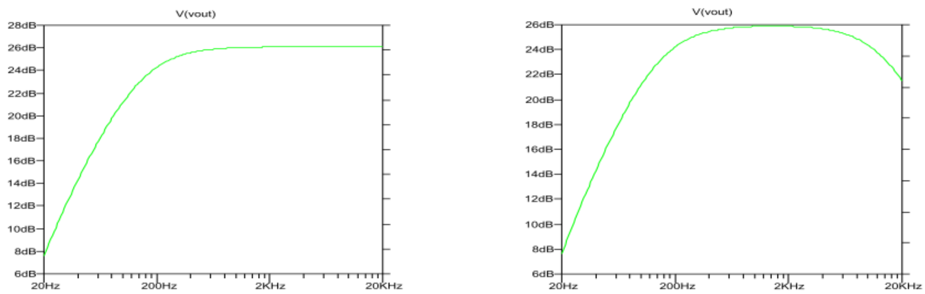

Figure 3.12 –Frequency response without (left) and with (right) Miller capacitance. [42]

In order to implement the dynamic model Cohen and Helie exploited a technique more commonly used in Automatics, linear state-space representation, re-writing the original Koren's system of differential equations as a system of first order differential equations (18). "The solution to this system is a vector that depends on time and which contains enough information to completely determine the trajectory of the dynamical system. This vector is referred to as the state of the system." [51]

$$\begin{aligned} U &= V_{in} \\ X &= [V_k \quad V_{out} - V_p \quad V_g - V_p]^T \\ W &= V_p \\ Y &= V_{out} \end{aligned}$$

Cohen and Helie's state-space variables. (18)

Where,

T = time. [42]

This state-space representation produced a "stiff problem", with some terms producing "rapid variation in the solution," meaning that certain terms for each variable (vector) of the space-state representation did not converge on one numerical solution, an unsatisfactory condition for dynamic simulation. To stabilise the parameter values they employed "the Newton-Raphson* algorithm for both the resolution of the dynamic and non-linear implicit equations." This resulted in a stable "new numerical scheme" implemented as a VST plug-in. [42]

Undoubtedly Cohen and Helie's model was a vast improvement on the prior 'standard' models, with the "simulation of the triode stage producing signals with a rich harmonic content giving a good approximation of a high gain guitar amplifier sound." However, the authors had to conclude, "The need for the Newton-Raphson algorithm for both differential equations and nonlinear implicit equations has a cost, in terms of CPU consumption, which is not negligible for real-time applications." [42]

Consequently Cohen and Helie decided that "code optimisation" was needed before the model could be used successfully for real-time application of a valve model, which ultimately is what is needed to replace the analogue version.

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*The Newton-Raphson algorithm is an iterative procedure that can be used to calculate maximum likelihood estimates (MLEs). The basic idea behind the algorithm is the following. First, construct a quadratic approximation to the function of interest around some initial parameter value (hopefully close to the MLE). Next, adjust the parameter value to that which maximizes the quadratic approximation. This procedure is iterated until the parameter values stabilize. [52]

Cohen and Helie's latest paper [43] produces the results of their "code optimisation" to "yield efficient stable simulations" of a triode multi-stage guitar preamplifier. They have also developed a new triode model, advancing on the previous nonlinear differential algebraic system by realizing measures of real triodes to "characterize the capabilities of aged and new triodes." [43].

They have developed a device "to measure the static behaviour of different kinds of triodes," although this has still to be furthered to study the dynamic behaviour. Interestingly they have adopted the previously discussed surface technique, as introduced by Scott and Voss [38], to visually display measured triplets of anode current, anode voltage, and grid voltage, shown in figure 3.13:

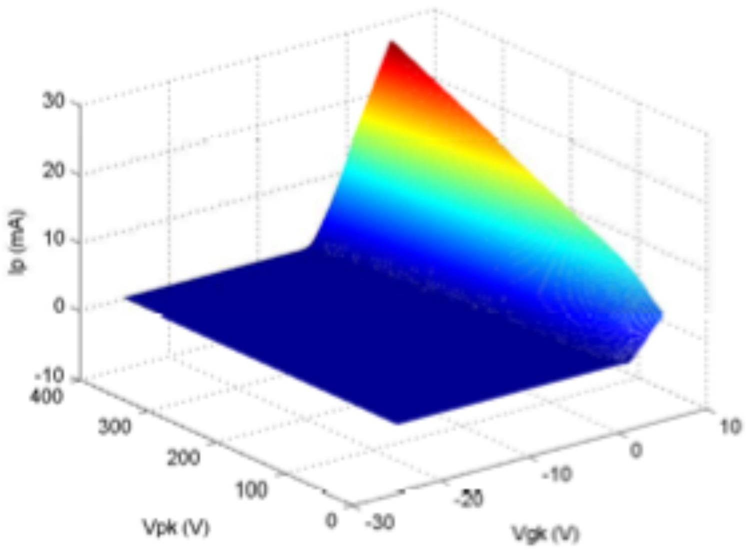


Figure 3.13 – Characterisation surface for the anode current of a triode. [43]

They use their measured parameters as an additional state-space variable (vector), $\theta = [\mu\ E_x\ K_g\ K_p\ V_{ct}\ K_{vb}]^T$, in conjunction with their previous state-space representation (18) to produce an anode characteristic plot, akin to those produced by Rydel, Leach, Scott, Parker, and Koren. The improvement in the models fit is clearly visible, figure 3.14, producing a "quite accurate approximation of a real triode" as well as visible output harmonic content with dominant even order harmonic distortion components, figure 3.15, as would be expected from a triode stage.

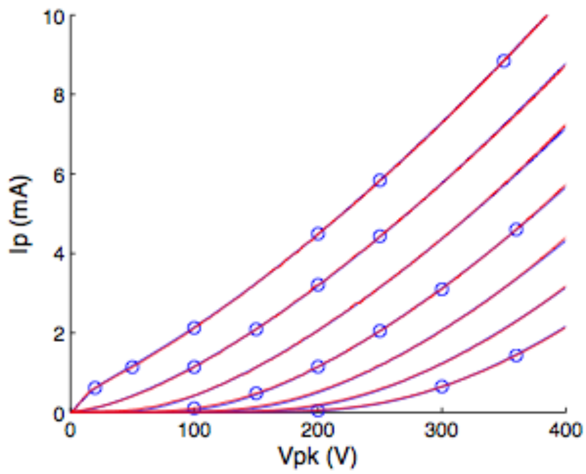


Figure 3.14 – Simulated anode characteristics comparison with measured data (circles). [43]

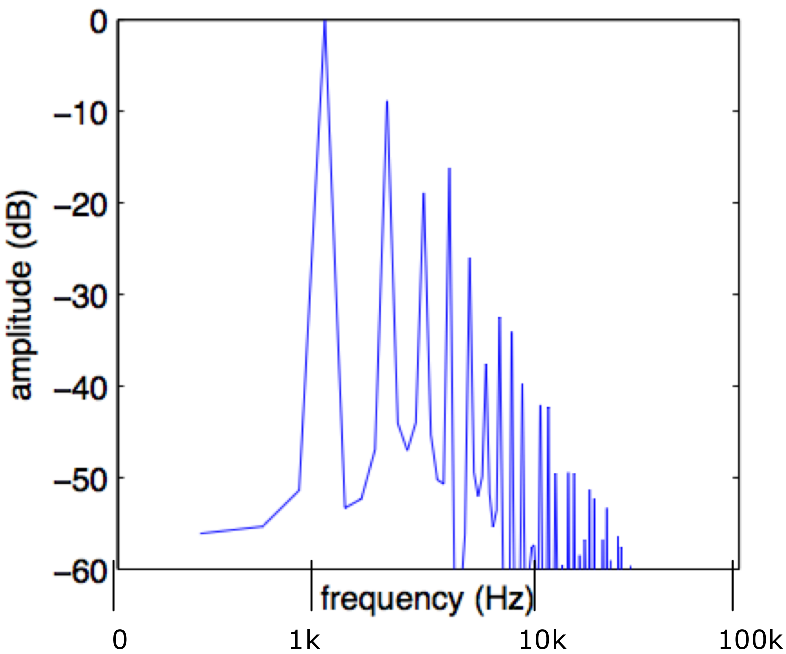


Figure 3.15 – Output FFT from simulated triode model. [43]

By their own admission though, the authors still do not deem their model as being able "to match exactly with a real triodes behaviour." This being the most up to date research in the field of valve modelling suggests that it could well be a while before the analogue valve has a completely accurate, real-time, very low-power replacement. Again to quote Pakarinen and Yeh;

In conclusion, the complicated interdependencies and dynamic nonlinearities in vacuum-tube amplifiers make their accurate physical modelling extremely demanding. As a result, approximate models simulating only some of the most noticeable phenomena have been developed by the amplifier modelling community." [46]

3.2.2- Analogue Valve Modelling:

The alternative to digital simulation is analogue modelling. This is achieved in one of three ways;

- Simulating the Child-Langmuire law, much like the digital simulations.
- Simulating the "soft limiting" effect as described by Rutt [17].
- Simulating the harmonic content of the output signal as described by Hamm [5].

In 2004, Dimitri Danyuk published a paper describing a triode "emulator" with "a tube-like transfer characteristic, producing harmonic distortion components similar to a triode preamplifier." [48] He recognised the importance of the production of a dominant second harmonic, explaining its presence using the previously referenced binomial series as described by Morgan Jones [9]. A full description of the circuit operation can be found in his paper and thus is not included in this thesis. A surmise of his findings is presented.

Danyuk attempted to use the transfer characteristics of a junction gate field-effect transistor (JFET), whose workings are more closely related to that of a valve than a transistor, to achieve a corresponding three-halves curve by changing the value of the JFET source resistance. The two transfer characteristics can be found in figures 3.16. He successfully matched the law across "one-third of the range of input signal," producing an output which generated harmonic content closely matching the theoretical distortion products shown in figure 3.4.

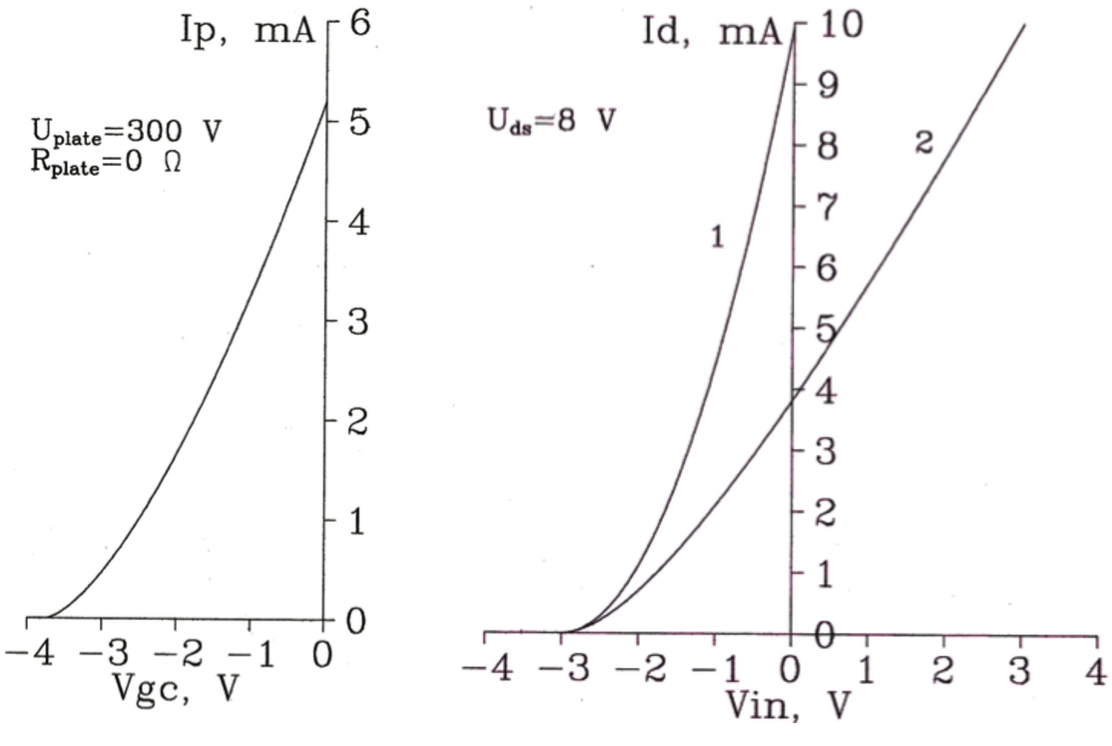


Figure 3.16 – Typical triode (left), and JFET (right) transfer characteristics. [48]

Although Danyuk's emulator successfully produced "tube-like" harmonics as a result of the JFETs ability to saturate relatively smoothly, it ultimately failed as an *accurate* emulator for much the same reasons as Leach and Koren's digital simulations did; it was based upon "the *static* transfer characteristics of the vacuum tube following the *thee-halves power law*." [48]

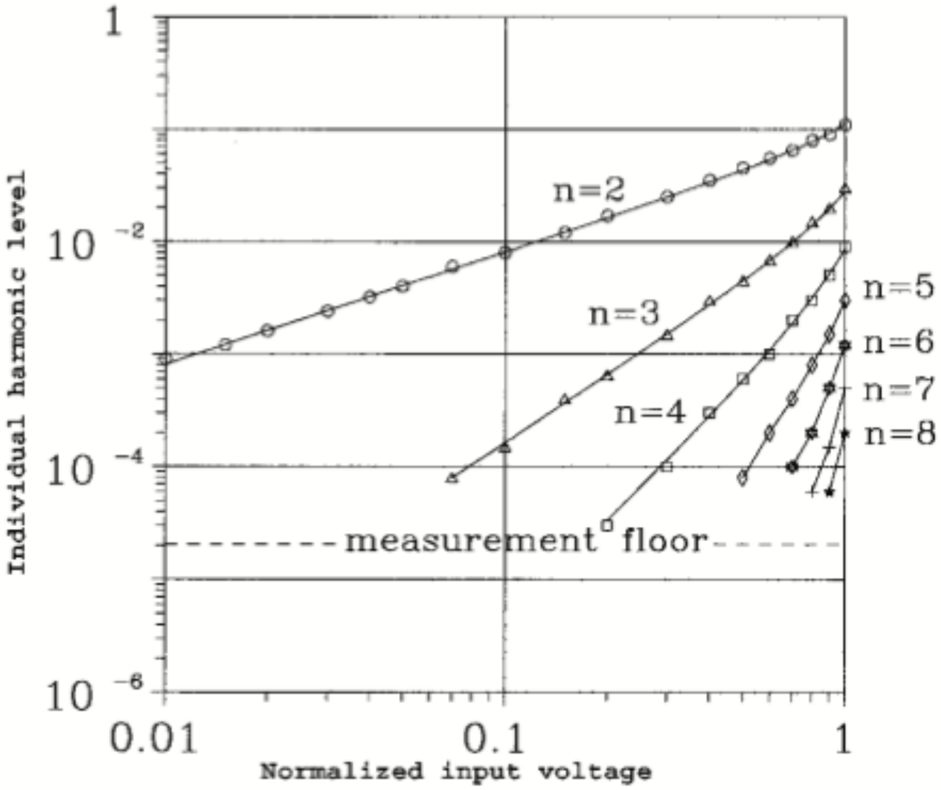


Figure 3.17 – Resultant measured harmonic content for the triode emulator. [48]

Aside from simulating the Child-Langmuire law, another common method of analogue valve simulation focuses upon the soft clipping of peak waveforms. This is done using the inherent forward bias voltage of a diode (0.7V) to gradually increase feedback in an operational amplifier, as figure 3.18 demonstrates.

Without the diode circuit the op amp is a unity gain amplifier, ie. Has an amplification factor of 1. When the input signal reaches peaks of 0.7V, D2 and D1 become forward biased, and R3 is introduced to the feedback circuit, reducing the stages gain. Likewise when the input voltage reaches peaks of 1.4V, D5, D6, D3 and D4, become forward biased switching in R4 to the feedback loop and further reducing the amplifiers gain. So it can be seen that as the input signal peaks reach increased levels the output signal is "soft limited" much like compression, and much like the characteristics of a valve.

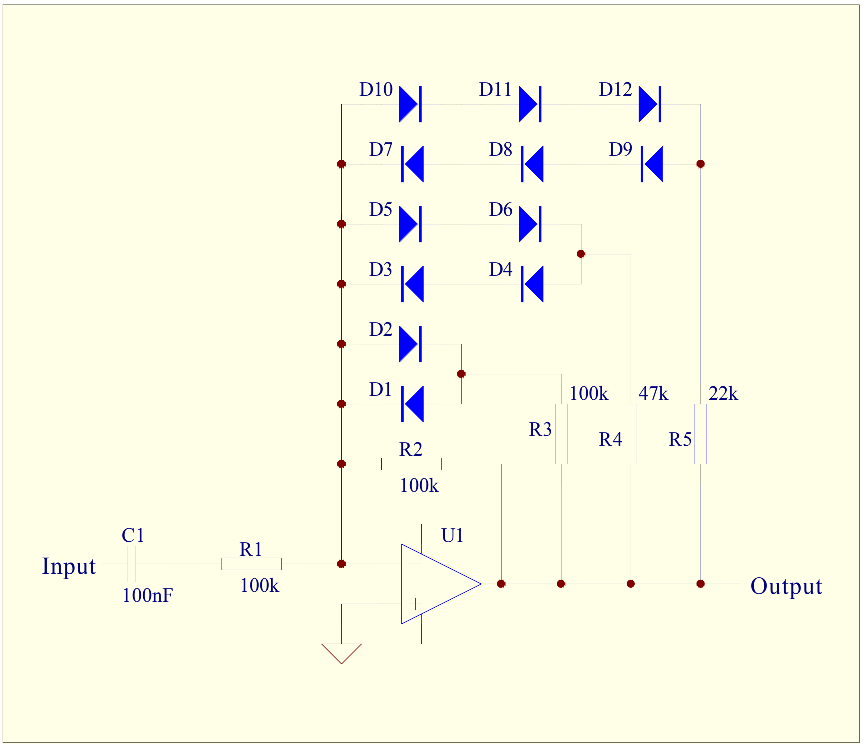


Figure 3.18 - Soft limiting circuit using diodes.

In order to achieve asymmetrical clipping a simple bias circuit needs to be introduced to shift the DC point of the input signal, doing so results in the selected positive or negative peaks to reach the 0.7 V threshold first. Unfortunately these designs fall short of the mark because the diode clip itself is inherently ‘hard’, so with prolonged peak voltages driving the stage into overload the output peaks will be flattened, figure 3.19. Ultimately flat peaks resemble a square wave, therfore the output harmonic content gains an increase in odd order harmonics.

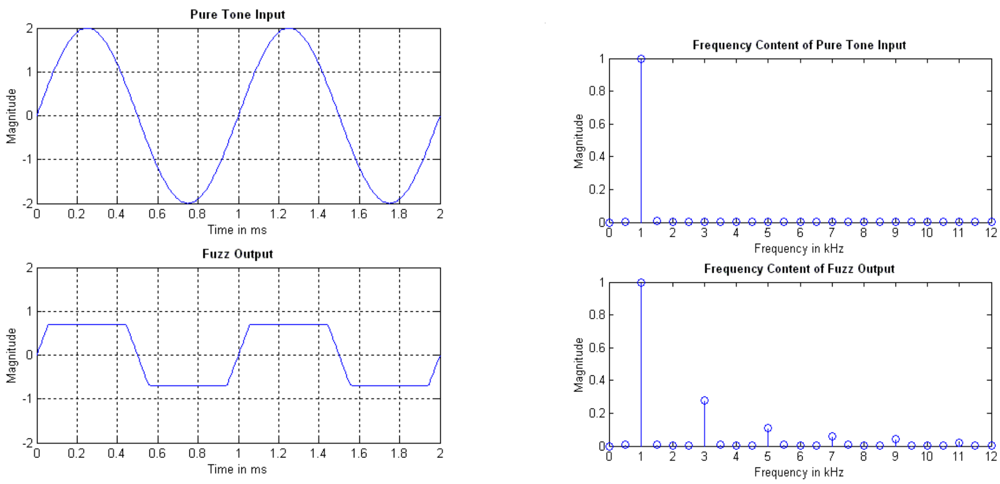


Figure 3.19 – 0.7V diode clip and its resultant frequency spectrum.

The third approach to analogue simulation is to specifically design a circuit in such a manner that, as Mintz describes, ”a particular ‘sound’ may be incurred or avoided at the designer’s pleasure no matter what active devices he uses.” [55] This is achieved through a series of experimental procedures of changing circuit values to alter the resulting output of an amplifier, until it produces the required harmonic content as defined by Hamm [5]. By losing the confines of the theoretical laws, the comeuppance of Danyuk [48], and striving to achieve circuit parameters which produce wanted output signal distortions, allows a less constrained approach of valve simulation.

Through this form of circuit experimentation Moneith and Flowers produced a circuit tailored to “produce harmonic distortion components comparable to, and perhaps more pleasing than, tube preamplifiers.” [54] Their paper entitled “Transistors Can Sound Better than Tubes” caught this author’s attention because of its acclaim of subjective accolade. Consequently this paper provided the starting point for the introduction of some original thought in to this thesis, resulting in the production of a published, and conference presented, white paper (ISBN: 978-1-86218-093-2), as is included in the subsequent chapter. Therefore this third form of analogue modelling is discussed in length in Chapter 4.

3.3 - Chapter Summary:

This chapter has presented research into valve modelling using digital and analogue implementations. It has been shown that both approaches, although producing certain sound signatures with parallels to the 'valve sound', are not yet able to accurately model the *dynamic* operation of a valve topology.

The computational demand of emulating Miller capacitances is currently the confining constraint preventing the progress of Cohen and Helie's [43] up-to-date dynamic model, although their inclusion of measured data parameters into Koren's [45] *static* model is producing useable VST plug-ins.

Analogue modelling is also falling short of producing an accurate replacement, although this approach does lend itself to the more *dynamic* modelling approach. This author would suggest that the overall success of analogue modelling relies upon a greater accumulation of data relating to the dynamic operation of the valve under varying test conditions. It is shown by Danyuk [48], that the performance of JFETs can be manipulated to match desired transfer functions, and by Lindley and Parulekar [53] that diode nonlinearities are being exploited to again produce useable if not entirely accurate distortion effects for guitar enthusiasts.

As a result of the reviewed literature, the objectives of the intended research for this thesis have been justified and can consequently be re-stated as being:

- To produce a wealth of data detailing the dynamic operation of the triode valve and its response to a range of increasing amplitude input signals.

- To extend this data accumulation to examine the effect on the dynamic operation of a valve when reducing the High Tension supply to the anode of a triode.

- To explore the possibility of using valves as opposed to valve models in low power applications for valve guitar amplification.

Chapter 4 - Can Transistors Sound Better than Tubes?

4.1 - Foreword:

The aforementioned Monteith and Flowers paper [54] clearly stipulated their belief that "transistors can sound better than tubes." As stated in the introduction, this thesis is based upon the premise of using the more advanced objective analysis techniques available today to expound theories from research previously carried out in to the characteristics of the 'valve sound'.

The Monteith and Flowers paper lends itself to this way of thinking, having been published by the Audio Engineering Society in 1977, there was no further work carried out into their circuit. In fact as Hamm noted in his comments on their paper;

"It is rather disturbing that the kind of work undertaken by Monteith and Flowers is not being supported in commercial circles. It seems to me there is a distinct lack of significant papers coming from companies with a commercial interest in audio." [56]

This "lack" of papers and discontinued line of research may well have been as a result of the technological direction of advancement in the late 1970s, very much focussing on the transistor as the way forwards. Valve electronics were seen at that time as being "in demise" [58], though, as has been discussed, this is not the case in the modern climate of the audio electronics industry.

Correspondingly the decision was made to construct the Monteith and Flowers preamplifier and a comparable valve preamplifier in accordance with the much-referenced ECC83/12AX7 dual triode preamplifier stage, and test the two using modern equipment.

4.2 - Preamp Comparison Methodology:

The circuit diagrams for the two preamplifiers can be found in Appendix 8.4. The Monteith/Flowers (M/F) preamp was constructed in a separate chassis to the valve preamplifier to ensure no stray AC signals from the valves heater supply would interfere with the results.

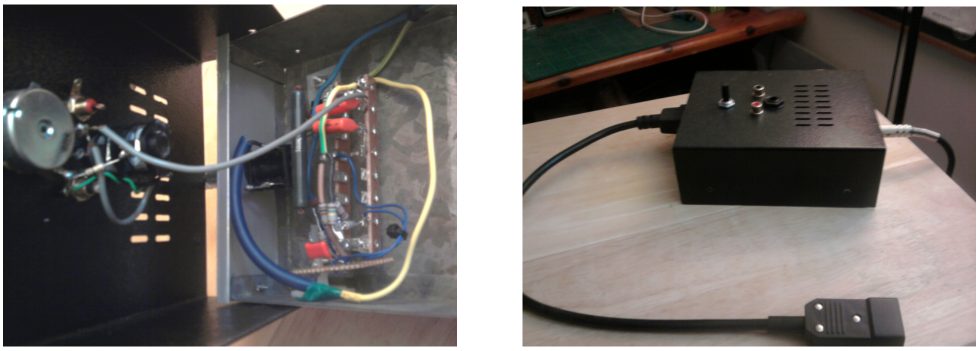


Figure 4.1 – Monteith/Flowers Preamplifier.

Before any testing could be undertaken, a somewhat lengthy process of Risk Assessment had to be completed in compliance with the University of Huddersfield regulations for use of high voltages within the electronics labs. The completed Risk Assessment form can be found in Appendix 8.5.

The two circuits were operated off the same smoothed power supply, figure 4.2; with the overall supply voltage controlled using a variac across the mains supply, figure 4.3. All needed points for measurement of high voltages were run to insulated test points on the fronts of both chassis to allow HT readings to be monitored without any physical interaction.

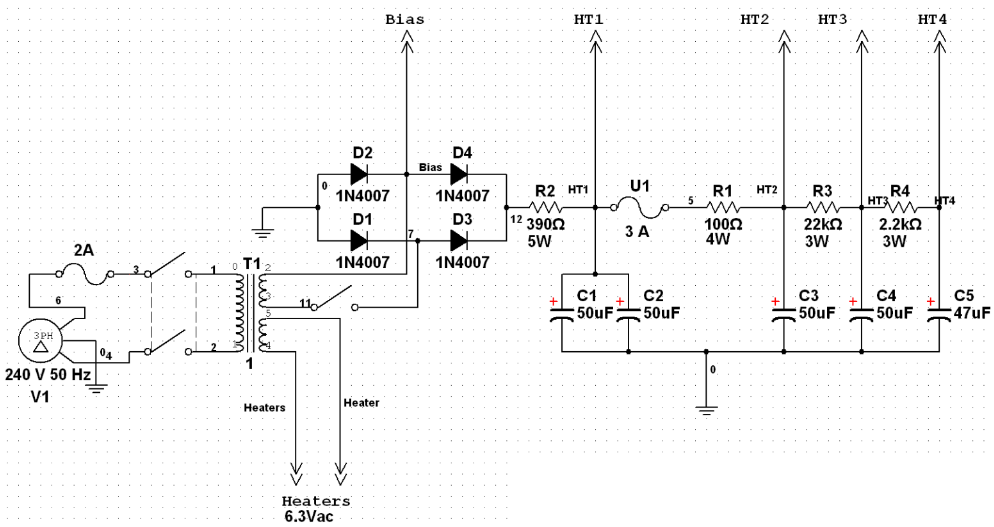


Figure 4.2 – Smoothed high voltage DC supply.

Solid- State				
Part	**Code**	**Value**	**No. Of**	**Cost (£)**
NPN transistor	MPS-A18	N/A	1	0.48
NPN transistor	MPS-U10		2	0.95
Resistor		1k	1	0.06
Resistor		1M	3	0.06
Resistor		150k	1	0.06
Resistor		10k	1	0.06
Resistor		100k	2	0.06
Resistor		120k	1	0.06
Capacitor		0.1u	3	0.1
			TOTAL	3.22

BOM (Bill of Materials) for the M/F preamp.

The test rig was set up as shown in figure 4.3 and data was collected as described in the following paper format.

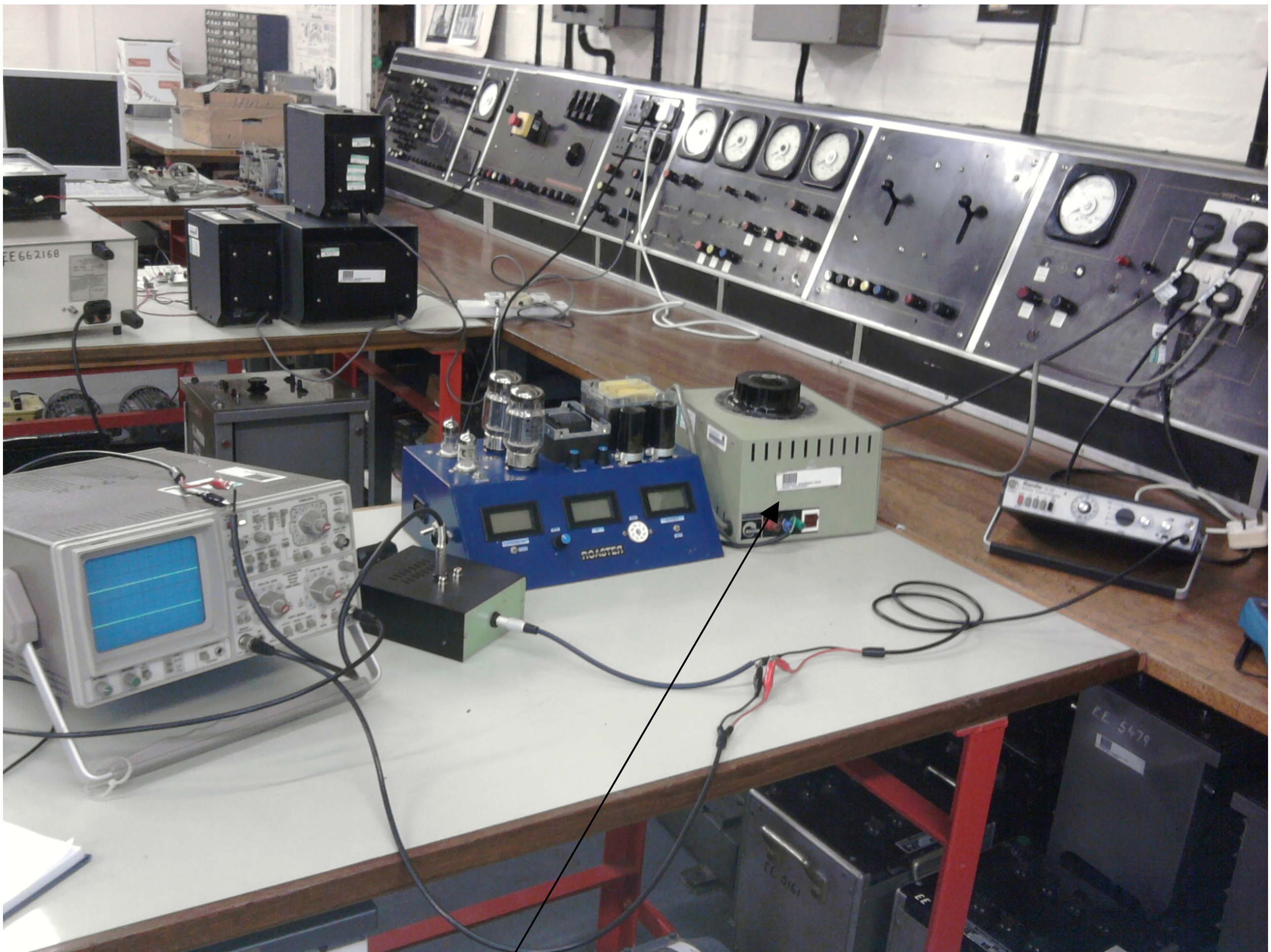


Figure 4.3 – Test rig setup with variac across the mains supply.

4.3 - Can Transistors Sound Better Than Tubes?

The University of Huddersfield published the following paper and the findings of the research were presented at the Computing and Engineering Annual Researcher's Conference 2010.

4.3.1- Abstract:

An objective comparison is made between a referenced high-voltage solid-state preamplifier with acclaims of large signal capabilities comparable to a valve amplifier, and an ECC83 based preamplifier topology. By analyzing the interaction of individual harmonic amplitudes throughout the amplifiers overload regions it is shown that there is correlation between the two systems signal outputs, but differences are observable. The paper describes the properties of the valve sound as having a dominant second order harmonic with an array of higher order harmonics producing the popular warmth of distortion often used for guitar amplification. The resulting dominance of 2nd and 4th harmonic components in the solid state system suggests that the sound could well be appealing to the listener, however the presence of prevalent higher order harmonics in contrast to the attenuated higher harmonics of the valve stage demonstrate that the two systems may sound different.

Keywords – Audio, amplification, valves, thermionic, tubes, guitar, harmonics.

4.3.2- Comparison of the Circuit With the Valve Sound:

Monteith and Flowers [54] have designed "a low-noise microphone preamplifier transistor circuit which has the same large signal capability as tube designs." Figure 4.4. They specifically mention that "this design exhibits the desirable overload characteristics of tubes" in accordance with the investigations of Hamm [32]. Hamm ultimately denotes that tube amplifiers react differently to transistor amplifiers in their regions of overload, particularly in the output signals interchanging dominance of odd and even harmonics when pushed up to 12dB into overload.

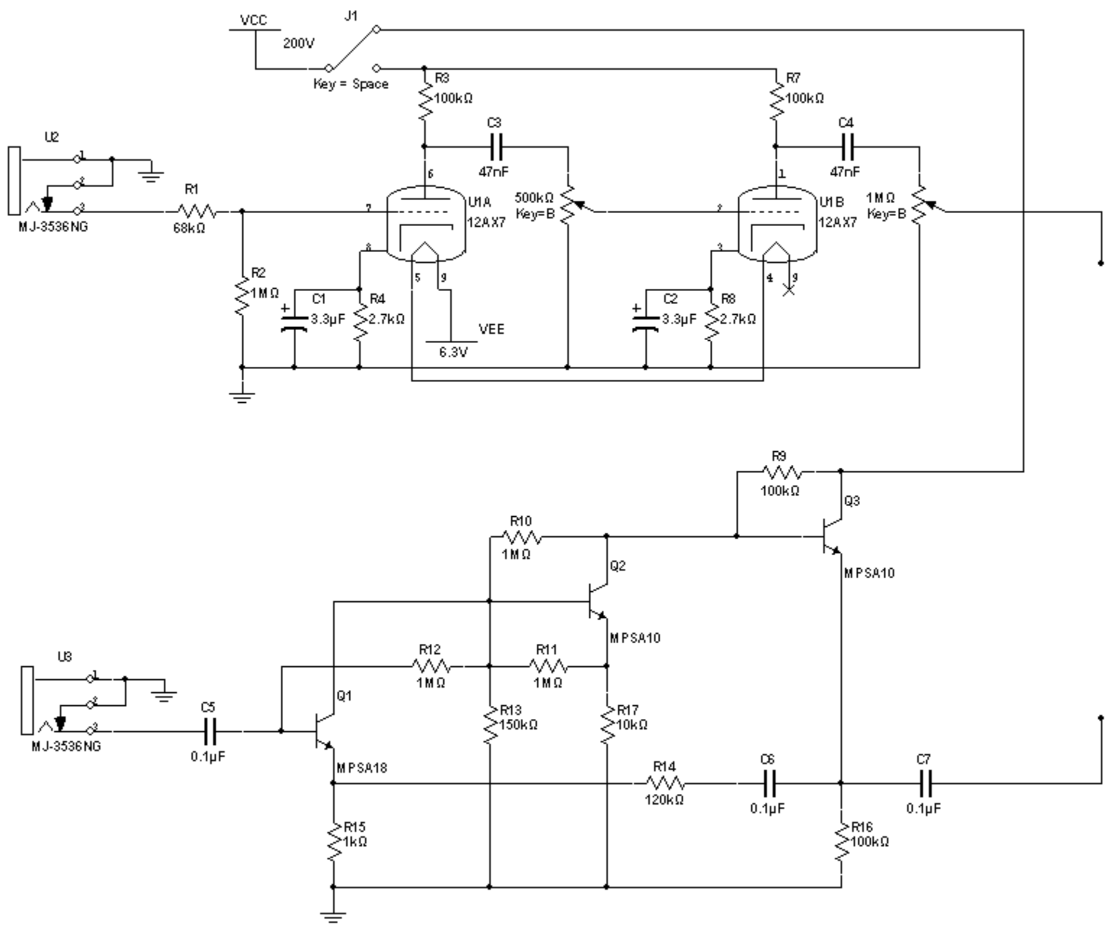


Figure 4.4 - Circuit diagrams for the DUT. [1]

The pretension of the Montieth/Flowers paper demonstrates the harmonic content of their systems performance approaching and passing through the point of output clipping showing the dominance of a 2nd order harmonic. It states that the circuit "produces harmonic distortion components which are comparable to, and perhaps more pleasing than, tube preamplifiers.'" [54]

Hamm's developed measurement technique [32] using Fourier analysis to study the percentages of ensuing harmonics in relation to the fundamental showed conclusively, figure 4.5, that triode preamplifiers "outstanding characteristic" was a particularly dominant 2nd harmonic in tandem with an initially dominant 3rd harmonic and an increasing 4th harmonic further in to overload.

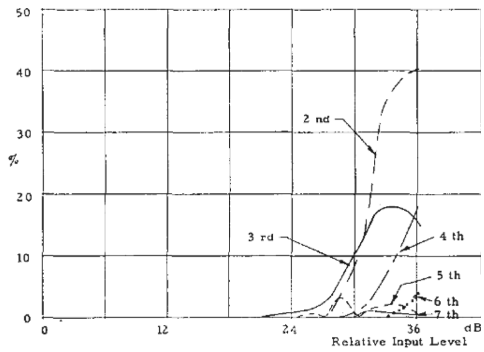


Figure 4.5 - Distortion components for two-stage triode amplifier. [2]

A comparison with the resulting plot of the Monteith/Flowers circuit, figure 4.6, shows that although there are some "harmonic distortion components which are comparable to" [54] Hamms plots, there are also some apparent differences; An ultimately more dominant 3rd harmonic increasing over 20% of the fundamental approaching maximum overload; A diminishing 4th harmonic falling to less than 5% of the fundamental as opposed to Hamms steadily increasing one; And a prominent 5th harmonic peaking at approximately 11% of the fundamental in contrast to Hamm's where the 5th, 6th and 7th harmonics all remain under the 5% line throughout clipping.

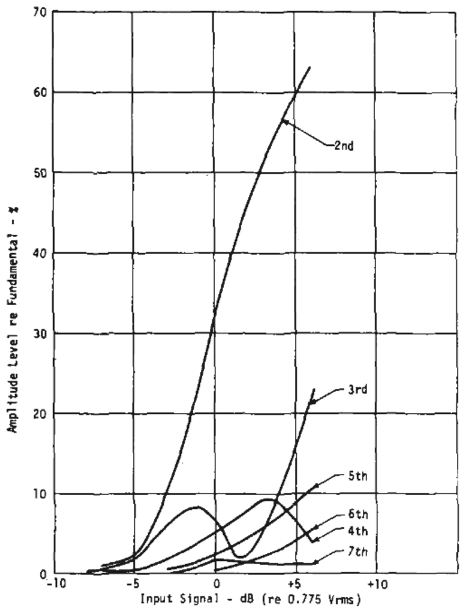


Figure 4.6 - Distortion components as function of input level for high-voltage preamplifier. [1]

Hamm's conclusion that "inaudible harmonics in the early overload condition might very well be causing the difference in sound coloration between tubes and transistors" [32] would suggest that these differences in harmonic content could well belie the fact that the Monteith/Flowers circuit can indeed sound better than a tube amplifier.

The other main characteristic identified by Montieth/Flowers as "tubelike" is an output waveform that displays asymmetrical clipping. This too was documented by Hamm, and is demonstrated by comparing the oscilloscope shots within each paper. It is also documented [57] that using Fourier analysis on complex signals can demonstrate that any waveform with a strong presence of 2nd harmonic will result in an asymmetric output, and as Mintz implies, "a particular 'sound' may be incurred or avoided at the designer's pleasure no matter what active devices he uses.'" [55]

It is the authors' belief that no conclusions should be made as to whether an amplifier system sounds "tubelike" or not by noting an asymmetrical waveform on an oscilloscope.

4.3.3- Guitar Versus High Fidelity Amplification:

The aforementioned papers were fundamentally focused towards high fidelity reproduction of sounds where accurate reproduction of signals is of utmost importance. This paper however is concentrated particularly on the use of valves in guitar amplification. Bussey and Haigler's paper [23] identifies the crucial difference between audio reproduction and guitar amplification in that, "In this application the amplifier becomes part of the musical instrument, and is frequently used to radically alter the signal from the guitar." [23]

Rutt [17] further studied into the use of valves for guitar amplification, paying particular attention to the preamplifier stage, almost exclusively triode nonlinearity, suggesting that the common ECC83 triode stage was perhaps the most commonly used stage in guitar preamplifier design. In conjunction with the usual test methods using single frequency test sources Rutt also based his research on the more complex guitar waveform constructed of many frequencies of sinusoidal wave.

Bussey and Haigler pointed out the difference in the way an amplifier responds to single plucked strings and chords specifically mentioning, "one subject felt that the difference between amps was an order of magnitude below the difference in striking the strings." [23] Rutt determines that a valves pleasing soft-limiting of signals is due to 'an induced voltage drop across the grid circuit source resistance' as a result of grid current, and that it is this soft/grid limiting that allows the small transient nuances produced by the higher frequency guitar strings to still be present at the output, giving greater harmonic detail to the sound.

With the emphasis of all the researched papers being on the harmonic content of signals it was proposed to investigate into the differing harmonics incited by both the Montieth/Flowers and an ECC83 based preamplifier, figure 4.4. Particular attention was to be paid to the region most exploited by guitarists in search of the fabled "warmth" of valve amplification, the overload region.

4.3.4- Experimental Procedure:

In keeping with the original paper [54] testing of the Montieth/Flowers preamp was carried out at a HT of 200VDC. The primary indicator that the circuit was performing as recorded by the original paper was to monitor its output waveforms in the overload region, figure 4.7.

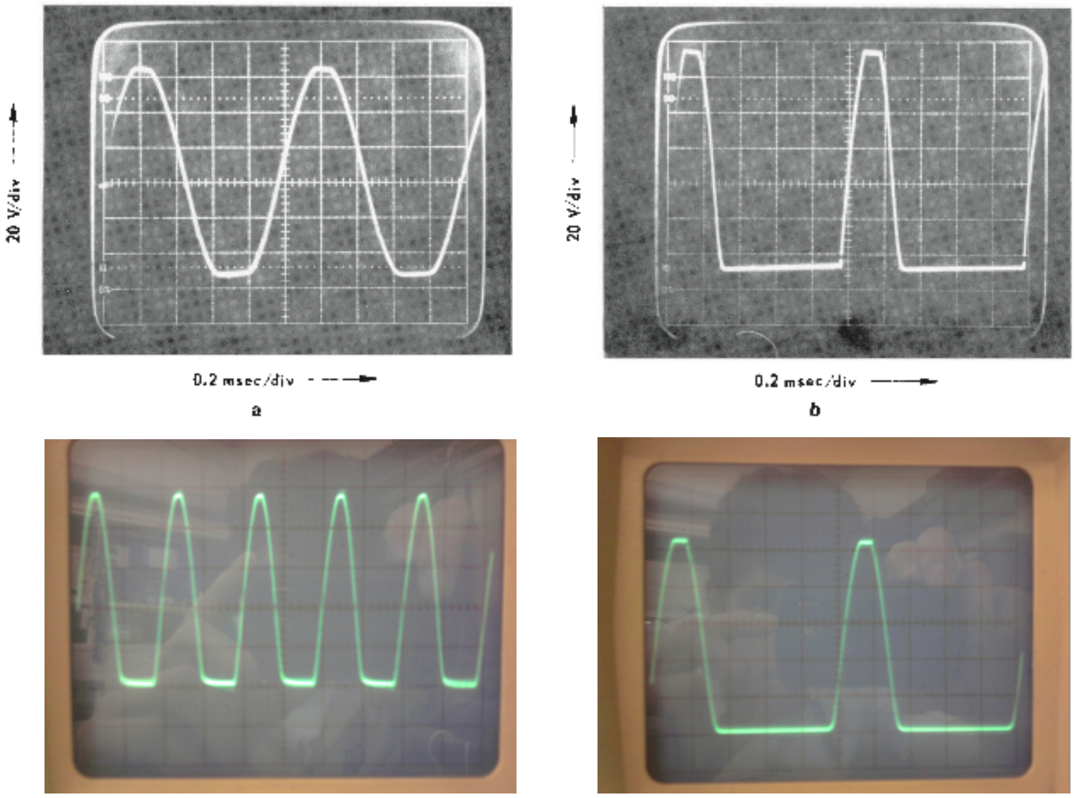


Figure 4.7 - Oscilloscope shots taken from original paper [54] and authors experiments.

Once satisfied that the circuit reacted to an input signal as expected, a systematic approach to increasing the input voltages from 10mV to 3.00V of a 1kHz sine wave was undertaken. At each incremental increase of 10mV the output was monitored and recorded both for its waveform shape and for its harmonic content using an Agilent 35670A signal analyser with its internal signal generator, thus reducing error and containing all testing within one piece of equipment. At each increment the HT supply was switched to the valve stage and adjusted using a variac at source to maintain $200V_{DC}$ and the same measurements were taken.

FFT plots were derived from data taken using the 35670A signal analyzer FFT function, the settings of which were 2 channel mode, 400 point resolution and a Flattop window function [71] [72] then saved to disk. All files were converted from .DAT to .csv and placed in to Excel for analysis. Readings of output voltage and total harmonic distortion were also taken giving a wealth of data from which to derive plots.

It was decided that the only fair way to ascertain if there were indeed similarities between the two preamplifiers was to focus upon their region of overload. As such further measurements were taken to marry up THD% readings for both stages, culminating in a further set of data as to what input voltage produces what percentage of output THD, figure 4.8.

THD%	Input to M/F mVrms	Input to Valve mVrms
0.500	306.000	10.000
1.000	315.000	18.500
2.000	380.000	22.700
4.000	466.500	28.800
6.000	493.000	36.300
8.000	513.000	46.700
10.000	531.000	62.800
12.000	548.000	89.700
14.000	568.000	134.400
16.000	587.000	199.600

Figure 4.8 - Equivalent input voltages for required THD% at output.

4.3.5- Experimental Results:

Purely taking into consideration FFT analysis of the two systems at maximum overload, with a 3.00V input, figure 4.9, the similarities between the two output signal characteristics are visible.

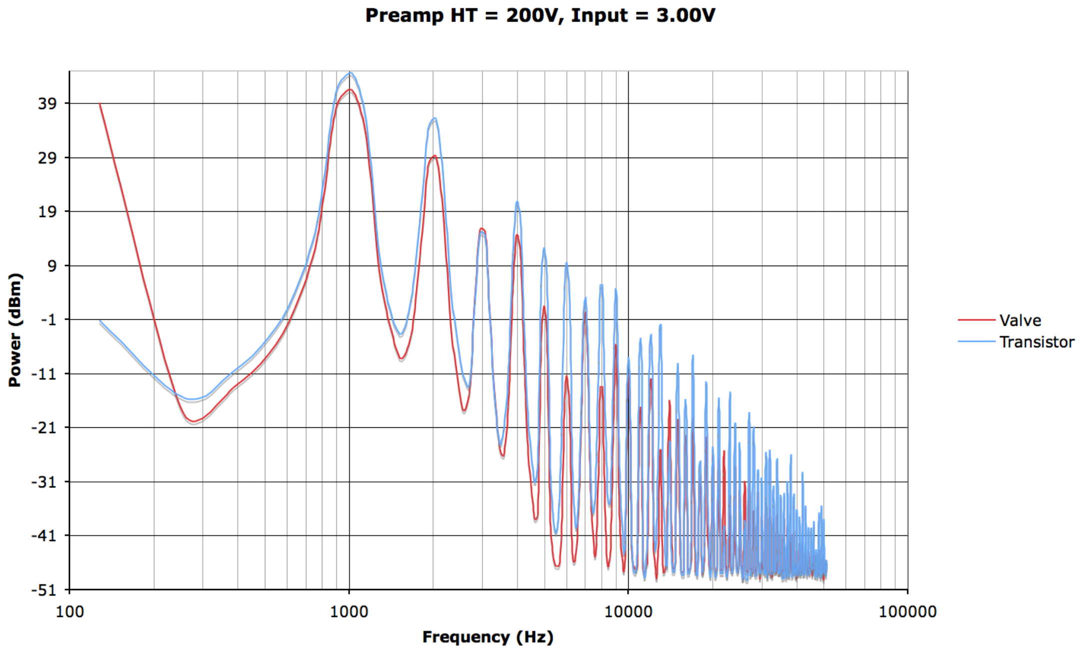


Figure 4.9 - Signal output FFT analysis of a 3Vpp Input to both preamps.

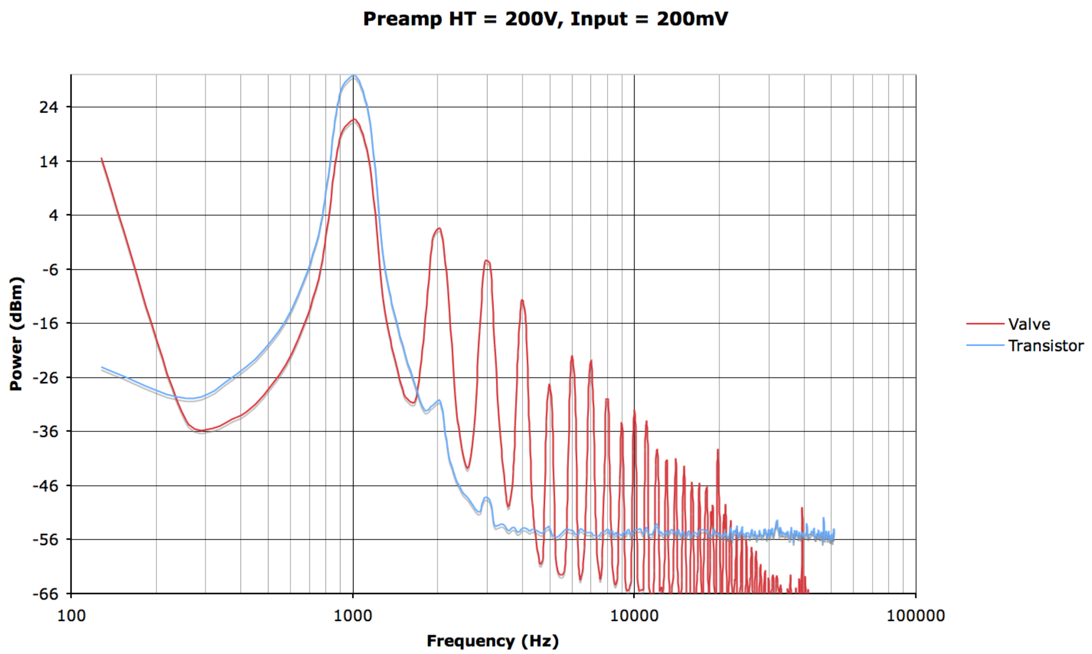


Figure 4.10 - Signal output FFT analysis of a 3Vpp Input to both preamps.

The majority of the differences are noticeable during the lower input voltages, figure 4.10 (The full range of output spectrum can be found in Appendix 8.6). For the Monteith/Flowers stage the 2nd harmonic remains below 1% of the fundamental amplitude up until a 90mV input signal is applied. In stark contrast to this the valve stage 2nd harmonic reads at 7% of the fundamental at 10mV input, with prominent 3rd, 4th and 5th harmonics from the outset.

Initial plots of harmonic amplitude in dBm (reference to 1mW into 600Ω) against output level in dBu (reference to 0.775VRMS) over the same scale provide evidence that the Monteith/Flowers has a much sharper knee than the valve stage, resulting in a smaller range of overload and producing different distortion and transfer characteristics. These plots also clearly demonstrate the soft clip properties of a valve. However, as expected, it is the THD% against harmonic amplitude plots that provide the best insight to the differing systems characteristics, figure 4.11.

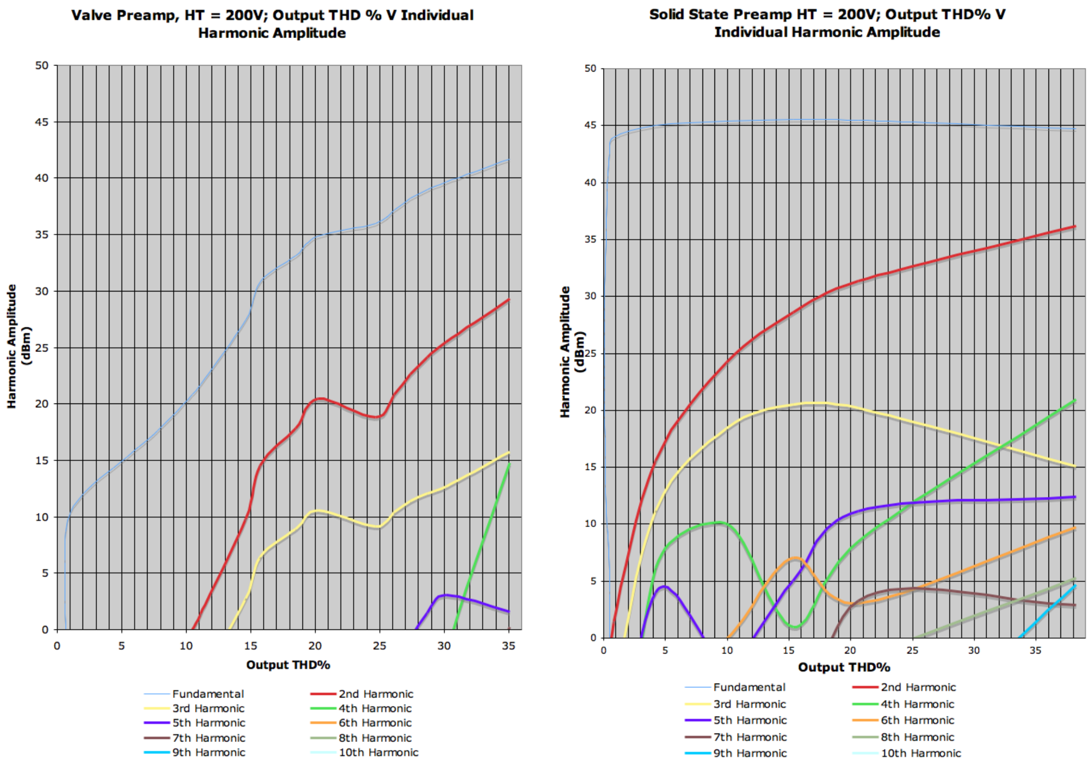


Figure 4.11 - Individual Harmonic Amplitude V Total THD% of each system.

Both systems produce dominant levels of 2nd order harmonic, however unexpectedly it is the valve stage whose 3rd harmonic is most prevalent between 25-35% THD. This appears to contradict Hamms [32] results. Likewise the Monteith/Flowers plot contradicts their findings, Fig. 3, in that the 3rd harmonic apparently attenuates as the THD levels at output increase. The other notable difference for this stage is that of the 4th harmonic which actually ends being the second most dominant harmonic at maximum clip.

4.3.6- Conclusions:

This paper concludes that despite the even order harmonic dominance of the output of the Monteith/Flowers design, the overall harmonic component density, particularly in the upper octave ranges, differs quite significantly to that produced by the valve pre-amplifier under test. There is therefore a marked difference between the two preamplifier topologies with particular respect to THD% vs. higher order harmonic amplitudes.

If Hamm's theory that the valve sound lies in the subtle differences of upper order harmonics is correct then the over crowded higher order harmonics of the Monteith/Flowers in comparison to the valve stage may well prove detrimental to its sound as perceived by the listener.

4.3.7- Further Work:

It must be noted that many other properties of amplifier stages such as their frequency response, transient response, and phase response all play key roles in shaping the sound of any system. Hamm's comments [56] on the Monteith/Flowers paper produces probably the most valid point of all research related to the area of audio, "they present no phychoacoustic data from real-live people that says, conclusively, that their amplifier lives up to the title of their paper." [56] With this in mind, the immediate further work to allow more conclusive evidence as to which system sounds better is to enforce the objective measurements of this paper with subjective measurements from a series of listening tests.

4.4 - Subjective Support of Objective Measurements:

As the paper concluded, the measured objective results needed subjectively analysing to give a true and fair comparison between the two systems. This subjective analysis would satisfy Hamm's comments and agree with Danyuk's [48] suggestions that "in the end, the listener should judge the measure of sound colouration." However the noticeable differences between the two systems output covered too large a range to allow an unbiased audio evaluation.

As much of the research literature suggested the 'valve sound' becomes most pleasurable when operating in the region of overload. As such it was the authors opinion that the preamplifiers regions of overload should be closer scrutinised. The transfer characteristics of the two systems needed to be compared, and the variables between the systems needed to be reduced to generate a more specific research question for an audio evaluation.

4.4.1- Further Analysis of Results:

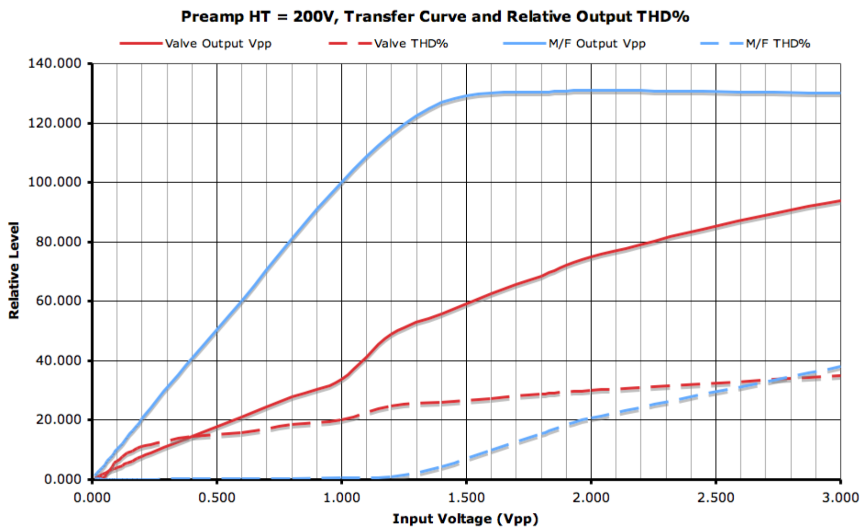
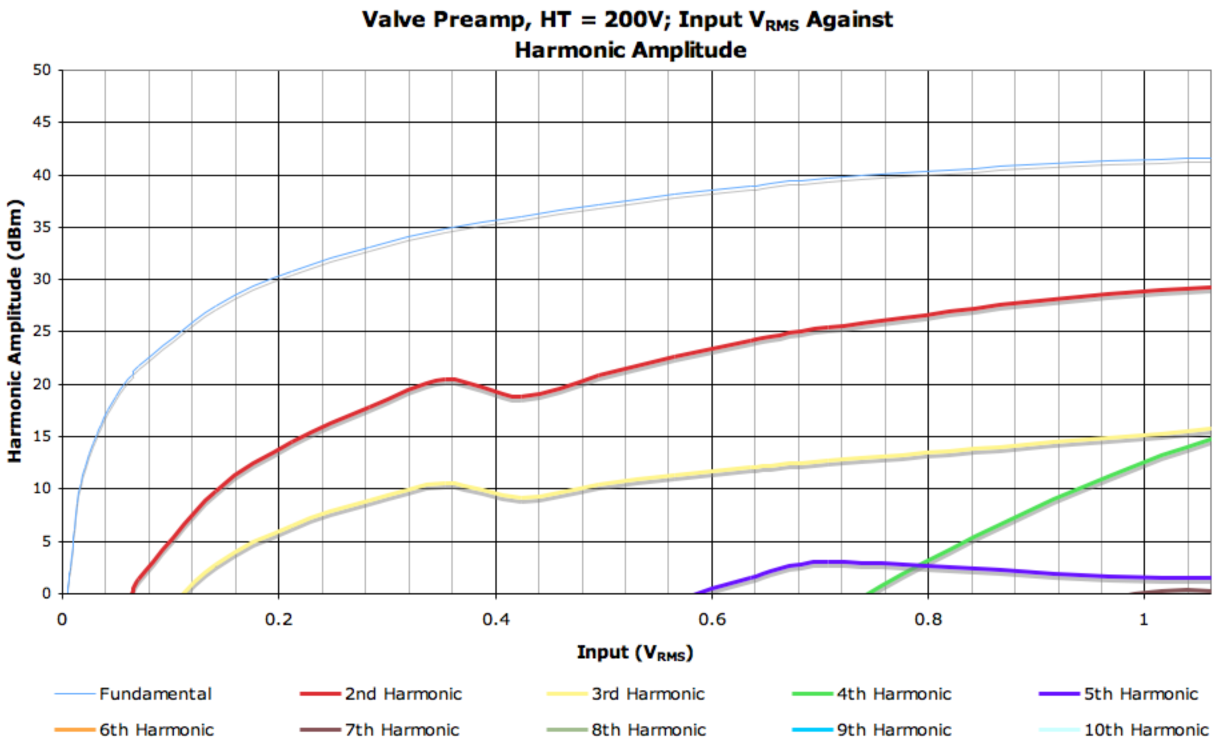


Figure 4.12 – Comparative transfer curves and relative THD%.

(Individual full-scale plots can be found in Appendix 8.7)

As figure 4.12 demonstrates, the M/F circuit reaches saturation long before the valve circuit as a result of its greater amplification factor of 100 compared to the valves 40. This produces a sharp knee, or a "hard limit", of any input signals greater than 1.5 V peak-peak ($0.53V_{RMS}$). The effect of this "hard limit" is demonstrated by figure 4.13. The M/F preamps linear region, from 0 – 1Vp-p, (relating to 0 – $0.35V_{RMS}$ in figure 4.13) clearly produces very undistorted output signal with very little generation of harmonic content. However once past the 'knee', harmonic distortion components are rapidly generated.



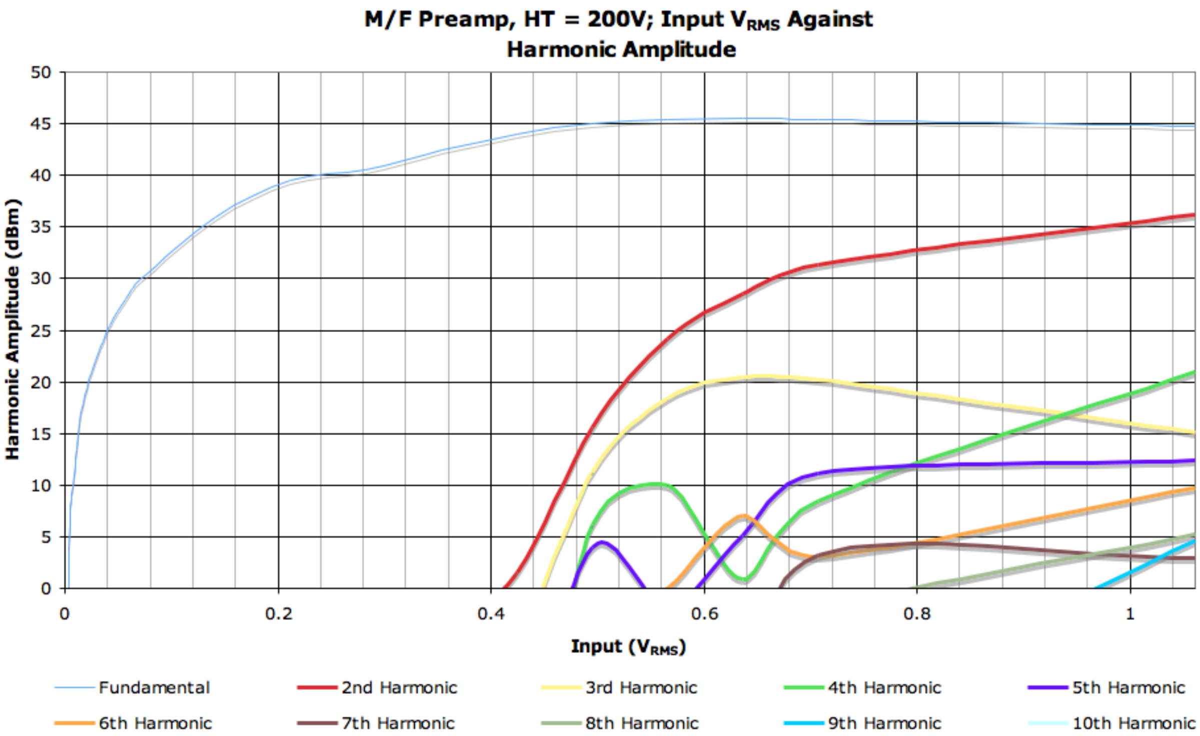


Figure 4.13 – Comparison of Output Harmonic Content at $200V_{DC}$.

In contrast to the M/F output, the valve displays the expected trait of low order harmonic generation throughout the range of input signals, with 2nd order being the most predominant. Further into overload and the M/F output shows a greater overall harmonic component density as described in 4.3.6, and it is this difference that requires closer inspection.

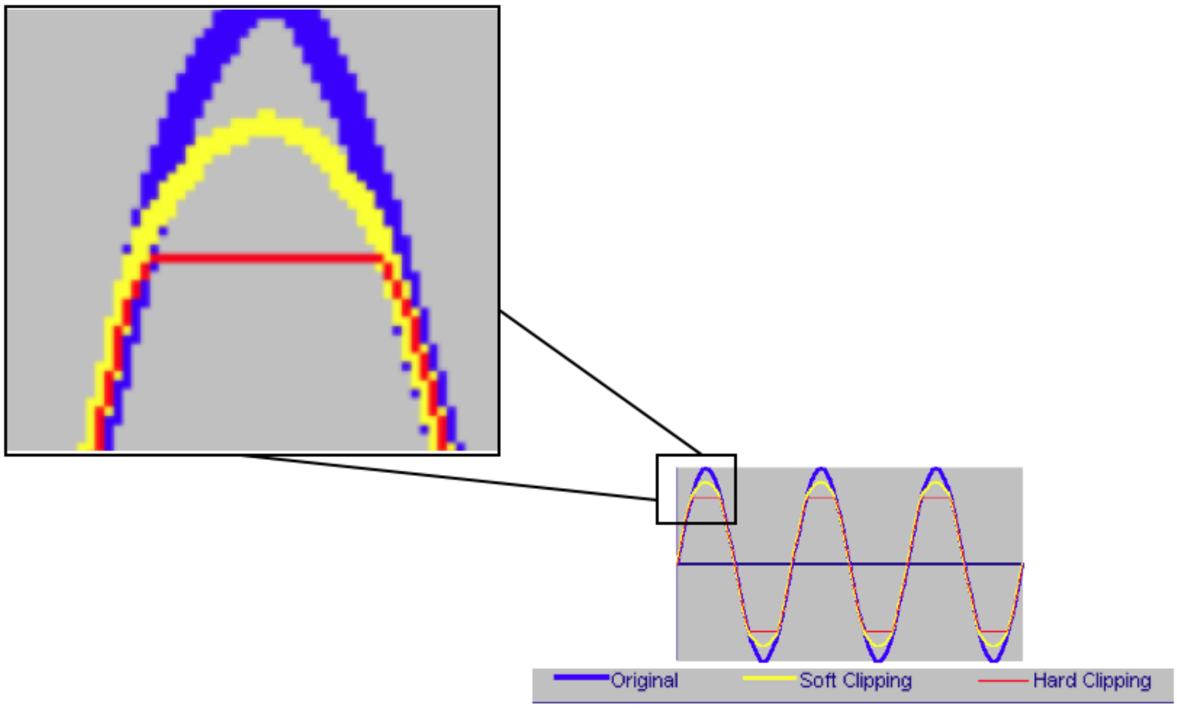


Figure 4.14 – Studying the knee of the transfer curve.

The decision was made to take a further large set of measurements to determine what value of input signal produced a set value of output THD%. Using the internal signal generator of the Agilent 35670A spectrum analyzer, input signal amplitude was gradually adjusted, giving time for each preamplifier to adjust and "settle" its operating parameters, until the measured output THD% was matched. Figure 4.8 displays the collected data, however the paper did not include appropriate analysis.

Figure 4.15 shows the similarity between the two preamplifiers output Fourier Transforms when their THD%s have been matched. The full range can be found in Appendix 8.8. The low order harmonics can be seen as having closely correlating amplitudes and bandwidths.

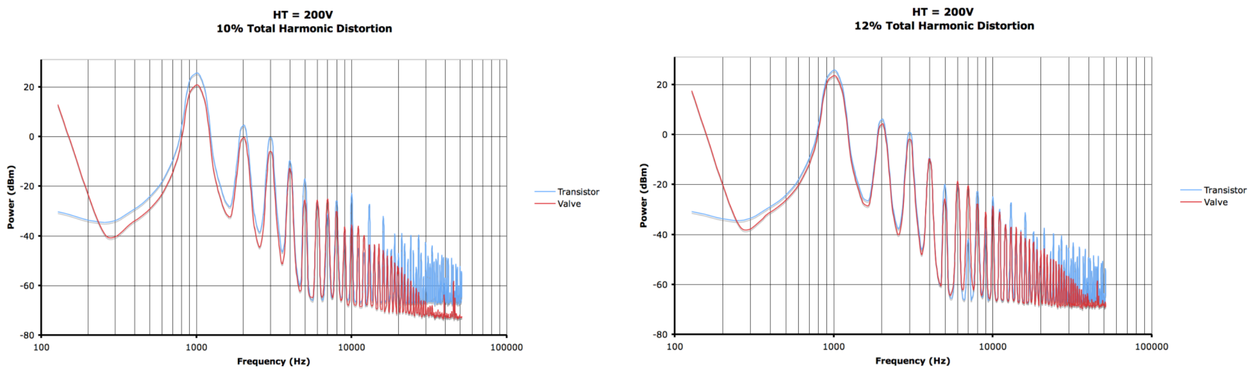
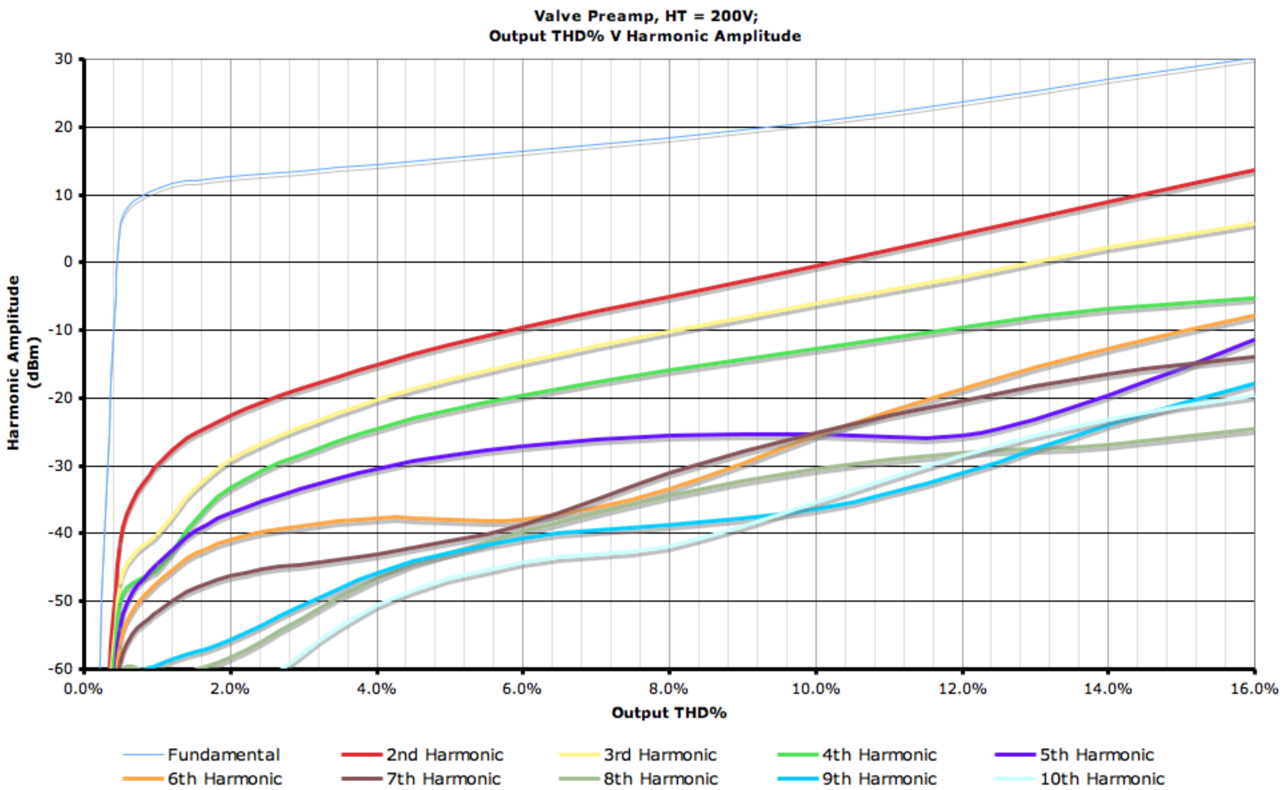


Figure 4.15 – FFT of both systems outputs at 10% and 12% THD.



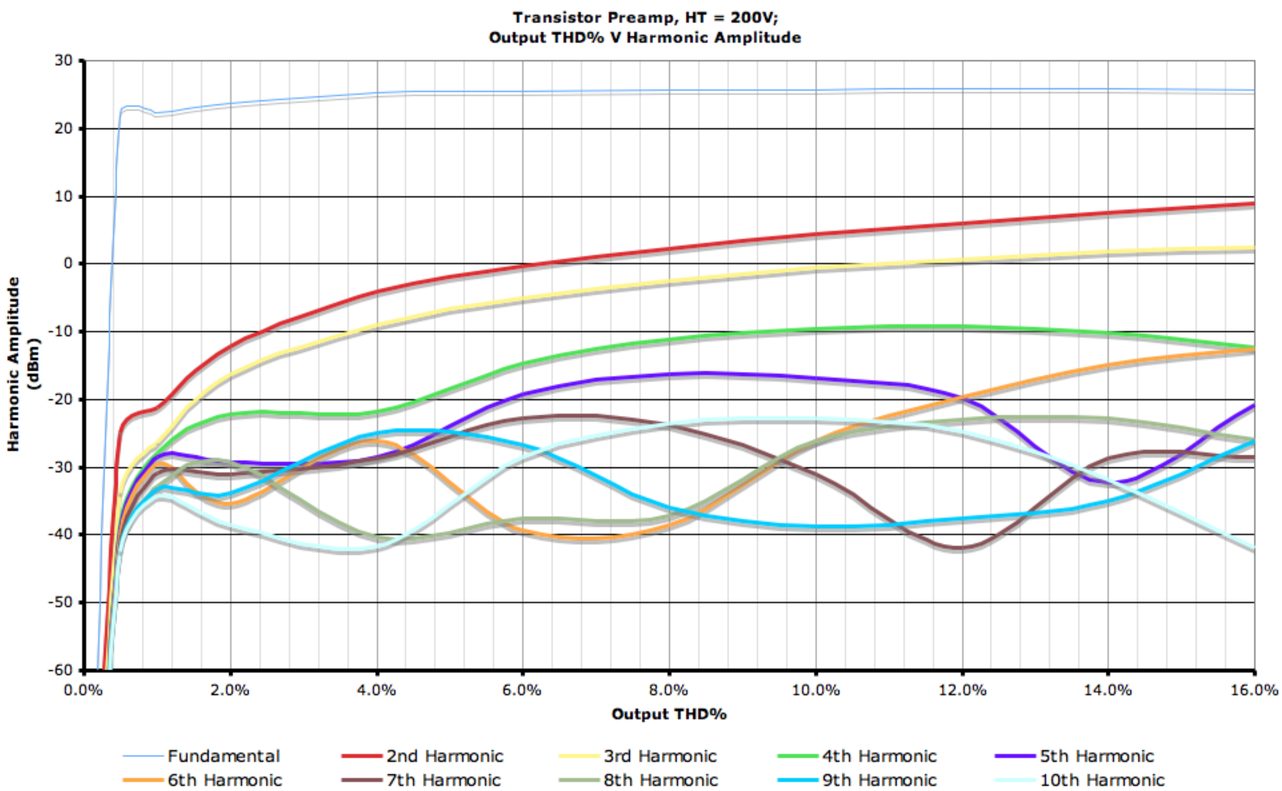


Figure 4.16 – Matched THD% Comparison of Output Harmonic Content at $200V_{DC}$.

In figure 4.16, the amplitudes of the fundamentals correlate to the 'knee' of the transfer curve nicely, reaching peak fundamental amplitude at approximately 0.5%THD. Both systems were now being tested at the point when the preamplifiers were saturated by the fundamental and beginning to produce increasing amplitude of harmonics. The harder limit of the M/F topology is still apparent judging by the rate at which its harmonic amplitudes increase; M/F 2nd harmonic amplitude has an increase of 40dBm per 1%THD, whereas the ECC83 has a slower increase of 40dBm per 2%THD. This greater speed of increase in amplitude is apparent in all the M/F harmonics.

Hamm articulated; "the primary colour characteristic of an instrument is determined by the strength of the first few harmonics," and also explained that "the ear seems very sensitive to the edge harmonics." [32] This suggested that closer inspection was required of the low order harmonics, but that the difference in the weighting of the higher order harmonics was likely to be a perceivable audible difference. The valves relatively uniform increase in harmonic amplitude of approximately 26dBm per 10%THD across harmonics order 1-10 could be speculated as aiding in the perceived "smooth" sound of a valve. The lesser M/F 10dBm per 10%THD, along with the interchanging dominance of its harmonics may result in a perceived "raspy dissonant quality." [32]

The individual harmonic amplitudes in dBm, (relative to 1mW into 600Ω) were converted into relative amplitudes in mW, (dBm to watts = $10^{(dBm/10)}$) then calculated as a percentage of the fundamental amplitude to produce plots with comparable axis to those used by Hamm and Monteith and Flowers. These plots display the correlation of the "first few harmonics" [5] between the two topologies, figure 4.17. The full-scale version along with magnified comparisons of the 2nd, 3rd, 4th, and 5th harmonics can be found in Appendix 8.9.

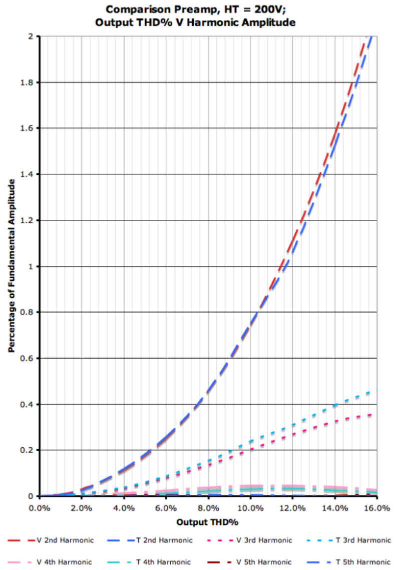


Figure 4.17 – Comparative "Hamm" plots of low order harmonics as percentage of fundamental amplitude.

Initially the 2nd harmonic of the valve amplifier is more dominant than the Monteith/Flowers. Between the regions of 3-8%THD the Monteith circuit has a slight increase in 2nd harmonic until 10.4%THD and upwards where the valve 2nd harmonic is the stronger presence again. The shapes of the two plots are very well correlated.

The Monteith/Flowers shows some relation to expected transistor technology harmonic production, with the 3rd harmonic being the more prominent of the two topologies throughout the whole range from 0-16%. Both stages show signs of harmonic reduction from 14%THD with the valve stage demonstrating a more severe roll off resulting in a 22.35% difference between the two harmonic amplitudes at 16%THD, the valve being the comparative lesser of the two.

The shape of the curves for the 4th harmonic again demonstrate similar characteristics between the two pre-amplifiers, but ultimately the valve stage has a greater amplitude in 4th harmonic than the Monteith/Flowers. With the knowledge that even harmonics produce "choral" or "singing" sounds it could be predetermined that the valve stages more prevalent even order harmonics, 2nd and 4th, would result in the valve sound being preferable in this instance.

However, when analyzing the 5th harmonic components, despite there being a dominance of the Monteith/Flowers 5th from 6.7-12.8%THD it is the valve stage whose 5th harmonic at 16%THD is 69.1% larger than that of its emulator, possibly "covering" or "masking" any "pleasant" perceived sound as a result of its 4th and 2nd harmonic.

With closer inspection of the objective measurements complete, and some speculative hypothesise made as to possible perceived subjective impressions of the two amplifiers, the project moved in the direction of developing a suitable and unbiased audio evaluation.

4.4.2- Audio Evaluation Research for Test Design:

Research Question: ""In relation to the reference, how much do you prefer the samples whilst comparing their distortion characteristics; More, less or the same?"

Hypothesis: The perceived distortion characteristics of an ECC83 valve preamplifier will be more pleasing to the listener than that of the Monteith and Flowers topology when run at 200VDC.

It must be stated from the outset that the aim of conducting this auditory evaluation was not to produce objective results but to produce a set of data reproducible in another laboratory. The aim was to acquire some subjective backing to objective results already obtained from both preamplifier stages using scientific evaluation.

It must also be stated that the audio evaluation can only be considered a 'pilot' because only eleven people gave their time to take the test.

It is the authors' belief that the attributes as outlined in ITU-R Recommendation BS.1116-1 [59] provide grounds for a suitable perceptual model for analysing basic audio quality. "This single, global attribute is used to judge any and all detected differences between the reference and the object." [59] However, as suggested by Bech [60] "This recommendation is intended for use in the assessment of systems that introduce impairments so small as to be undetectable without rigorous control of the experimental conditions and appropriate statistical analysis. If used for systems that introduce relatively large and easily detectable impairments, it leads to excessive expenditure of time and effort and may also lead to less reliable results."

In the case of this audio evaluation, initial objective tests showed that the two systems exhibited "intermediate levels of quality" [61] and as such ITU-R BS.1534-1 (Appendix 8.11), more commonly known as MUSHRA (Multiple Stimulus Hidden References and Anchor) was regarded as more appropriate. "The MUSHRA test method uses the original unprocessed programme material with full bandwidth as the reference signal (which is also used as a hidden reference) as well as at least one hidden anchor." [62] By having a hidden reference the validity of each subject's responses could be determined, and using a hidden anchor helped in the statistical evaluation of the results, reducing possible contraction bias [61] of listeners being afraid to use the extremities of the scales provided, figure 4.18.

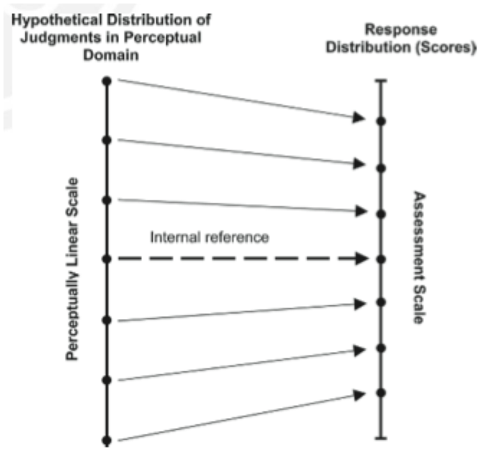


Figure 4.18 – Contraction bias Model. [61]

Despite there being numerous scales stated in various ITU standards, it was decided to utilise a "polar scale", an idea proposed by Watson [63], using "a vertical line 20mm long with no labels other than a + sign at the top and a – sign at the bottom, to indicate the polarity of the scale." In this case using a positive and negative labelled axis to demonstrate whether the sample had more or less preferable distortion characteristics to the reference. An interesting side note about the use of an unlabelled scale is that it allows the test to translate without bias across all languages, figure 4.19 explains.

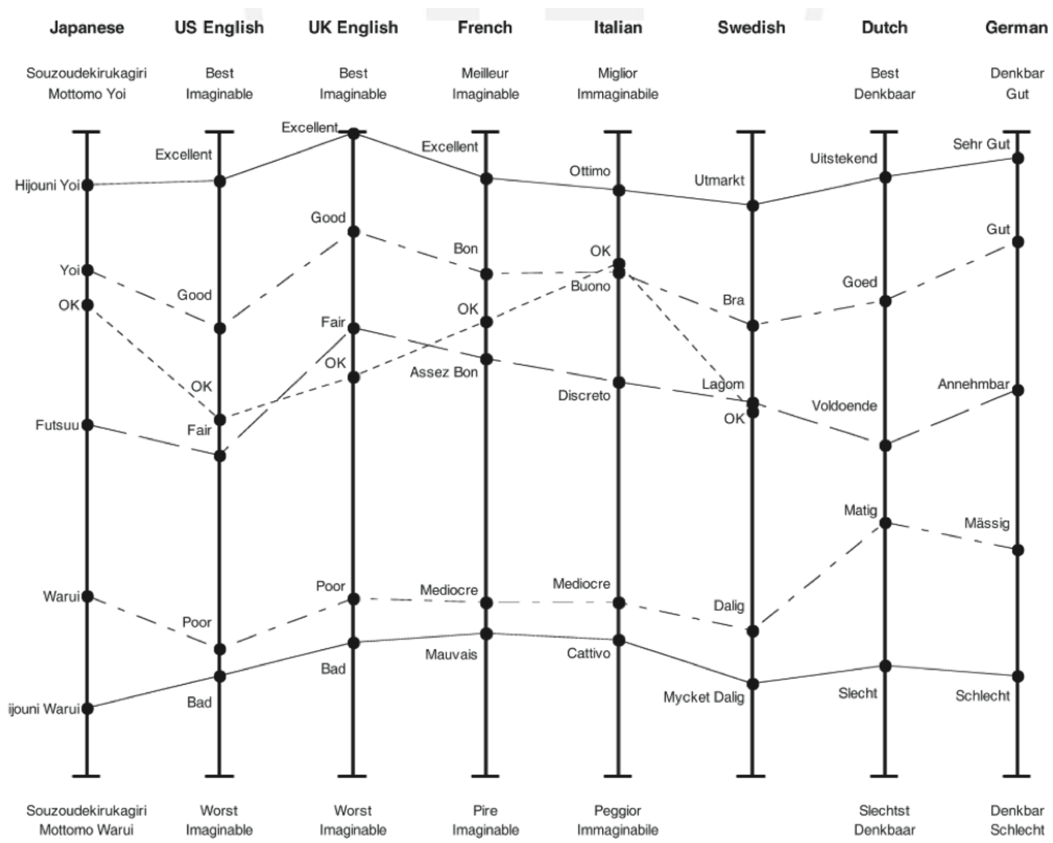


Figure 4.19 – Scaling of Labels in Accordance with Language Perception of Terms. [61]

4.4.3- Sample Recordings:

Two differing samples were recorded of a dry guitar signal using a DI (direct input) box. The first sample consisted of a riff of single picked notes; the second sample combined single picked notes with chords.

The recorded samples were then measured and adjusted in level accordingly using Matlab and SoundForge to provide average RMS levels at equivalent dBFS (relative to full scale of a digital signal) to the 1kHz sine waves used to achieve set levels of THD% at output in the objective measurements.

HT = 200V Monteith/Flowers
0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	306.000	-26.021	432.749	1638.400	0.050	2.970
1.000	315.000	-25.769	445.477	1686.588	0.051	2.965
2.000	380.000	-24.139	537.401	2034.614	0.062	3.520
4.000	466.500	-22.358	659.731	2497.757	0.076	4.172
6.000	493.000	-21.878	697.207	2639.644	0.081	4.297
8.000	513.000	-21.533	725.492	2746.729	0.084	4.387
10.000	531.000	-21.233	750.947	2843.106	0.087	4.429
12.000	548.000	-20.959	774.989	2934.128	0.090	4.467
14.000	568.000	-20.648	803.273	3041.213	0.093	4.497
16.000	587.000	-20.362	830.143	3142.944	0.096	4.518

HT = 200V Valve Stage
0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	10.000	-55.735	14.142	53.542	0.0016	0.437
1.000	18.500	-50.392	26.163	99.054	0.0030	0.824
2.000	22.700	-48.615	32.103	121.541	0.0037	0.973
4.000	28.800	-46.547	40.729	154.202	0.0047	1.226
6.000	36.300	-44.537	51.336	194.359	0.0059	1.572
8.000	46.700	-42.349	66.044	250.043	0.0076	2.066
10.000	62.800	-39.776	88.813	336.247	0.0103	2.853
12.000	89.700	-36.679	126.855	480.276	0.0147	4.252
14.000	134.400	-33.167	190.070	719.611	0.0220	6.530
16.000	199.600	-29.732	282.277	1068.708	0.0326	9.840

Figure 4.20 – Conversion Calculation for Sample Level dBFS at HT = 200V.

Each sample was adjusted to the calculated input dB value in SoundForge, then run through a standard deviation program in Matlab to obtain the average full-scale (digital) sample level of the excerpt. These samples then became the signal source to the two preamplifiers

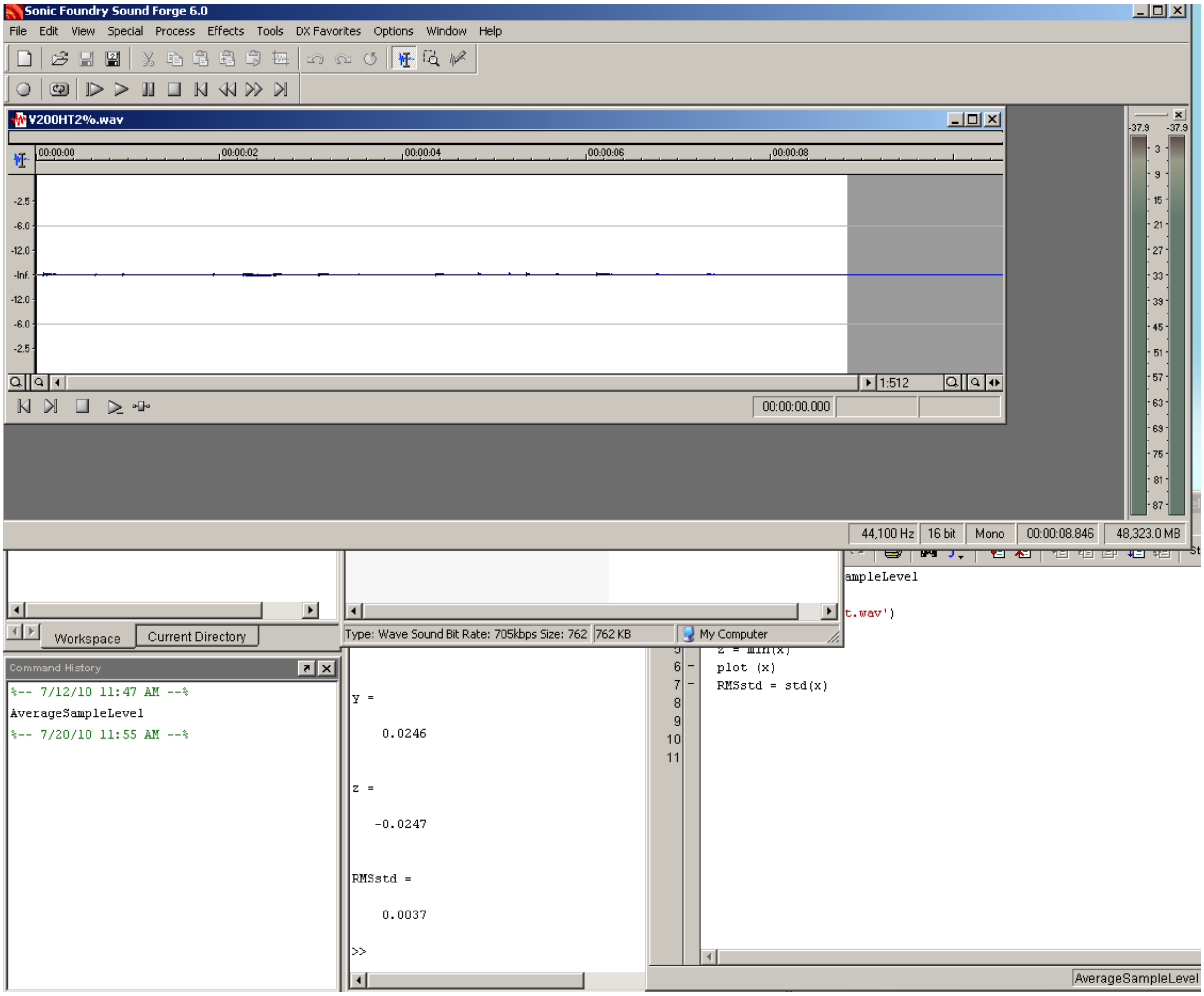


Figure 4.21 – Input signal for valve stage 2%THD showing the required 0.0037 double-value for the standard deviation of the waveforms sample level.

4.4.4- Audio Evaluation Methodology:

The level adjusted samples, one of length 7.00 Seconds, and one of length 9.00 seconds, were 're-amped' out of a Digidesign 003 rack to the inputs of the valve and M/F stages. There was inherent measured noise of amplitude 32.7mV$_{RMS}$ from the output of the 003. Due to the low levels of signal amplitude required for the lower percentage harmonic distortions, this noise, seen at the input to the two preamplifiers, was not negligible. The signal to noise power ratios were calculated and plotted, figure 4.23, full-scale in Appendix 8.13. The outputs of each system were then recorded at 16 bit 44.1KHz as mono wav files.

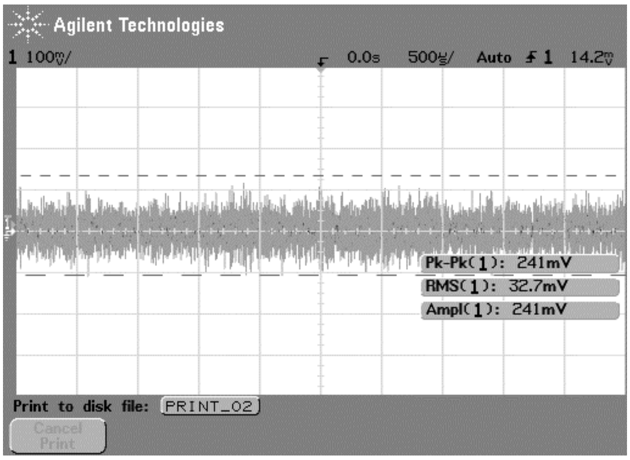


Figure 4.22 – Measured noise from the 003.

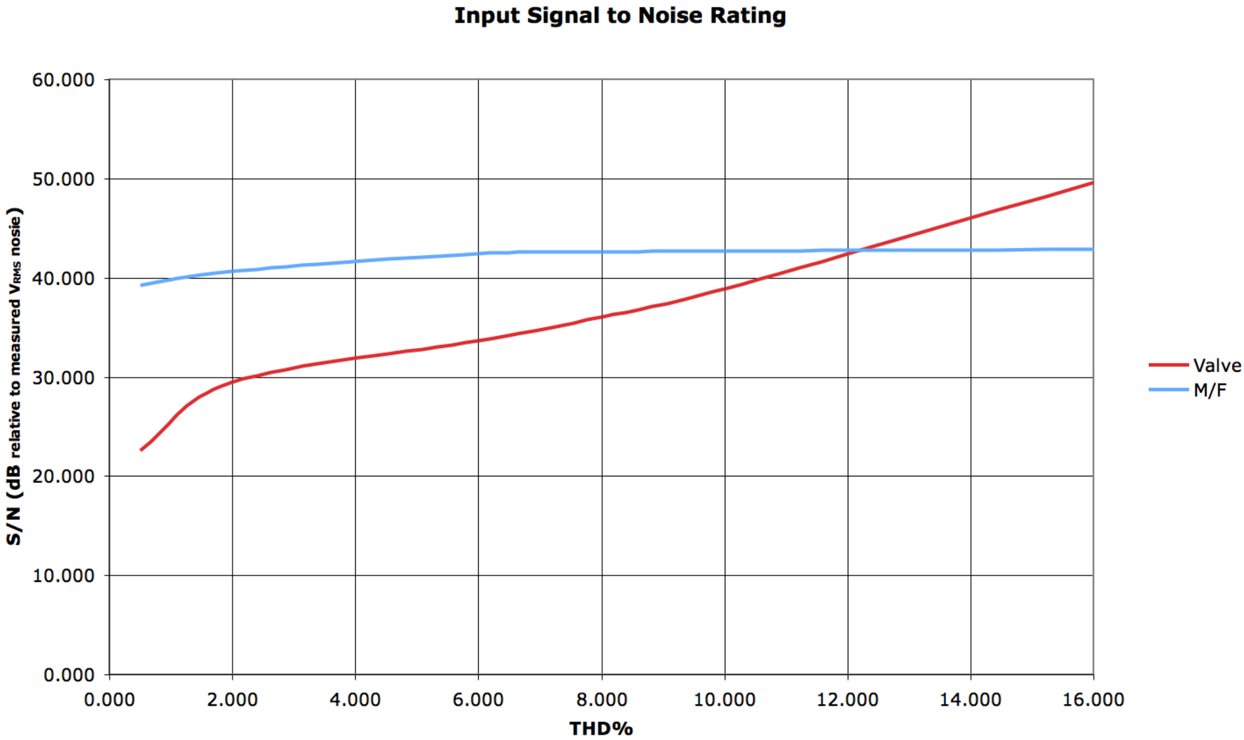


Figure 4.23 – Measured S/N power ratios for input signals.

Figure 4.24 – Test set up for recording of samples.

The resulting recorded samples of varying amplitude were monitored for equal loudness using the Orban level meter, adhering to ITU-R BS.1770 [64] and the levels changed accordingly. This prevented any preference of sample as a result of listening level. The higher percentage THD recordings were attenuated to the lower loudness levels of the lower THD% recordings to avoid unnecessary amplification of the noise.

These levelled recordings were then reproduced at a reference listening level of 75dBA* [65], calculated using a binaural head, figure 4.25, through Sennheiser HD650 headphones with a known SPL of 103dB at 1kHz 1VRMS and nominal impedance of 300Ω.

Figure 4.25 – Calculation of 75dBA listening level.

The recorded and loudness equalled samples were embedded into a modified MUSHRAM.m Matlab file producing nine experiments testing in bands of 0-4%THD, 4-10%THD, and 10-16%THD, figure 4.26. The hidden reference for each band was the clean input signal used for the recordings; the anchor for each experiment was the reference sample low-pass filtered at 3.5kHz with a roll off of 25dB per 500Hz, both samples equal loudness levelled.

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*dBA calculation using the A-weighted filter responses determined by Fletcher-Munson curves of human ear sensitivity. This filter is needed "to obtain objective measurements which correlate well with human response." [66]

	Files								
0 – 4% THD	Experiment 1	M 0.5%	V 0.5%	M 1%	V 1%	M 2%	V 2%	REF	ANCH
	Experiment 2	M 0.5%	V 0.5%	M 1%	V 1%	M 4%	V 4%	REF	ANCH
	Experiment 3	M 2%	V 2%	M 1%	V 1%	M 4%	V 4%	REF	ANCH
4 – 10% THD	Experiment 4	M 4%	V 4%	M 6%	V 6%	M 8%	V 8%	REF	ANCH
	Experiment 5	M 4%	V 4%	M 6%	V 6%	M 10%	V 10%	REF	ANCH
	Experiment 6	M 8%	V 8%	M 6%	V 6%	M 10%	V 10%	REF	ANCH
10 – 16% THD	Experiment 7	M 10%	V 10%	M 12%	V 12%	M 14%	V 14%	REF	ANCH
	Experiment 8	M 10%	V 10%	M 12%	V 12%	M 16%	V 16%	REF	ANCH
	Experiment 9	M 14%	V 14%	M 12%	V 12%	M 16%	V 16%	REF	ANCH

Figure 4.26 – Audio evaluation sample organisation.

A previous study with loudspeakers addressed the issue of whether or not “the preferences of trained listeners can be extrapolated to a larger population of naïve, untrained listeners.” [67] This study concluded, “Trained listeners have the same preferences as untrained listeners.” For this reason it was suggested that although it would be advantageous for the all of the subjects used to be experienced trained listeners, the possible impracticalities of this meant some untrained but still experienced listeners could also be used.

The objective of this auditory evaluation was to provide evidence as to whether the Monteith/Flowers transistor preamplifier sounds “better” or more preferable than a valve based stage. It seems reasonable to suggest that using untrained experienced listeners would not have an adverse effect upon the resulting data. Testing was not carried out on specific quality attributes because it was felt that this introduced too much room for subjective misinterpretation. A preference test was deemed appropriate because of the findings presented in Olive’s research. [67]

In accordance with ITU-R BS. 1534-1 a training stage was utilised before the grading stage to allow all listeners to become accustomed and comfortable with the GUI, (general user interface) programmed using Matlab script, figure 4.27. The instructions given to each test subject prior to the evaluation can be found in Appendix 8.10.

Figure 4.27 – Grading stage GUI programmed in Matlab.

All testing took place in a controlled environment to minimise any distractions. While the MUSHRA test was running, all on-screen objects that did "not correspond to the currently auditioned stimulus" [61] were disabled.

Figure 4.28 – The controlled environment courtesy of the University of Huddersfield.

4.5 - Audio Evaluation Analysis:

Of the eleven subjects to partake in the audio evaluation, two sets of results were declared unusable. One subject demonstrated contraction bias, "a conservative tendency in using the grading scale," [61], scoring all samples on a reduced scale of -1 to 1. The other suggested in his comments about the test; "I love distortion, it all sounds good to me," subsequently scoring all results on a shifted and reduced scale of +18 to +70.

The remaining nine subjects results were arranged into useable data format in Excel, and an initial plot of individual sample score was produced, figure 4.29. Both samples within both test sets demonstrated normal distribution and equal variance. This allowed the data to be put through a two-way ANOVA test*, figure 4.30, the results of which allowed four conclusion to be drawn:

- Interaction between sample and THD% was *not* significant.
- THD% was *not* a significant factor for the transistor stage.
- THD% *was* a significant factor for the valve stage.
- Sample selection was *not* significant for either transistor or valve stages.

Because the choice of sample had no significant effect upon the overall result, the scores from sample 1 and sample 2 could be combined, figure 4.31.

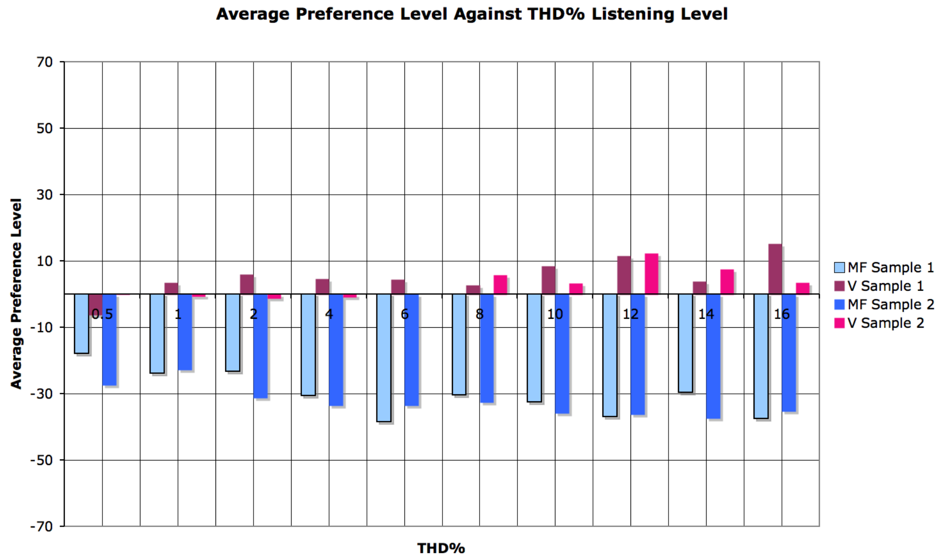

Figure 4.29 – Initial plot of individual sample scores.

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* A two-way ANOVA test was preferred to a paired t-test purely because the data was formatted in such a way as to allow Excel to be used for the statistical analysis. "The results of a paired t-test are mathematically identical to those of a two-way anova, but the paired t-test is easier to do." [69]

ANOVA - Transistor						
Source of Variation	*SS*	*df*	*MS*	*F*	*P-value*	*F crit*
THD%	4391.67222	9	487.96358	1.27823276	0.25250584	1.93881904
Sample 1 or 2	301.605556	1	301.605556	0.79006327	0.37541536	3.90023603
Interaction	923.338889	9	102.59321	0.26874547	0.98212697	1.93881904
Within	61079.7778	160	381.748611			
Total	66696.3944	179				

ANOVA - Valve						
Source of Variation	*SS*	*df*	*MS*	*F*	*P-value*	*F crit*
THD%	2949.97917	9	327.775463	2.48495809	0.01105092	1.93881904
Sample 1 or 2	258.001389	1	258.001389	1.95598118	0.16387909	3.90023603
Interaction	1292.29028	9	143.587809	1.08857961	0.37406479	1.93881904
Within	21104.6111	160	131.903819			
Total	25604.8819	179				

Figure 4.30 – Two-way ANOVA test for Valve and M/F stage.

THD%	0.50%	1.00%	2.00%	4.00%	6.00%	8.00%	10.00%	12.00%	14.00%	16.00%
Transistor	-51	-55	-62.5	-61	-68	-65.5	-65.75	-63.5	-64	-70
	-24.5	-26.5	-39.5	-29.5	-36	-34	-37.5	-45	-34.5	-41.5
	-22.5	-13	-26.5	-33.25	-21	-7	-28.25	-16	-28	-16
	-22	-25.5	-23	-24	-25.5	-17.5	-25	-18	-17	-33.5
	-16.5	-17.5	-18	-34.5	-26.5	-33.5	-31.75	-34	-30.5	-27
	-5	-30	-5.5	-36	-30	-23	-33.75	-40	-12	-33.5
	-22.5	-34	-35	-40.5	-54	-35	-41.25	-44	-54	-49
	-40	-35.5	-36	-33.25	-36.5	-37.5	-28.75	-36.5	-40.5	-33
	1.5	28.5	1.5	4.5	-25	-29.5	-14.5	-31	-19.5	-22.5
Valve	-21.5	-19	-5.5	-8.25	-6	0	-2.75	10.5	0	0.5
	14.5	22.5	5.5	6	-2.5	20	16.5	36	4.5	29
	-3.5	6	6.5	10.5	0	7	8	20.5	28.5	15.5
	-4	-5.5	-5	-3	0	4	-1	4	0	2
	-6	9	2.5	3.25	5.5	-1	0.5	4	0	-3.5
	10.5	10	10.5	2.75	12.5	8	7.25	5	16	7
	0	0	0	0	13	0	11	9.5	0	25.5
	6	2.5	2	8	1.5	0	4	-4	2	3
	-23.5	-13	4.5	-1.5	-4.5	0	9.5	22	0	4.5

Figure 4.31 – Combined results of audio evaluation.

The combined averages were plotted and 95% confidence levels calculated for each result. Because the sample size was less than thirty, z-values could not be used, instead confidence levels were calculated using t-values and a degrees of freedom chart. The results of this combined plot, figure 4.32, showed conclusively that all through the region of overload, defined in this case as 0.5 – 16%THD, it could be stated with 95% confidence that the valves distortion characteristics were preferred to those of the Monteith and Flowers pre-amplifier.

One interesting observation is the comparison between the signal to noise plot, figure 4.23, and the preference plot for the valve stage. The correlation between the two lines could well suggest that the slight trend in valve preference of higher THD%s in comparison to low THD%s could be as a result of the noise on the sample recordings.

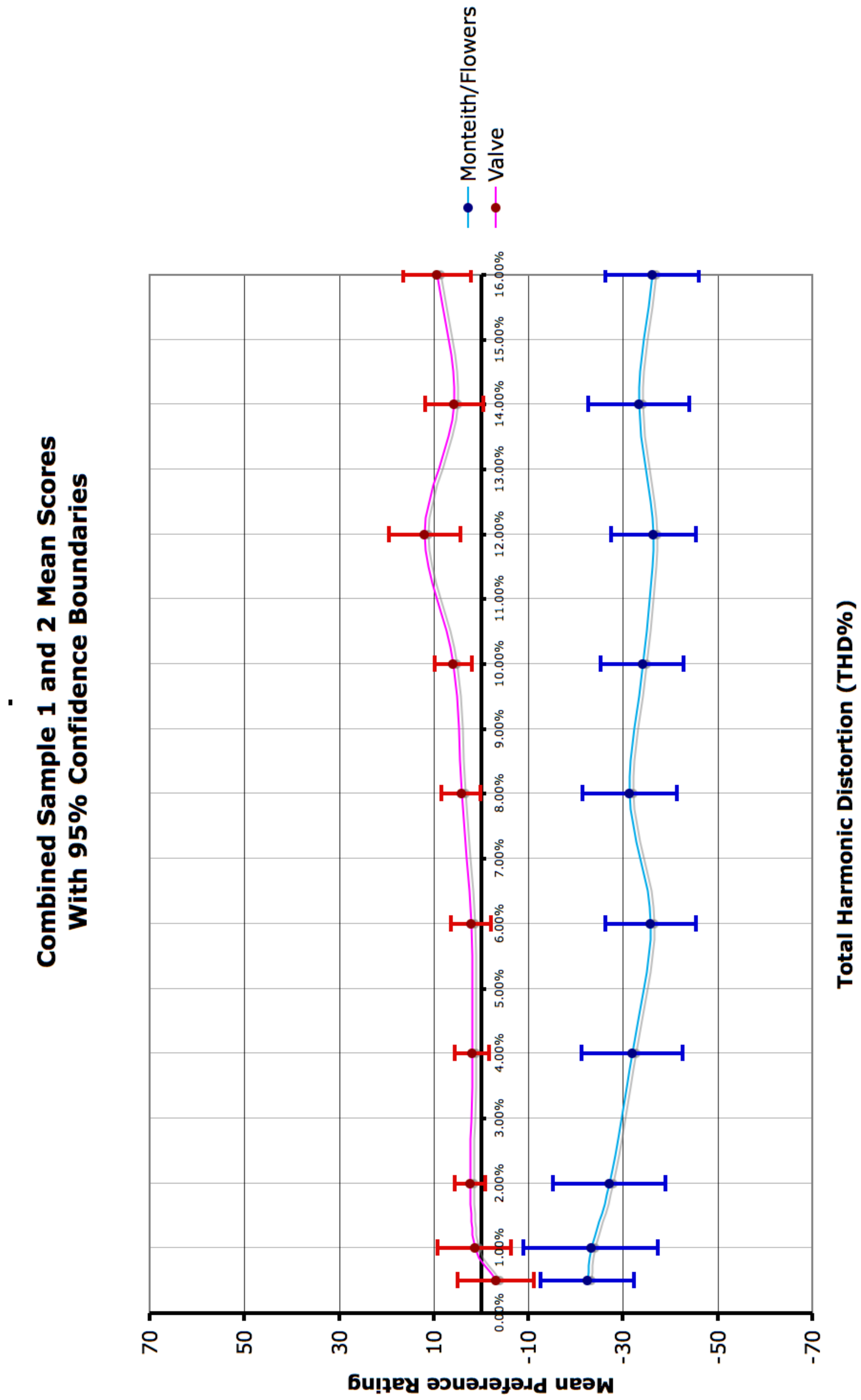

Figure 4.32 – Resultant plot of combined samples preference rating with 95% confidence levels.

Further to the two-way ANOVA, analysing the variance within the M/F and Valve results separately, an n-way ANOVA was completed on the entire group of results, figure 4.33, allowing further conclusions to be drawn.

ANOVA

Source of Variation	*SS*	*df*	*MS*	*F*	*P-value*	*F crit*
Transistor Vs Valve	56347.9587	1	56347.9587	296.739024	2.8026E-38	3.90023603
THD% significance	429.350347	9	47.7055941	0.2512267	0.98592052	1.93881904
Interaction	3241.47535	9	360.163927	1.89669146	0.05590139	1.93881904
Within	30382.5	160	189.890625			
Total	90401.2844	179				

Figure 4.33 – N-way ANOVA analysis of combined audio evaluation results.

An F-statistic of 296.54 in conjunction with a very small p value of 2.80262E-38 allowed conclusion that with 99% (1-pvalue) certainty the valve sound in this scenario was preferable to the Monteith and Flowers design.

It could also be said that with 98.6% confidence the total harmonic distortion percentage was not a significant factor when directly comparing the distortion characteristics of the preamps. Consequently it would seem fair to conclude that the use of matched output THD percentages was a reasonable way of producing comparative results between these two systems distortion characteristics.

The overall research and findings of this section have been collated into a white paper, found in Appendix 8.14, the abstract of which has been sent in to the AES in hope of publication in the forthcoming 130th AES convention in London.

Chapter 5 - Anode HT Supply reduction:

The research conducted into the Monteith and Flowers paper particularly demonstrated that the weighting of harmonic components in an output signal have significant effect upon perceived distortion characteristics when operated in the region of overload. The overall objective of this thesis was to study the use of the valve in low powered electronics for guitar amplification. As determined continuously throughout the previous writings, guitar amplification exploits the overload characteristics of the amplifier state.

Considering this, and relating it to the thesis objective, this chapter adopts the same methodology used for the fabrication of the authors research paper "Can Transistors Sound Better than Valves?" to closely study:

- The effect of HT supply reduction and increasing levels of input signal amplitude upon the output harmonic component weighting of a preamplifier.

- The effect of HT supply reduction upon the output harmonic component weighting of a preamplifier in the region of overload, defined as 0.5-16% output THD.

By undertaking this research, the thesis objective of mass data accumulation for the dynamic operation of a valve was also satisfied, with tables of data formulated to describe a valves reaction to input signal amplitudes from 0 – 3.00 V peak-peak, and output harmonic content from 0.5-16%THD, at anode voltages of $50V_{DC}$, $100V_{DC}$, $150V_{DC}$, $200V_{DC}$, $250V_{DC}$, and $300V_{DC}$.

This section will surmise the adopted test methodology and present results and analysis of the findings.

5.1 - Methodology:

As stipulated, the testing for the HT supply reduction research followed the same proven procedure used to obtain results at a HT supply of $200V_{DC}$. The methodology was as followed:

1 – Vary the HT supply to the required DC voltage using a variac across the mains to control the AC voltage to the constructed power supply for the preamplifier, shown in figure 4.2.

2 – Using an Agilent 35670A signal analyser and its internal signal generator, incrementally increase the input signal peak-to-peak amplitude of a 1kHz sine wave from 10mVpp to 3Vpp, in quantization levels of 10mV.

3 – At each 10mV increment record the output signal frequency spectrum .DAT file, THD% level, and peak to peak amplitude waveform, using the Agilent 35670A.

4 – Determine what amplitude of input signal is required to produce 0.5%, 2%, 6%, 8%, 10%, and 16% THD of the output signal.

5 – Apply the required input sine wave amplitude and record the output signal frequency spectrum .DAT file, and peak-to-peak amplitude waveform, using the Agilent 35670A, for each output THD%.

6 – Adjust the RMS amplitude of the two dry guitar samples, using SoundForge and Matlab, to achieve sample levels in dBFS* of equivalent amplitude to the measured peak-to-peak sine waves required to produce the set level of output THD%. (Appendix 8.12 includes conversion tables and screen shots of conversions.)

7 – Use Digidesign 003 rack to 're-amp' amplitude adjusted samples through the preamplifier, recording the output at 16 bit 44.1KHz as mono wav files.

8 – Using the Orban level meter [64] set the volume of each sample to equal loudness.

9 – Play the samples back using MUSHRA test configuration in Matlab at an SPL of 75dBA.

10 – Analyse the objective and subjective results.

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*0dBFS of a 16bit system relates to a sample level of 32768.

5.2 - Analysis of HT Supply Reduction:

With the lengthy process of data collection completed, all the .DAT files were converted to .csv files and imported into Excel. This data was formatted into large tables of results, which detail the dynamic operation of an ECC83 dual triode preamplifier valve. The resulting spreadsheets can be found on the CD entitled "Dynamic Operation of an ECC83 Data", attached to the back page of this Thesis. It is hoped that this large set of data can be used for further work.

As well as collecting data for the valve stage, data was also collected for the M/F preamplifier at reducing supply voltages of 50, 100, 150, 200, 250, and $300V_{DC}$, using the applied methodology described in 5.1. Although the subjective analysis of this preamplifier concluded that it does not sound better than a valve preamplifier, it did objectively show similarities in low order harmonic content. As such it seemed pertinent to study the effect of reducing its supply voltage upon the output harmonic content.

Studying the effects of HT reduction on both preamplifier states allowed comparisons to be made between valve and transistor state technology as well as comparisons between one states reduction in HT. it also allowed conclusions to be made as to what, if any, the use of the thermionic valve has for low powered guitar amplification, and if transistor based topologies could be used for low power modelling of the valve.

5.2.1- Objective Analysis of Transfer Curve and THD%:

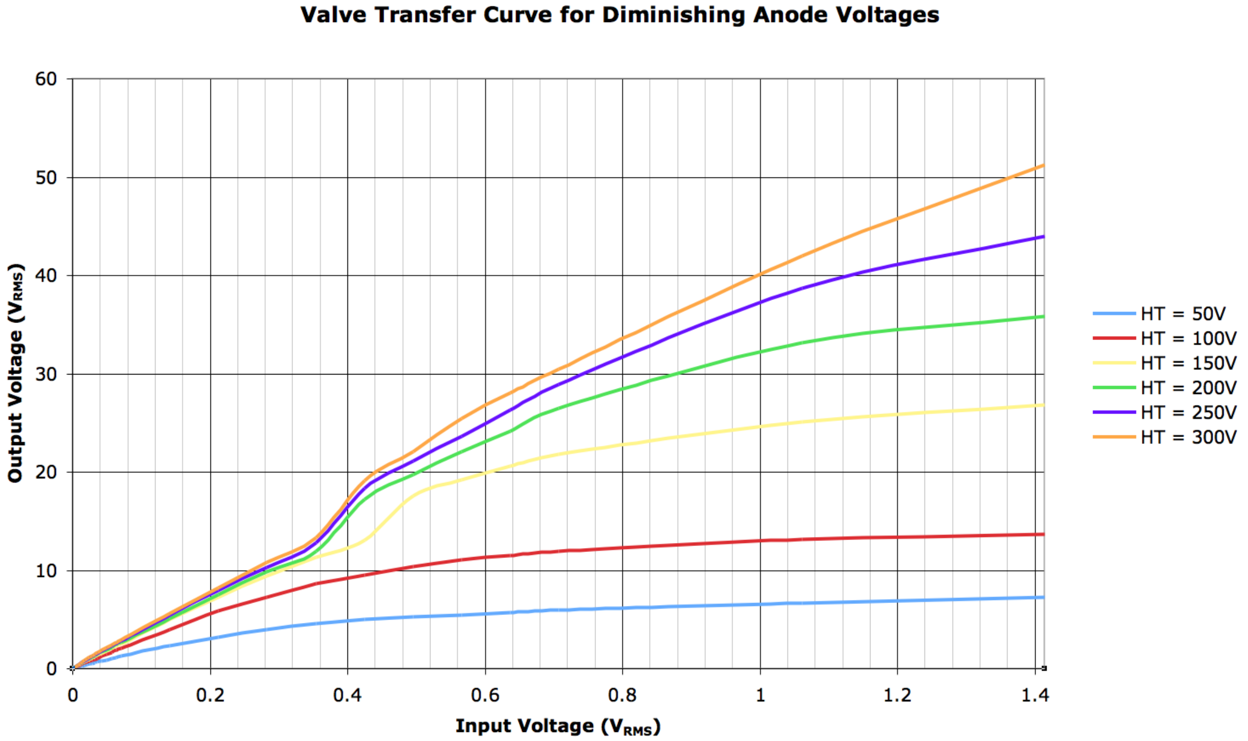

Figure 5.1 – Effect of anode voltage reduction on valve transfer curve.

The ECC83 dual triode has a maximum datasheet value for anode DC voltage of 300V, likewise the MPS-U10 high voltage transistor amplifier has a maximum V_{CEO} (collector-emitter voltage) of $300V_{DC}$. Testing consequently began at an operating voltage of HT = 300V, then gradually decreased in 50V increments. Figures 5.1 and 5.3 display the resulting transfer curves of input versus output V_{RMS} for the two systems.

It can be observed that the valve exhibits a decreasing "soft limit" effect (roll off), decreasing amplification factor, and a relatively uniform increase in the ratio of amplification reduction as a result of halving the anode voltage:

Va = anode voltage		µ = amplification factor	
Va_{DC}	**µ**	**Roll off**	**Reduction ratio**
300	41.87	33	
200	36.13	19.9	1.158870745
100	15.25	4	2.557377049
50	8.25	2.8	3.767676768

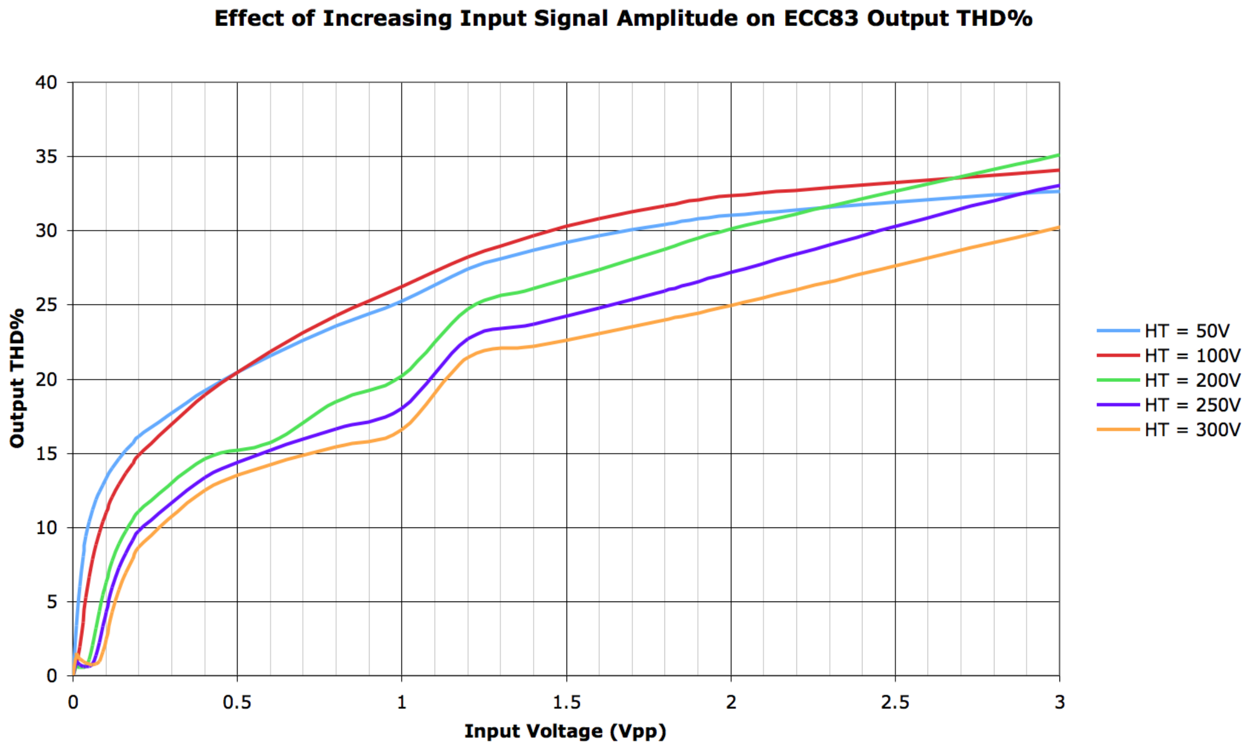

Figure 5.2 – Effect of valve input signal generation of THD% at varying anode voltages.

Interestingly the effect of anode reduction has the reciprocal effect upon the output signals THD%. So a reduction in anode voltage results in an increase in output signal distortion. This directly agrees with the hypothesis proposed on page 36 at the end of section 2.1.6 as a result of anode characteristic theory.

For the M/F preamp, figure 5.4, there is a greater ratio of increase in output THD% as supply voltage is reduced as a direct effect of the shift in the onset of "hard limit" shown in figure 5.3. It can be seen that as the supply voltage is reduced the transfer curve is uniformly reduced, and the "knee" of the curve approaches lesser-input voltages.

Interestingly for both systems the reduction from 300V to 150V had little impact upon the shape of the transfer curve, but at 100V and 50V the change in shape is obvious. This would suggest that in order to keep the shape of the transfer curve the input signal amplitude must be monitored and adjusted. This is precisely what was done in achieving matched THD%s at output, the input signal amplitudes were carefully adjusted. The resulting plots of THD% against Output Vrms can be seen in figures 5.5 and 5.6.

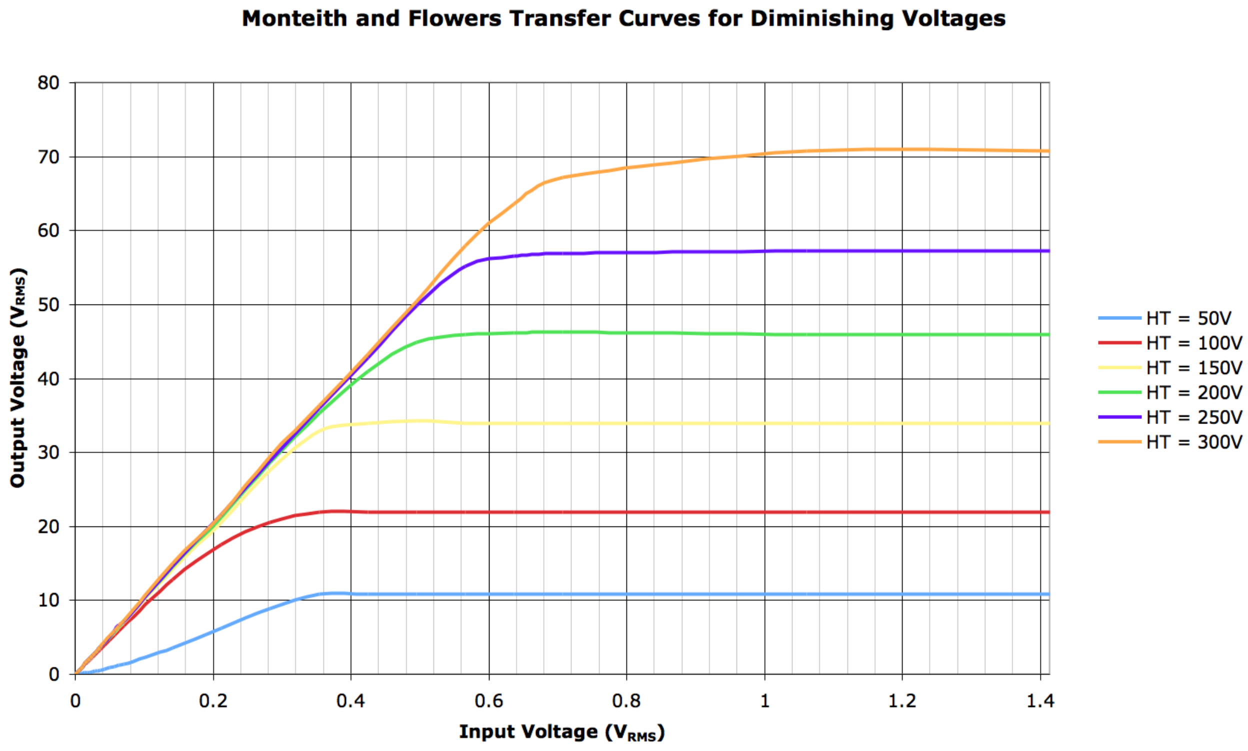

Figure 5.3 – Effect of supply voltage reduction on M/F transfer curve.

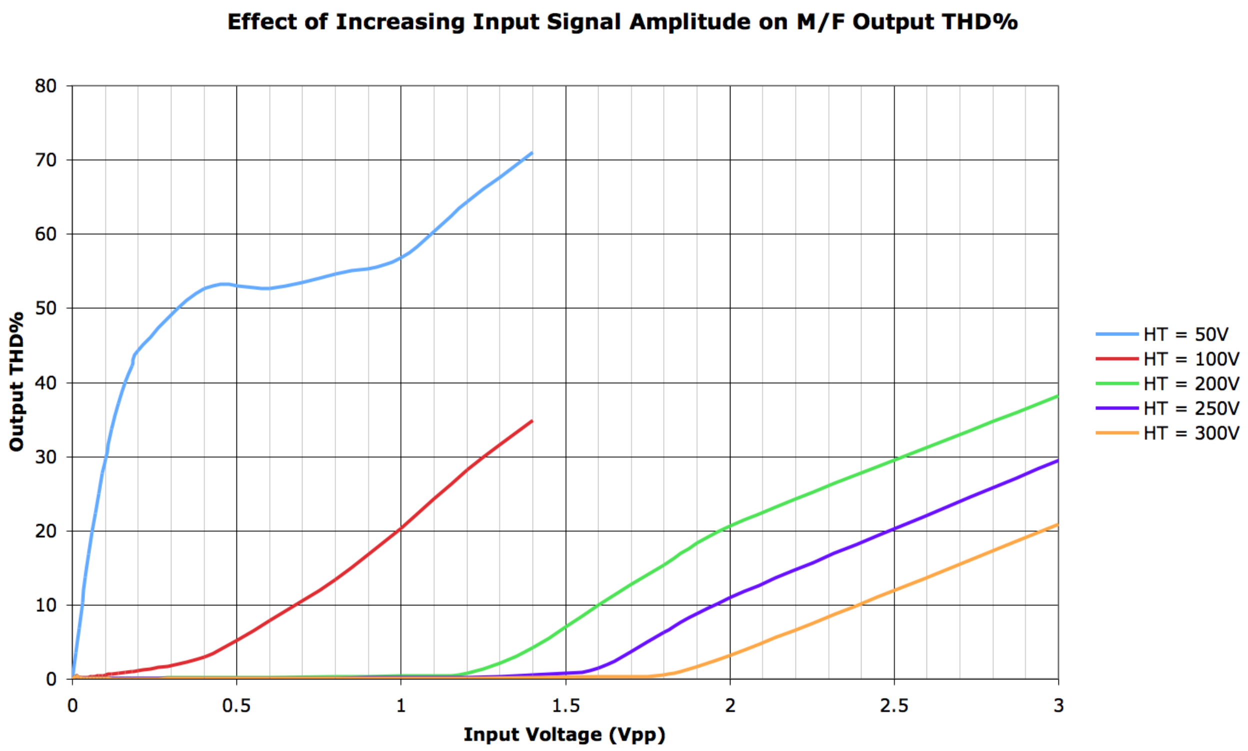

Figure 5.4 - Effect of M/F input signal generation of THD% at varying supply voltages.

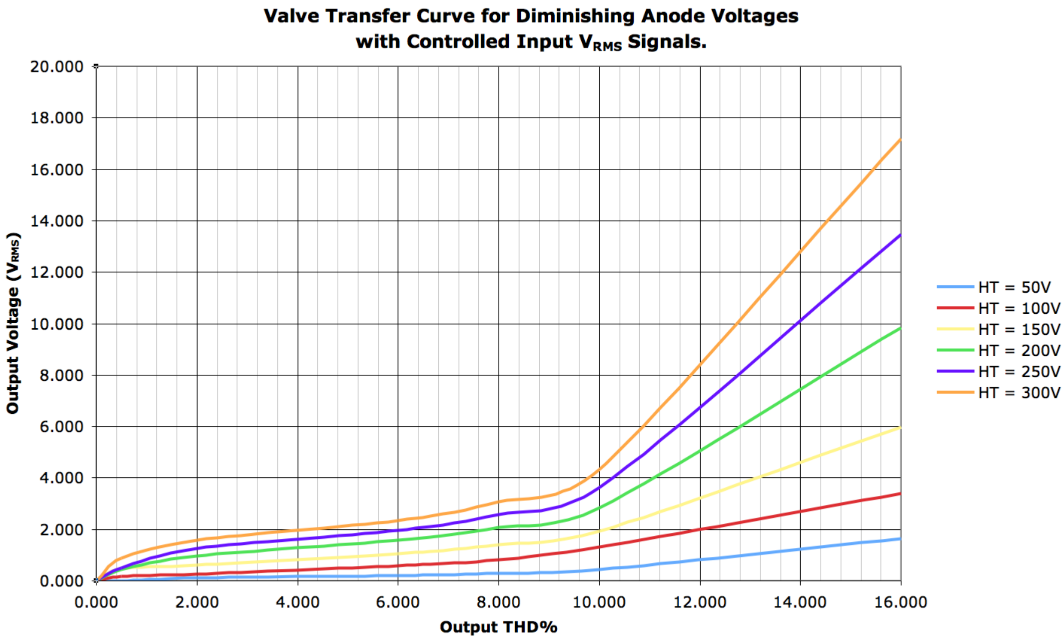

Figure 5.5 – Matching of the transfer curve shape for an ECC83.

By controlling the input signal amplitude to the preamplifier the shape of the transfer curve was maintained as the anode voltage was reduced. The similarity in curve shape suggests that it can be possible to use the valve at lower anode voltages without much deviation from the full power 'valve sound' so long as the input levels are controlled. The same can be said for the M/F stage, figure 5.6. Note how the "knee" of the transfer curve is now at the same place for each supply voltage, this suggests that the resulting sound of the amplifier will maintain its characteristics at low power usage.

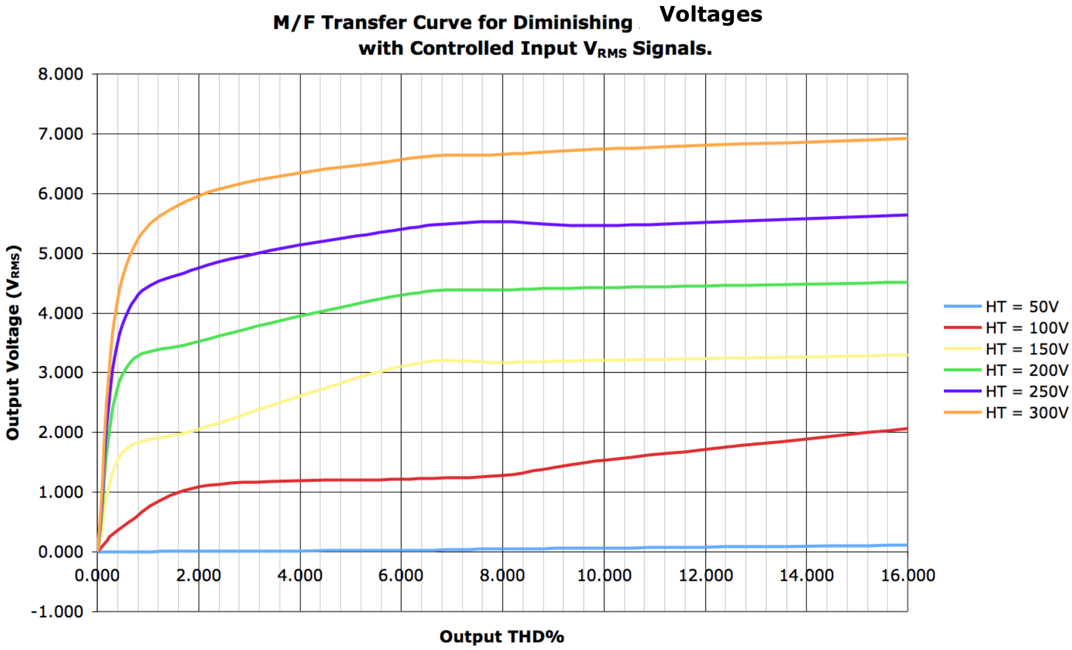

Figure 5.6 - Matching of the transfer curve shape for the M/F.

The actual power reduction achieved by reducing the anode voltage was calculated using measured anode current multiplied by anode voltage. By determining the voltage dropped across the 24.3kΩ resistance in the PSU, figure 5.7, taking DC voltage readings at HT1 and HT4 (HT4 being the anode voltage for the preamplifier stage) and using Ohms law, the anode current could be found.

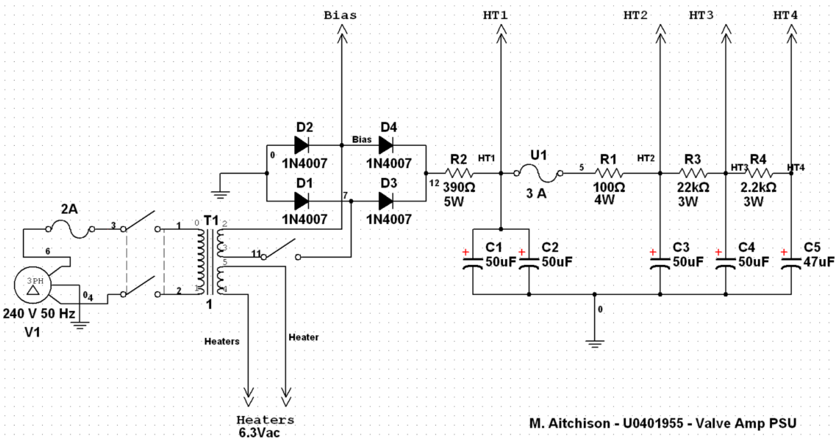

Figure 5.7 – Preamplifier PSU with 24.3kΩ between HT1 and HT4.

The plot of anode current against input Vrms, figure 5.8, shows the current draw of the amplifier as input signal amplitude is increased. The average current draw was determined for each anode voltage, and anode power was calculated by multiplying the two (P=VI), the resulting table is shown in figure 5.9.

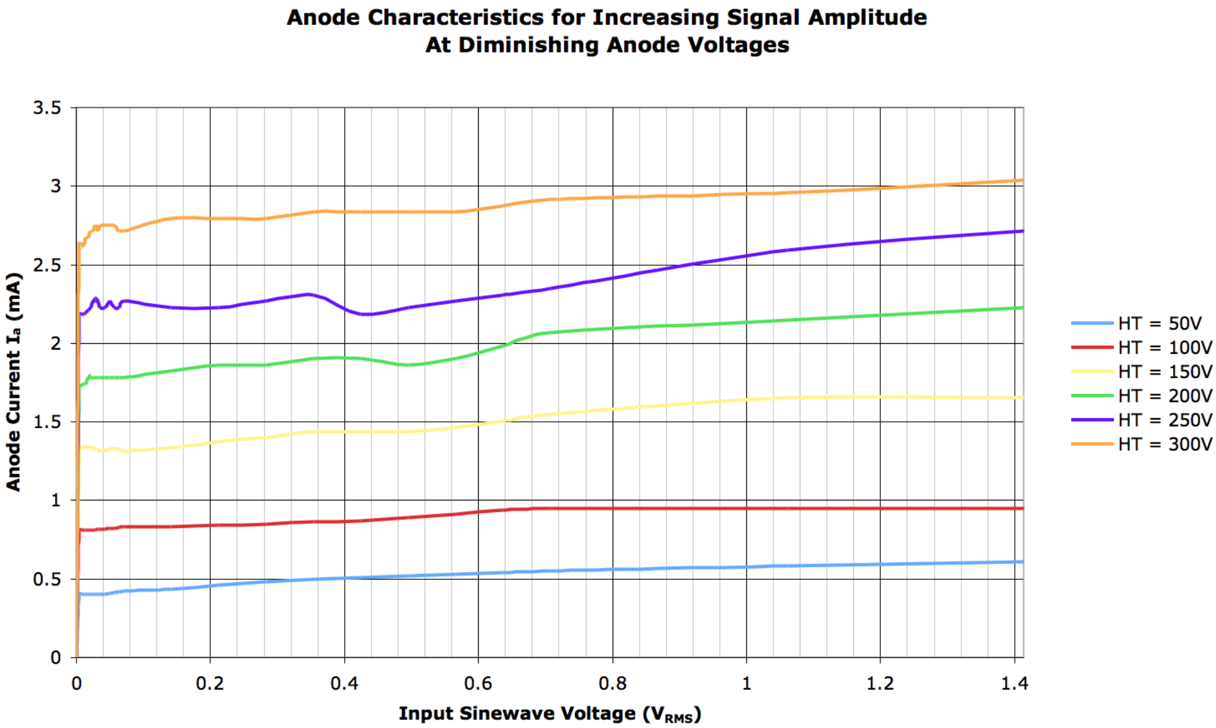

Figure 5.8 – Anode current at varying HT voltages.

	Anode Voltage V_{DC}					
	50	100	150	200	250	300
Aerage Ia (mA)	0.45533851	0.84914478	1.39218619	1.84908247	2.2740443	2.77284481
Average Pa (mW)	22.7669253	84.914478	208.827928	369.816494	568.511076	831.853444
Reduction Ratio from HT = 300V	36.5378036	9.79636764	3.98343963	2.24936816	1.46321414	1
Reduction Ratio from HT = 200V	16.2435854	4.35516419	1.77091492	1		

Figure 5.9 – Power reduction ratios for valve anode voltage drop.

Reducing the anode voltage from 200V to 50V results in a power reduction factor of 16.24. In an amplifier running at maximum HT levels a power reduction factor of 36.54 can be achieved by reducing the anode voltage from 300V to 50V. However the affect of this anode voltage reduction upon output harmonic content weighting still needs to be analysed.

5.2.2- Objective Analysis of Output Harmonic Content:

Analysis of the transfer curves suggests that in order to power scale the 'valve sound' in a non-detrimental way, the input signal amplitudes must be closely monitored. This is in agreement with Bruce Clement's findings [40], as described in section 3.1. This theory is confirmed by analysis of the harmonic content of both the valve and M/F preamplifier as a result of increasing input signal level, figure 5.10.

As the supply voltage is reduced both sets of plots demonstrate interchanging dominance of harmonics of third order and above. This indicates that the output sound is changing, an unsatisfactory effect for reducing the power level of the preamp whilst maintaining the full power 'valve sound'.

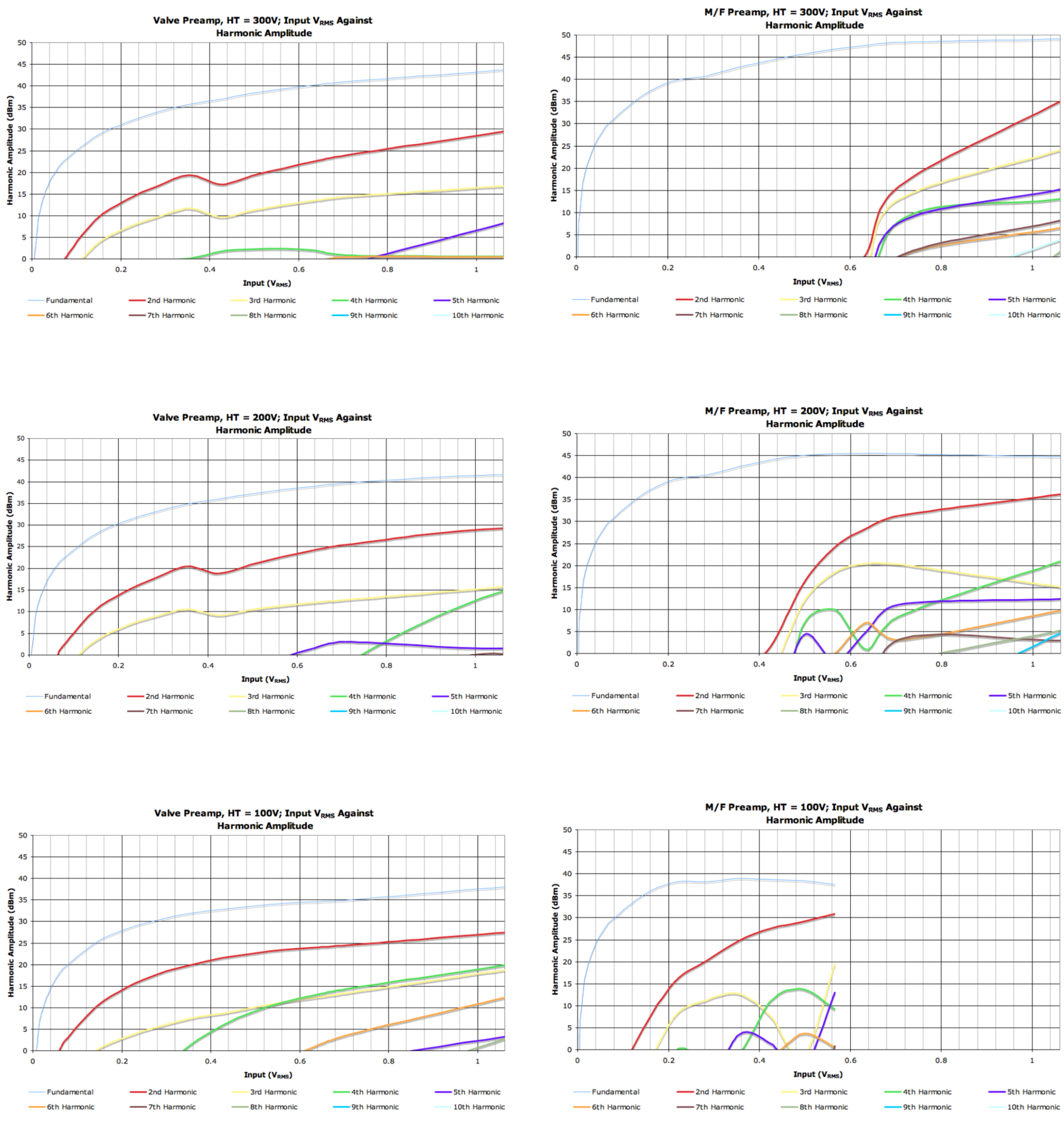

Figure 5.10 – Harmonic content analysis of dynamic reaction to increasing input level.

Figures 5.11 – 5.16 show the harmonic weighting of the output signals when the input signal levels are controlled to give required THD%s.

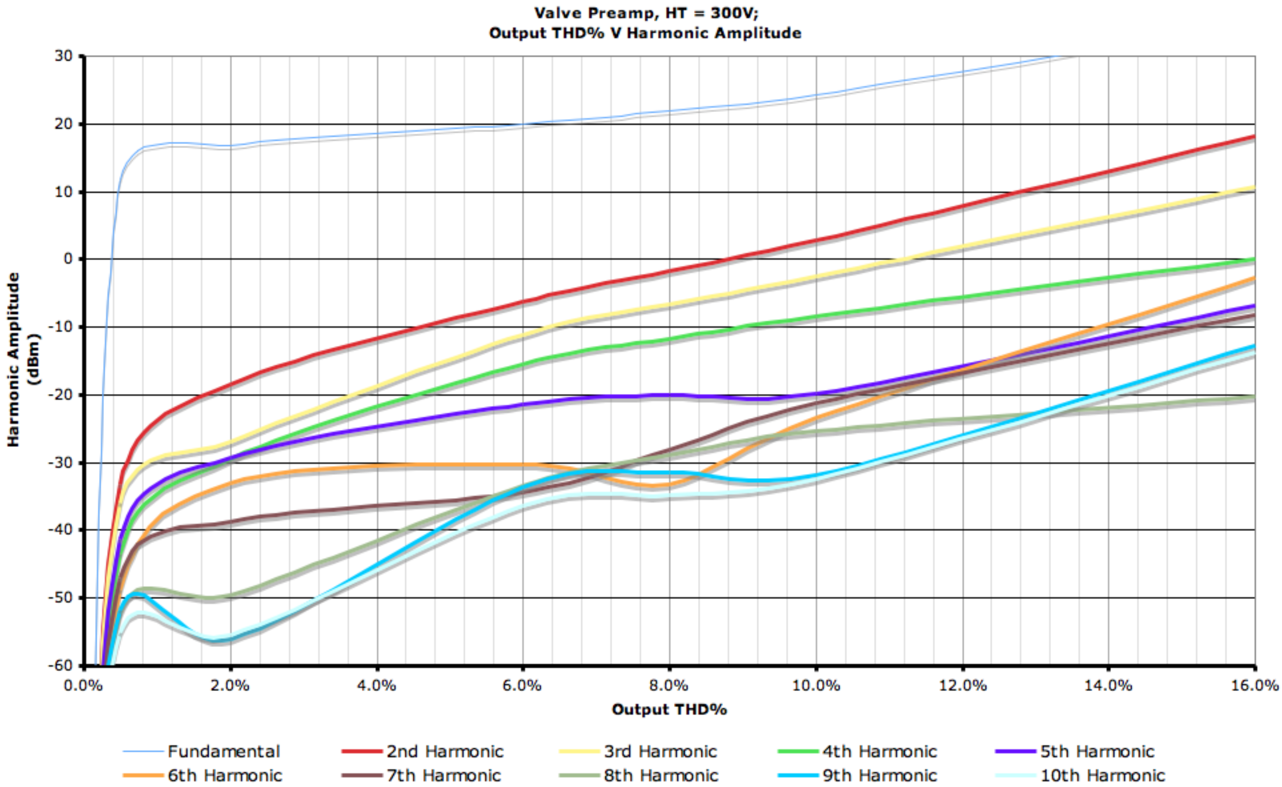

Figure 5.11

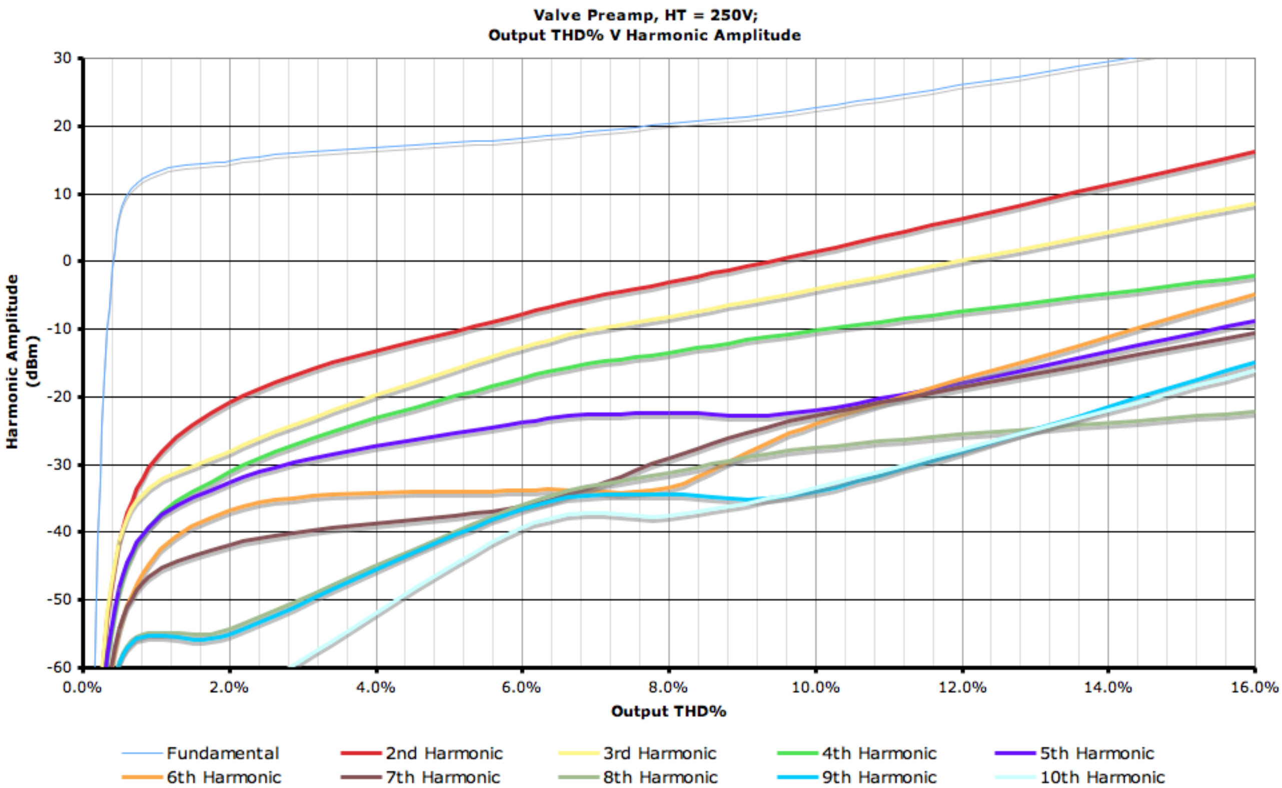

Figure 5.12

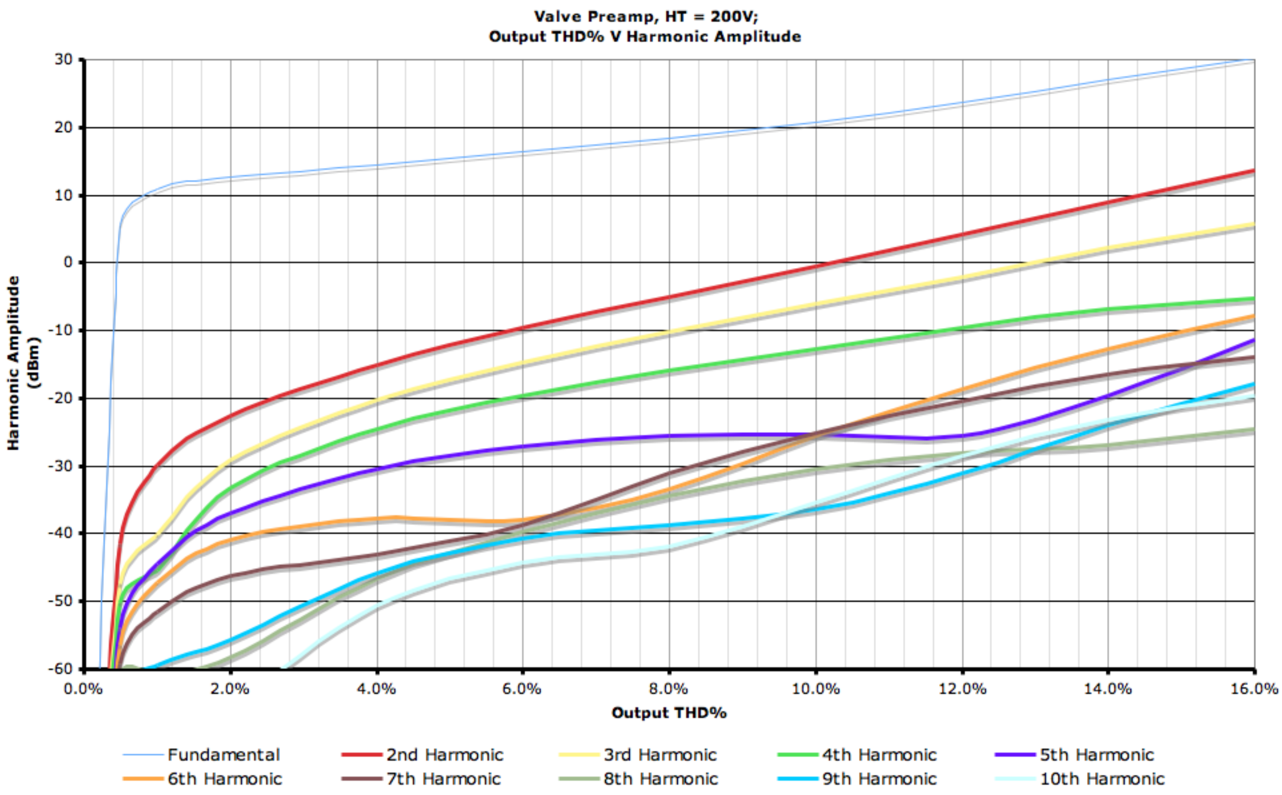

Figure 5.13

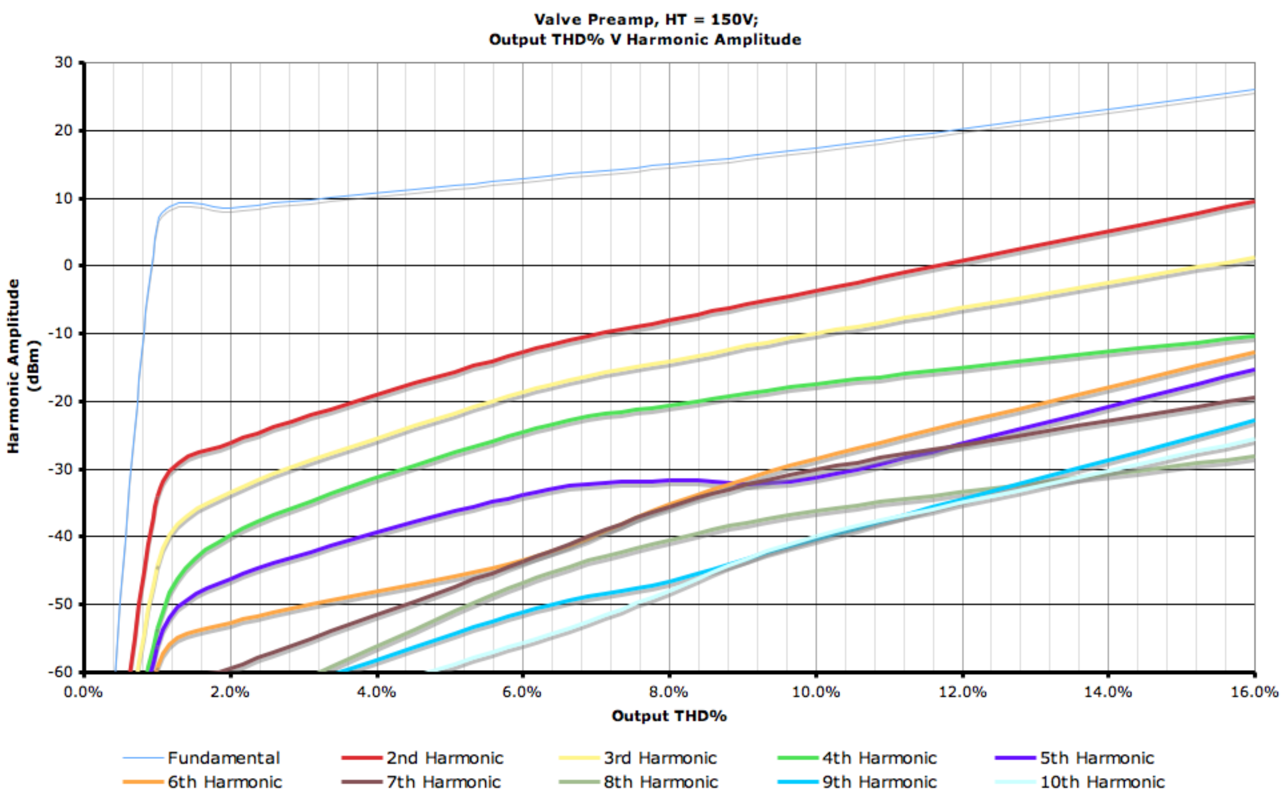

Figure 5.14

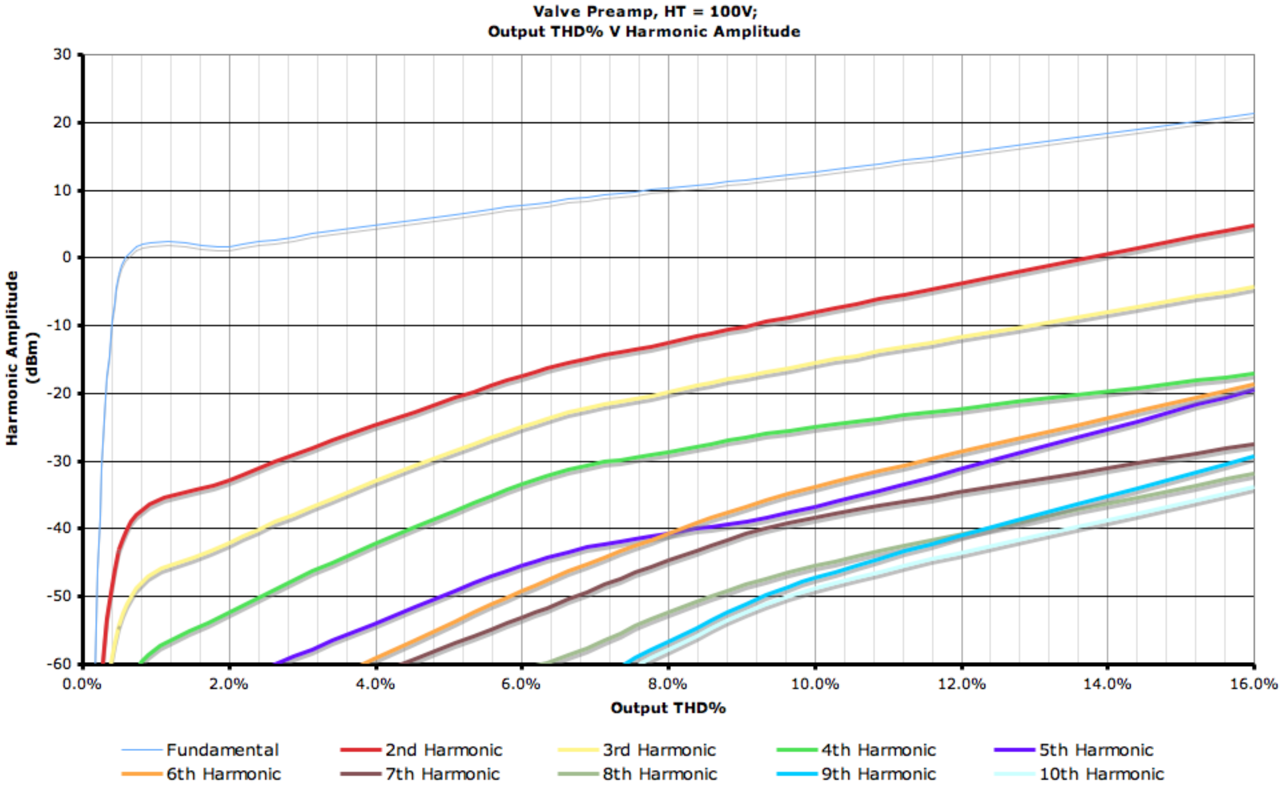

Figure 5.15

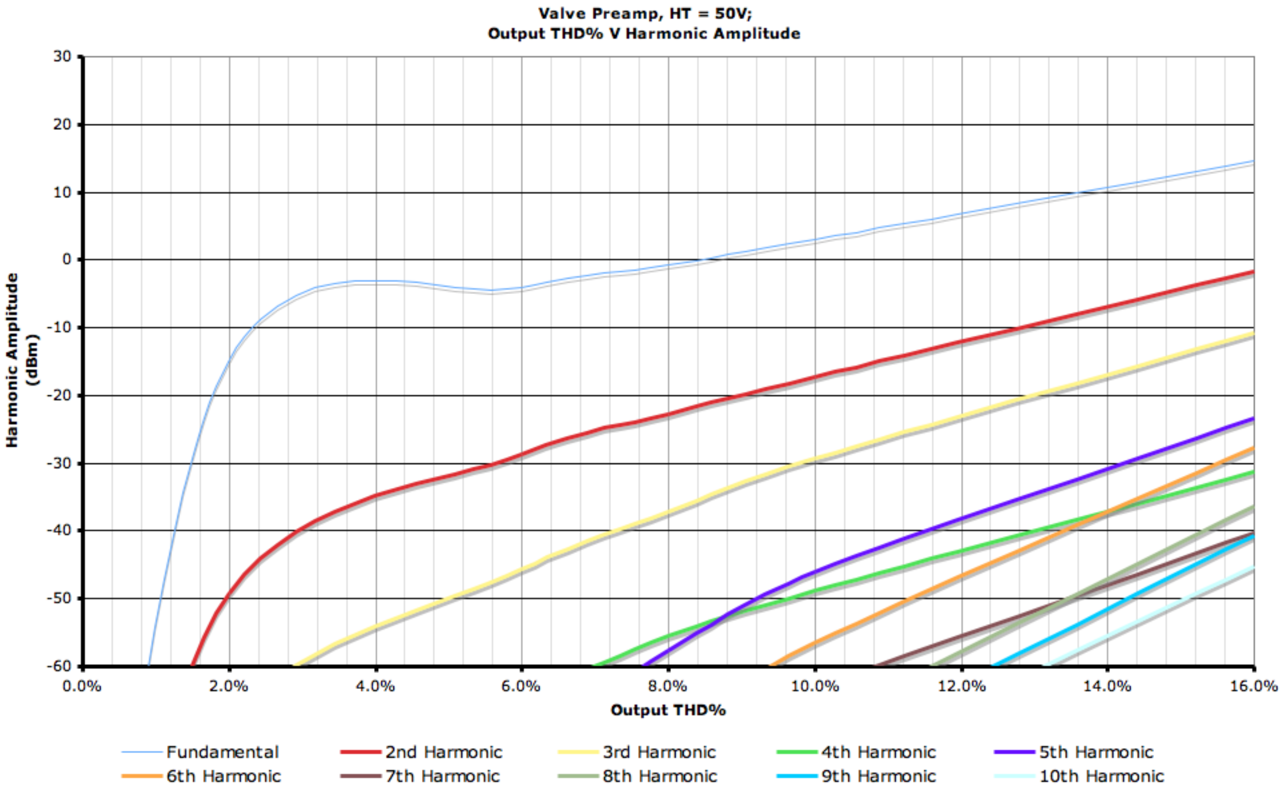

Figure 5.16

The relative relationship of 2nd to 6th order harmonic amplitudes remains almost constant as the anode voltage is reduced. Obviously the overall amplitude of the harmonics is reduced due to the aforementioned amplification factor reduction, but even the amplitudes of the 8th, 9th, and 10th harmonics have an unchanging order of dominance from 300V-100V.

There is a noticeable difference in harmonic weighting for the 50V anode voltage. Looking specifically at the range from 14-16%THD illuminates the fact that as the anode voltage is reduced, the even order harmonics are increased in amplitude compared to the odd order harmonics. The output waveforms, figure 5.17, show correlation to Leach's [44] simulated waveforms in figure 3.9.

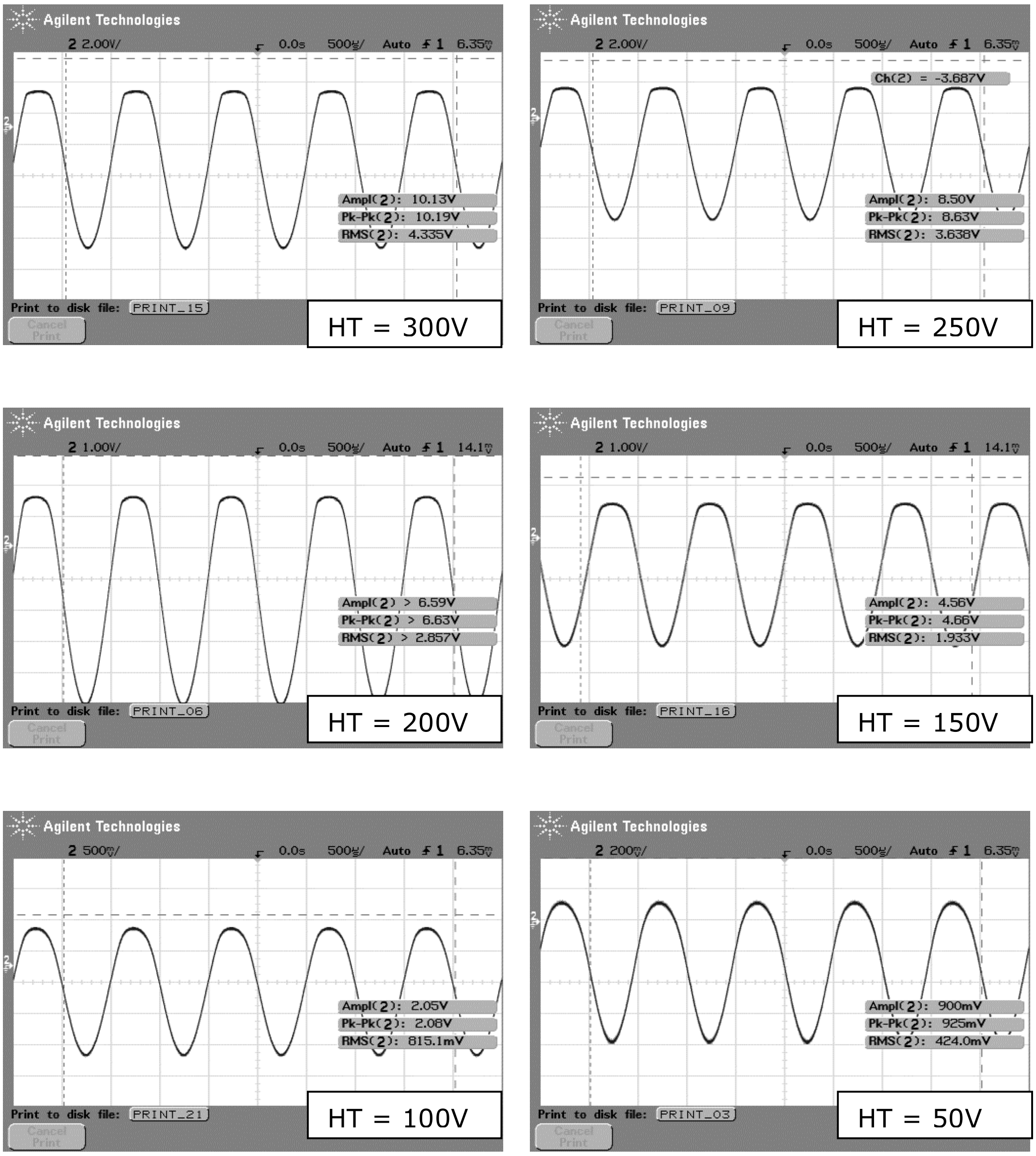

Figure 5.17 – Output waveforms at 10%THD.

As HT supply is reduced, the shape of the positive peaks of the output waveform changes; At 300V_{DC} the positive peak is flatter than at 50V_{DC}, closer resembling a square wave, increasing odd harmonic amplitudes. The explanation for the reduction in the peak flattening relates directly to a key factor in the generation of the 'valve sound', the grid current.

As the HT supply is reduced the current flow through the valve is reduced and simultaneously the grid leak current is reduced. As grid leak current reduces the positive peaks of the input signal become less impeded (explained in section 2.21) and thus less flattened. So as the anode voltage is reduced, it is actually the odd order harmonics that are reduced in amplitude, producing the predominantly more even weighting of harmonic content.

The M/F preamplifier, like the valve, demonstrated a close correlation between the weighting of its low order harmonics, 2^{nd} to 6^{th}, through supply reduction, figure 5.18. It did not show any trend in its weighting of higher order harmonics however, suggesting that this interchange in "edge" harmonics, of which "the ear seems very sensitive," [5] will still produce the perceivable audible effect highlighted in the subjective audio evaluation at 200V supply.

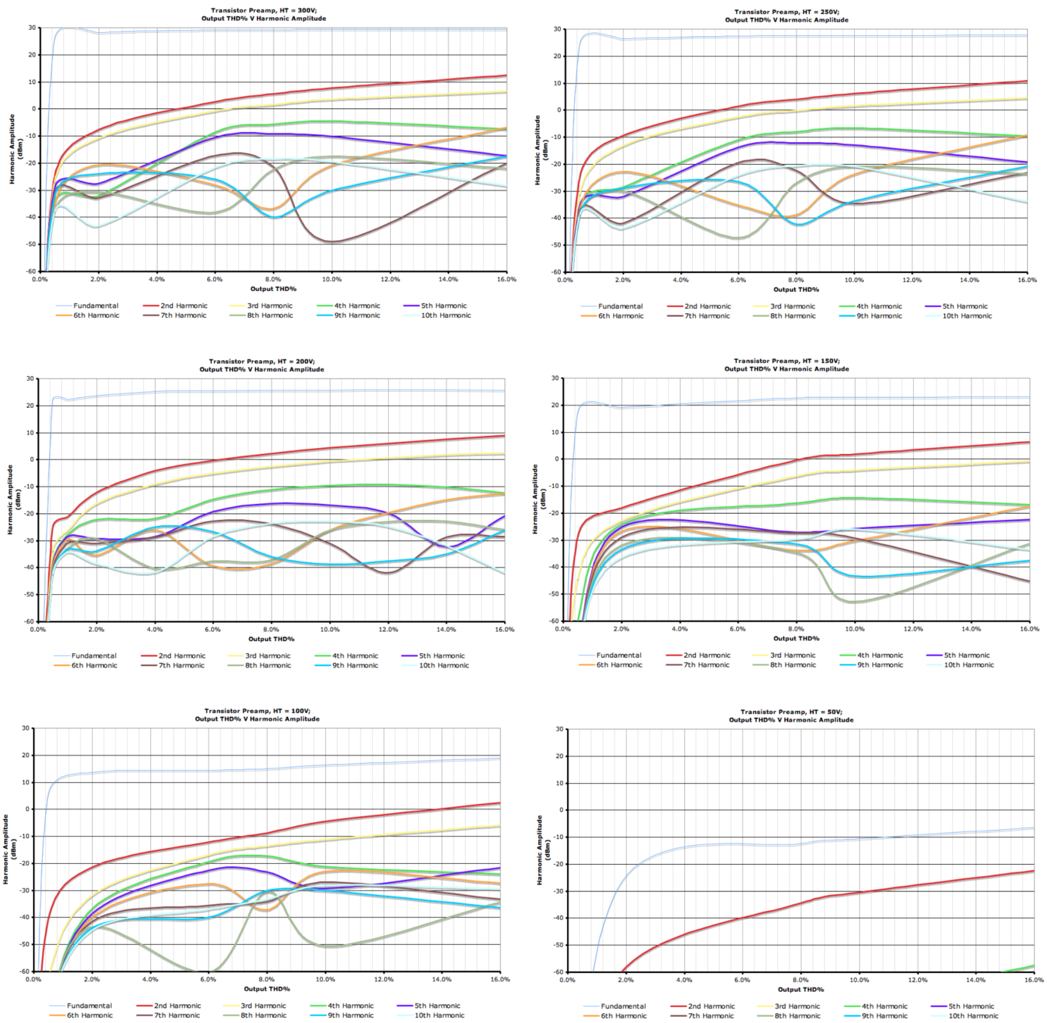

Figure 5.18 – M/F harmonic amplitude against THD% at reducing supply voltage.

In controlling the amplitude of the input signal, resulting in good correlation of the shape of the transfer curves, the valves "primary colour characteristics," [5] determined by the low order harmonics, have very close correlation throughout the anode voltage drop range 300-100V, figure 5.19.

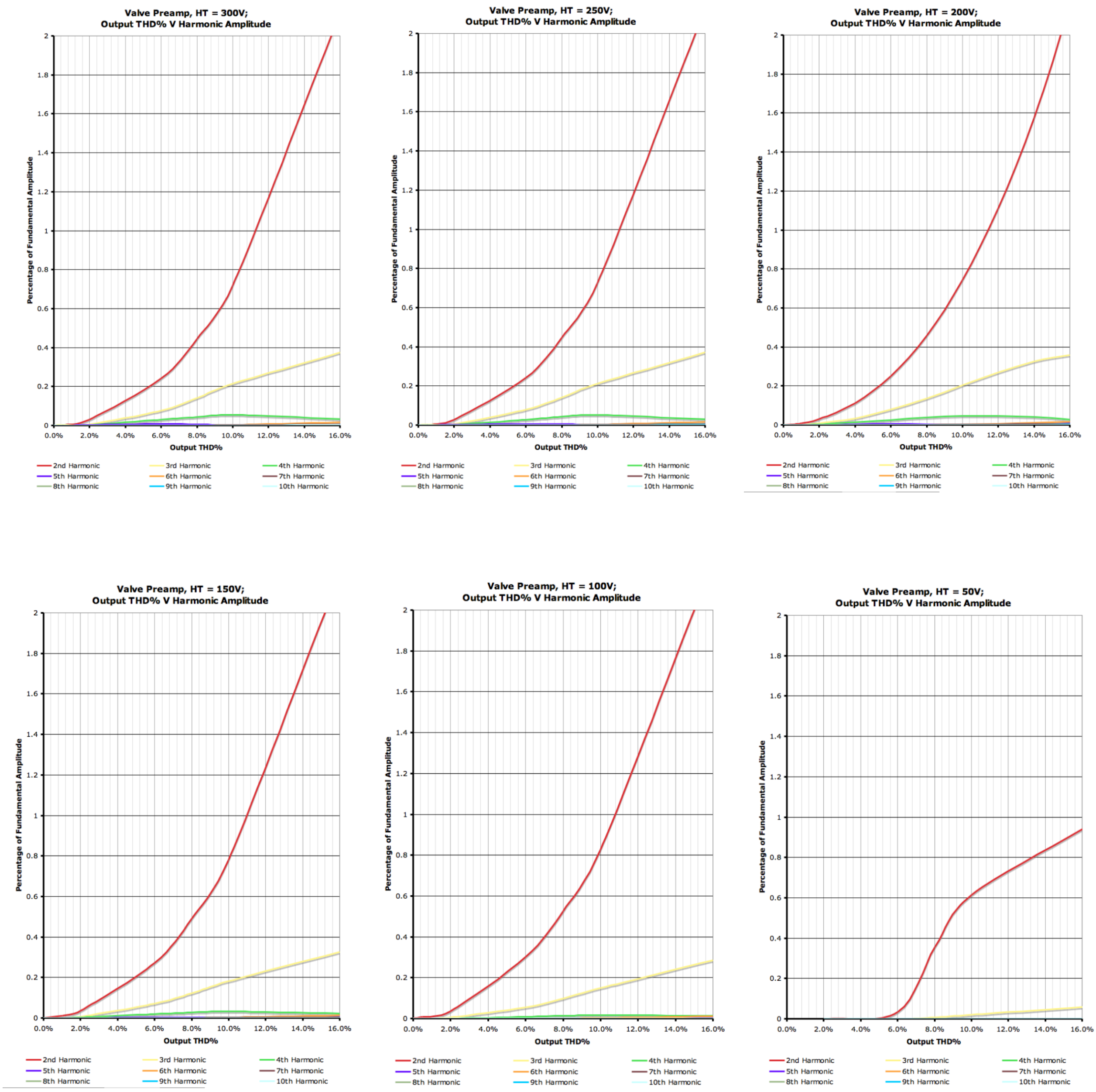

Figure 5.19 – Equivalent Hamm plots for reducing anode voltages.

(Full-scale plots found in Appendix 8.15.)

The 50V plot however, shows a distinct difference in low order harmonic weighting. As the grid leak current is further reduced the asymmetrical clipping has less affect and, as can be seen in figure 5.17, the output waveform at 50V is relatively symmetrical. This distributes the weighting of harmonics evenly amongst both odd and even order, significantly reducing the amplitude of the 2nd order harmonic.

Likewise, comparison plots between the valve and M/F preamps "primary colour characteristics" demonstrate extremely good correlation throughout supply reduction, figure 5.20.

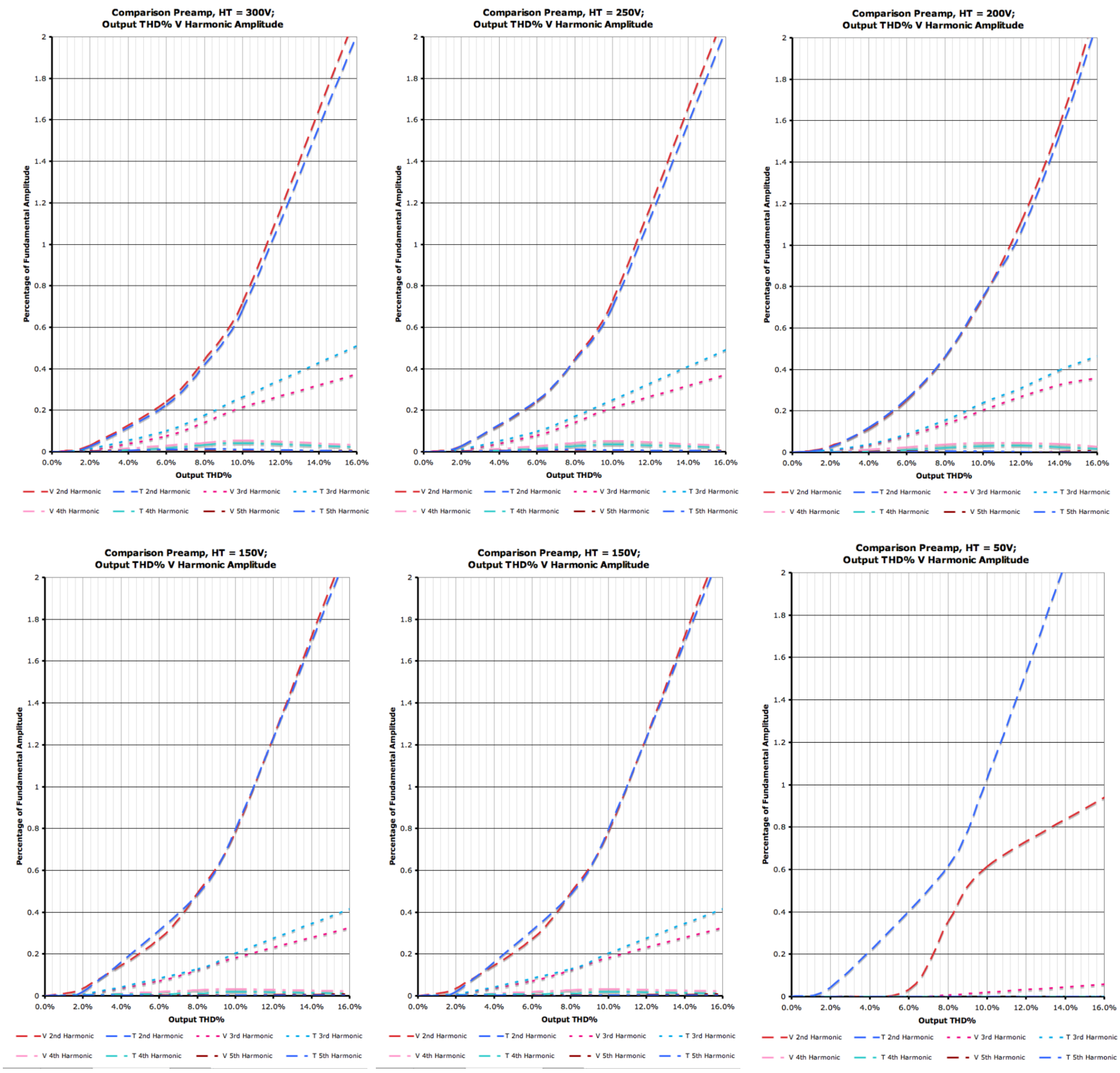

Figure 5.20 – Comparison of valve and M/F low order harmonics.

(Full-scale plots found in Appendix 8.16.)

Unlike the valve preamp, the M/F preamp continues to produce good correlation of its harmonic content at 50V. Because the asymmetrical clip of the M/F preamp is achieved by electronically increasing the bias point using a DC shift of the input signal, the 2^{nd} harmonic is still prevalent at lower supply voltages. This indicates that at amplifier powers involving supply voltages of less than $50V_{DC}$ the 'valve sound' may indeed have to be achieved using modelling techniques such as those used by Monteith and Flowers [54].

5.2.3- Subjective Pilot Evaluation of Valve Anode Reduction:

Objective measurements indicated that, when input signal amplitudes were controlled, there was little variation in harmonic weighting of the valves output signal. The most notable differences appeared from 8-16%THD, this region required subjective evaluation. Although the M/F preamplifier objectively demonstrated differences, subjective analysis was not carried out upon its output as a result of the already drawn conclusion that, with 95% confidence, the valve preamplifier produces more preferable distortion characteristics.

A pilot audio evaluation was conducted to determine if the affect of anode voltage reduction had any bearing upon perceived preference of output signal from 8-16%THD. The evaluation exploited the methodology described in section 4.4.4 with one change. The test would use a combination of principles from the MUSHRA and double blind A/B/X test methods. [70]

Clark [70] stated, "Listeners often fail to prove they can hear a difference after non-controlled listening suggested that there was one." For this test the primary evaluation aim was to determine if there was a perceivable difference in a *controlled* environment between recordings made at varying anode voltages. The secondary aim was to determine if there was a trend of preference. Thus the research question was derived:

"Can you identify a difference between the samples, if so, in relation to the reference, how much do you prefer the remaining samples distortion characteristics; More, less or the same?"

The same eleven test subjects were asked to listen to a reference sample then determine which of the two graded samples was the reference sample. A preference grade was then allocated to the remaining sample. The test instruction can be found in Appendix 8.17.

The recorded test samples were arranged into experiments such that each sample was individually compared with every other sample. The recordings at 50V were correctly identified all the time, 100V identified 55.6% of the time, and above 100V only 26% of the results were correctly identified as being the reference. This demonstrates that there is a perceivable difference between the output signals, but the difference is more noticeable at lower anode voltages. This relates well to the objective harmonic analysis that showed greater correlation at higher anode voltages than lower. It must be noted that the accuracy of these percentages would be vastly improved with a larger sample of subjects. The preference ratings from the correctly identified sample differences were collated, figure 5.21. These results were then compared with the relative recording signal/noise levels.

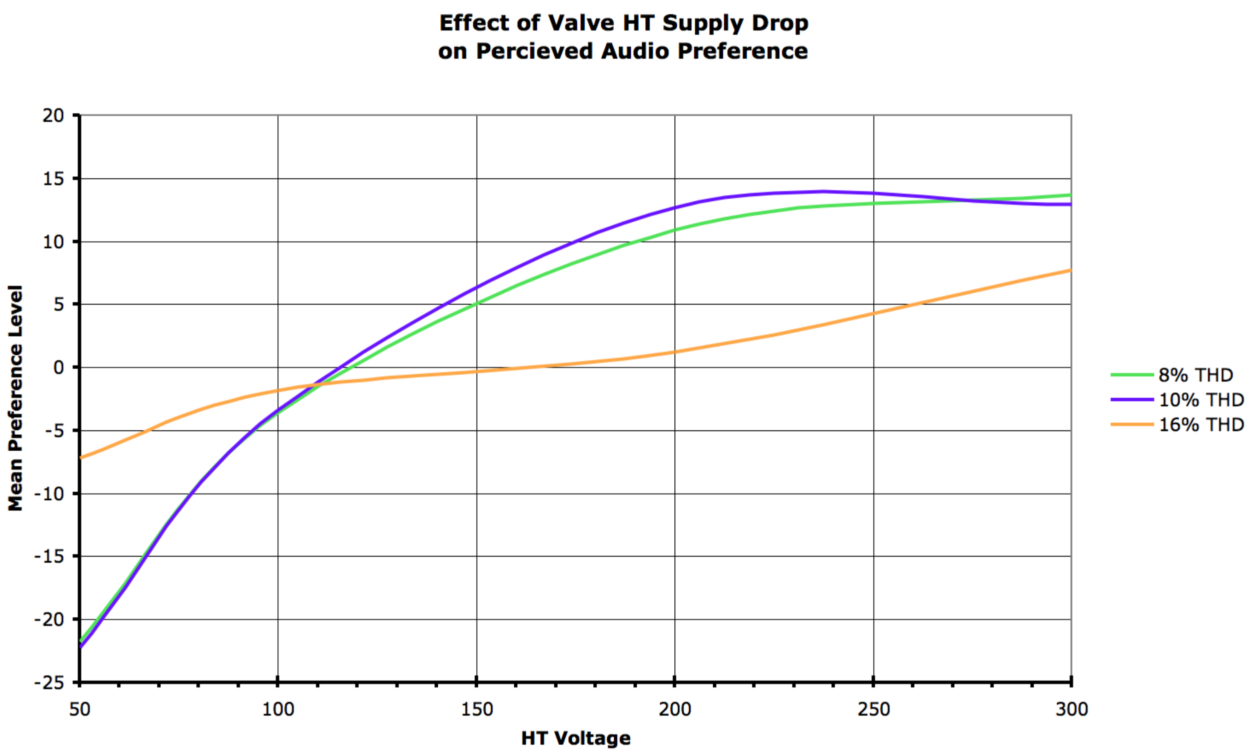

Figure 5.21 – Relative preference of sample as a result of varying anode voltage.

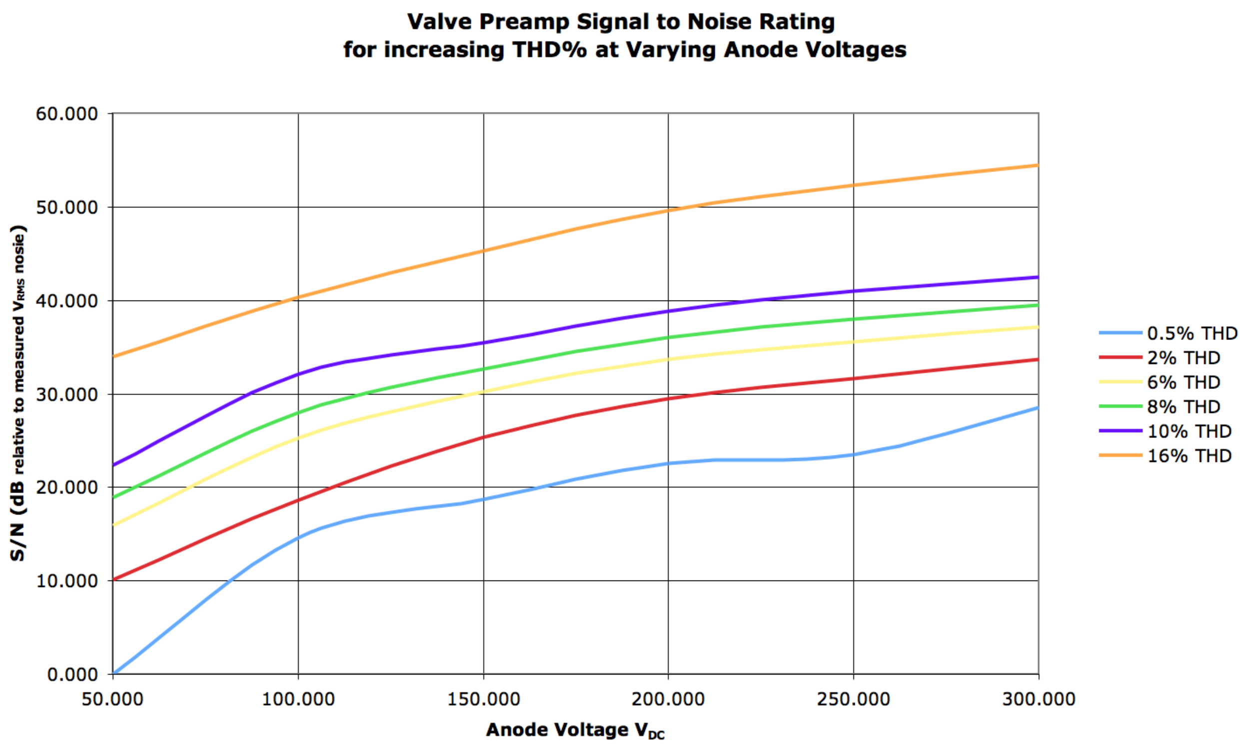

Figure 5.22 – S/N rating of recorded samples.

The general trend of the resulting curves for preference level show correlation with the slope of the S/N power ratios. This indicates that the S/N ratio has a significant bearing upon the preference of the distortion characteristics. The point where all the preferences merge (HT = 110V) coincides with a threshold of 30dB S/N. Above this threshold the perceived increase in preference is significantly reduced.

It is concluded from this pilot test that there are perceivable differences in output signals at reduced anode voltages. Nonetheless, without improving the S/N ratio from the output of the 003 it cannot be stated with any confidence whether the perceivable difference is due to objectively differing distortion characteristics or simply poor S/N power ratios.

Chapter 6 - Conclusions

The research section of this thesis clearly identified the main characteristics of the 'valve sound' and determined from which part of the valve topology the favourable characteristics yield. With these characteristics in mind alternative low power methods of simulating the 'valve sound' were analysed and the conclusion was made that digital simulation is lacking enough dynamic data on which modelling can be based. An ECC83 dual triode preamplifier was subsequently measured throughout its dynamic operating range for reactance to input signal amplitude, and harmonic content at set THD%s. The results were collated into an Excel database (attached to the back cover of this thesis).

A subjective and objective comparison between the valve preamplifier and a referenced high-voltage transistor topology [54] was conducted at an operating voltage of $200V_{DC}$. The analogue simulation demonstrated objective similarities with the valves low order harmonic components, particularly the defining characteristic of 2^{nd} order harmonic dominance. However, it was conclude with 95% confidence that the valve was subjectively preferable. This highlighted the sensitivity of the ears response to high order harmonic weighting.

Detailed analysis of the harmonic content of an ECC83 preamplifiers output signal was presented for varying anode voltages ranging from $50\text{-}300V_{DC}$. It was concluded that by controlling the amplitude levels of input signals to a system, the transfer characteristic could be 'scaled' down to achieve closely correlating low and high order harmonic content at a power reduction factor of 9.8 from maximum anode operation.

Running the ECC83 at $50V_{DC}$ produced a power reduction factor of 36.54, but the objective results exhibited little correlation in the weighting of the output harmonics. A subjective pilot test confirmed that below $110V_{DC}$ the output signal of a valve became less preferable, although it was inconclusive as to whether this was because of perceived distortion characteristics or poor signal to noise power ratios of below 30dB.

The conclusion of this thesis is that the thermionic valve can be used in low power guitar amplification when input signal amplitudes are reduced correspondingly. Noticeable differences in output harmonic weighting at anode voltages below $110V_{DC}$ can be audibly perceived and consequently, below this threshold voltage, valve simulations must be used. Therefore, development of a dynamic triode model, based on 'real world' measurements, is required before any further accurate reduction in the power of the 'valve sound' can be achieved.

Ultimately, the objectives of this Thesis have been satisfied.

Chapter 7 - Further Work

As a result of this thesis a number of avenues for further work have become apparent. Foremost is the requirement of an accurate dynamic valve model, which does not require the computational needs encompassed by the use of the Newton-Raphson algorithm to create stability for both differential equations and nonlinear implicit equations.

To achieve such a dynamic model, the production of a database of 'real world' dynamic valve parameter measurements is a necessity. Although this thesis has provided a portion of this data it ultimately has only objectively analysed the effects of a 1kHZ sine wave of varying amplitude on the harmonic content of a 12AX7/ECC83s output. The harmonic content data must be expanded to include the valves reaction to frequencies spanning the entire audible frequency range of 20-20,000 Hertz, and even beyond that.

A possible further research project could utilise such a collation of dynamic data to produce a hybrid analogue/digital topology that constantly monitors input signal amplitudes and frequencies, introducing the required measured harmonic content at its output. Such a topology would require integration of A/D converters and microcontrollers to either digitally produce the output sound, or even to control the dynamic parameters of the analogue thermionic valve itself.

Further analysis is also required of the M/F topology. It was somewhat of a surprise that the results of the pilot audio evaluation so conclusively demonstrated the preference of the triode preamplifier. The objective comparisons alluded to a large amount of correlation between the low order harmonic weightings. This suggests that un-tested factors other than the differing high order harmonic weighting such as transient response and slew rate, may be responsible for the subjective differences. In addition to this, the M/F demonstrated excellent correlation of its "primary colour characteristics" which would suggest that further research could produce an accurate analogue model of the valve for low power reproduction of the 'valve sound'.

Finishing with the idea of incorporating the emissive properties of single wall carbon nanotubes, as researched by NASA [12], into audio valves; the possibilities for further work are never ending! Ultimately it is the authors hope that regardless of whether or not production of the thermionic valve continues in the future, the sound of the valve amplifier will continue to be a corner stone of the audio industry.

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Chapter 8 - Appendices

8.1 - Material Emissivity Chart:

Element	eV	Element	eV	Element	eV	Element	eV	Element	eV
Ag:	4.52-4.74	Al:	4.06-4.26	As:	3.75	Au:	5.1-5.47	B:	~4.45
Ba:	2.52-2.7	Be:	4.98	Bi:	4.34	C:	~5	Ca:	2.87
Cd:	4.08	Ce:	2.9	Co:	5	Cr:	4.5	Cs:	2.14
Cu:	4.53-5.10	Eu:	2.5	Fe:	4.67-4.81	Ga:	4.32	Gd:	2.90
Hf:	3.9	Hg:	4.475	In:	4.09	Ir:	5.00-5.67	K:	2.29
La:	4	Li:	2.93	Lu:	~3.3	Mg:	3.66	Mn:	4.1
Mo:	4.36-4.95	Na:	2.36	Nb:	3.95-4.87	Nd:	3.2	Ni:	5.04-5.35
Os:	5.93	Pb:	4.25	Pd:	5.22-5.6	Pt:	5.12-5.93	Rb:	2.261
Re:	4.72	Rh:	4.98	Ru:	4.71	Sb:	4.55-4.7	Sc:	3.5
Se:	5.9	Si:	4.60-4.85	Sm:	2.7	Sn:	4.42	Sr:	~2.59
Ta:	4.00-4.80	Tb:	3.00	Te:	4.95	Th:	3.4	Ti:	4.33
Tl:	~3.84	U:	3.63-3.90	V:	4.3	W:	4.32-5.22	Y:	3.1
Yb:	2.60 [2]	Zn:	3.63-4.9	Zr:	4.05				

Figure 8.1 – Material emissivity table.

8.2 - Effect of Anode Voltage Reduction on Primary Parameters:

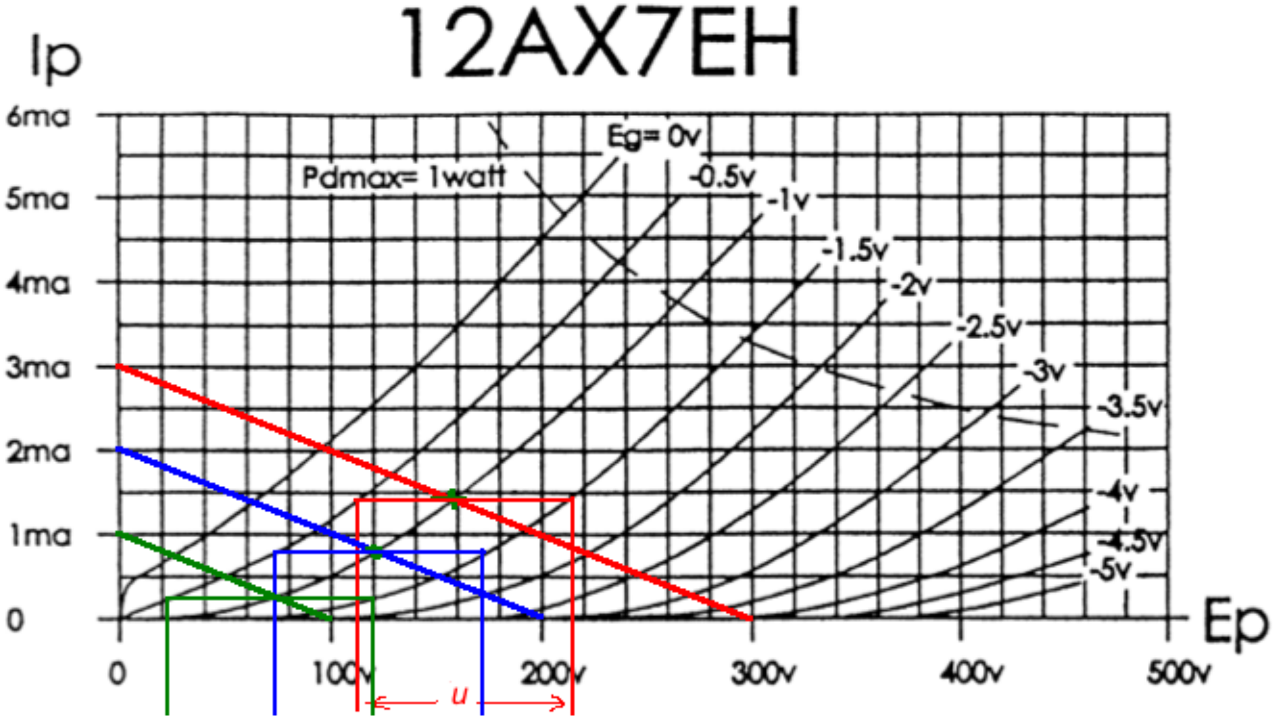


Figure 8.2 - Effect of reducing anode voltage on the amplification factor.

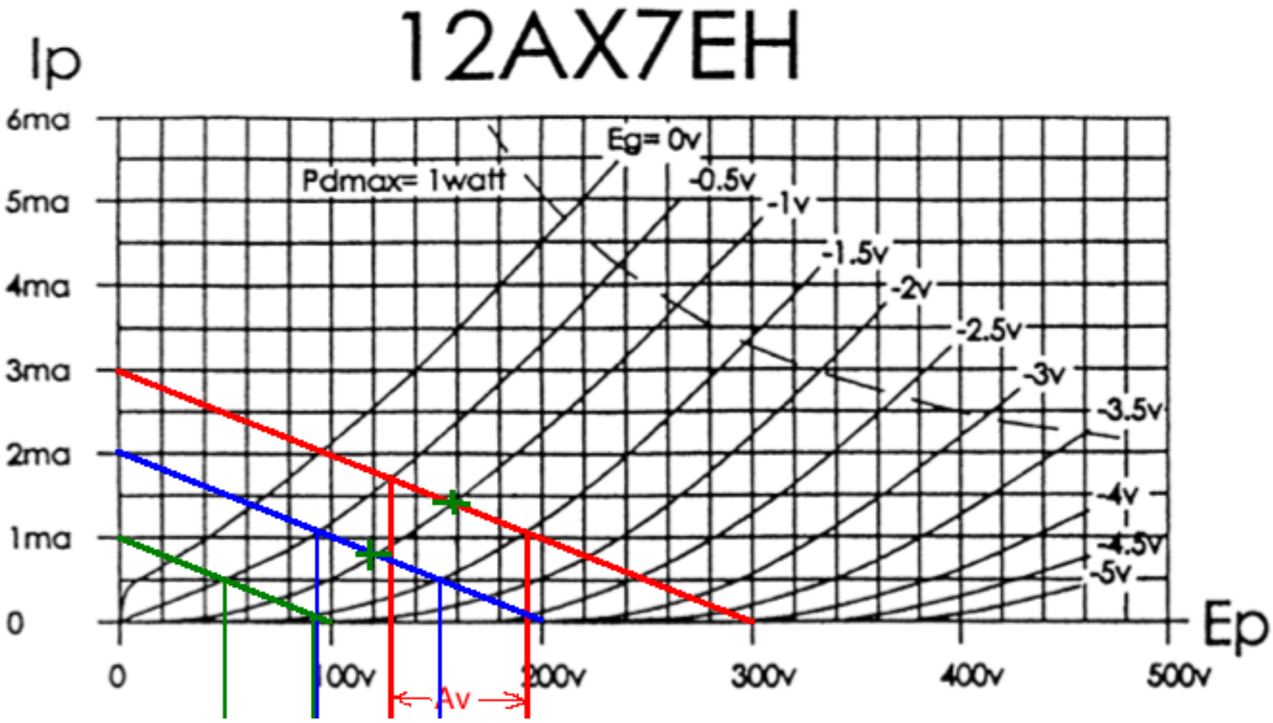


Figure 8.3 - Effect of reducing anode voltage on the stage gain.

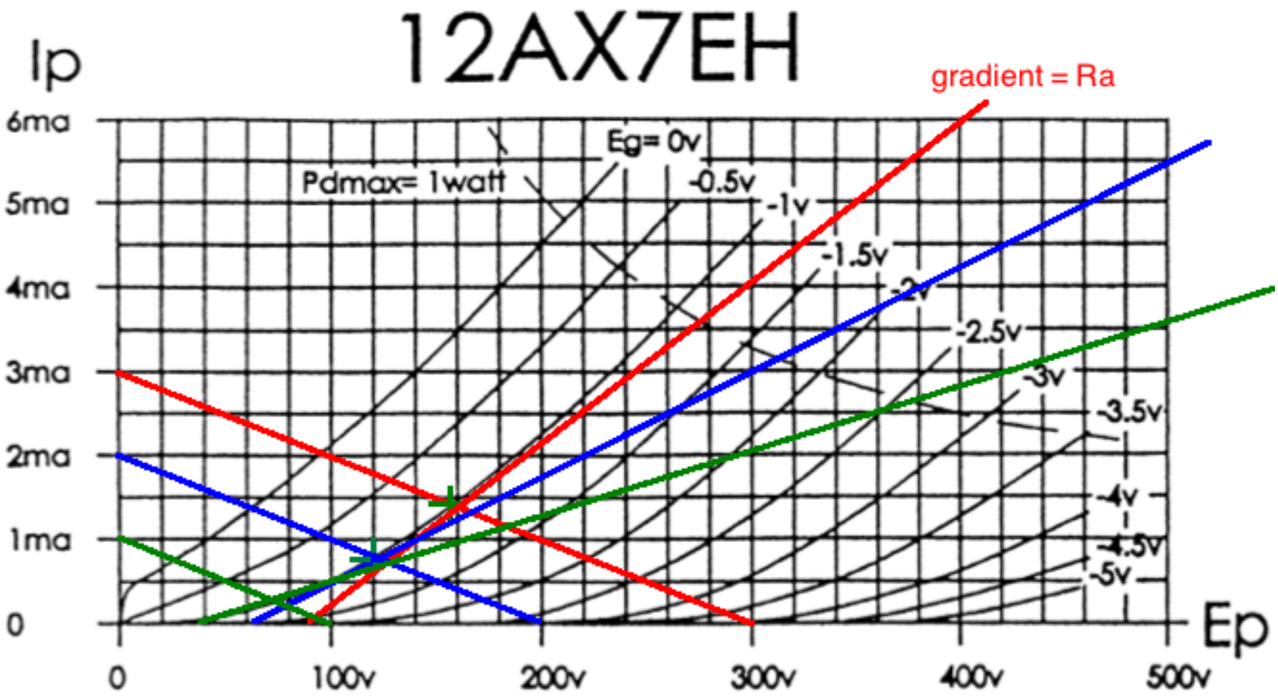


Figure 8.4 - Effect of reducing anode voltage on the anode resistance.

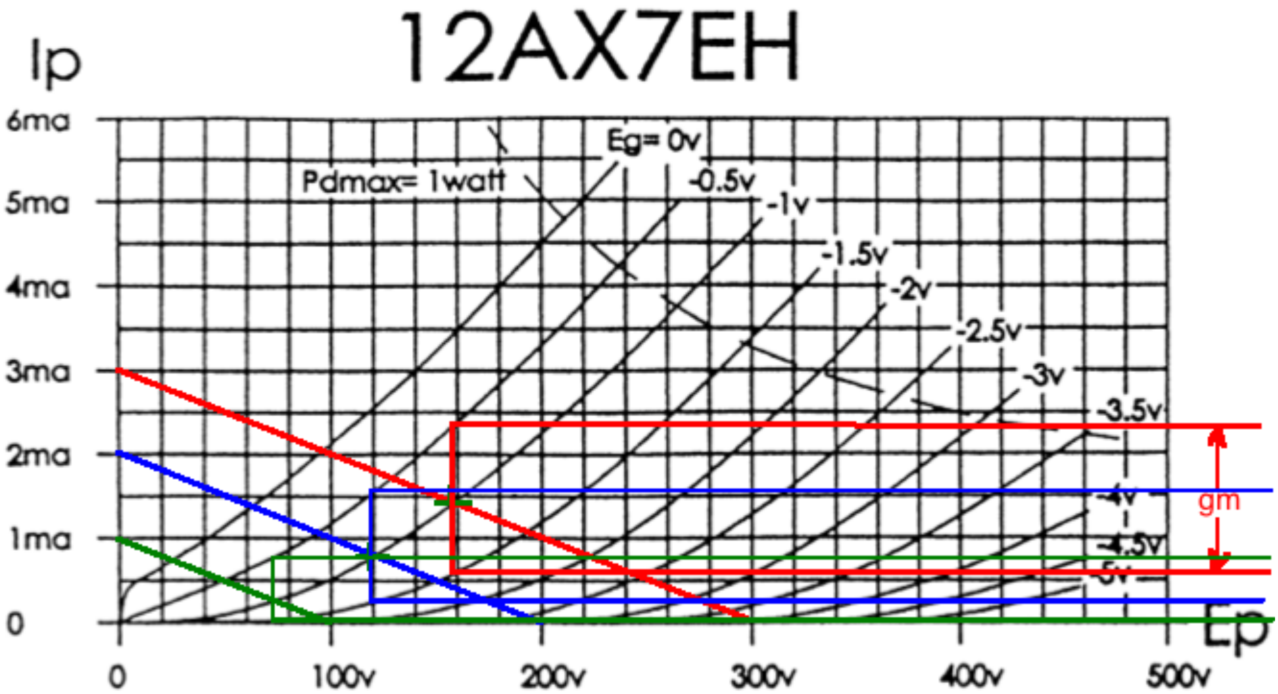


Figure 8.5 - Effect of reducing anode voltage on the mutual conductance.

8.3 - Electo Harmonix 12AX7 Datasheet:

ELECTRO-HARMONIX

12AX7EH

The 12AX7EH is a dual high mu triode with a spiral filament and special construction to minimize microphonic behavior. Ideal for replacement use in guitar amps and any place low noise and high gain is important.

Pin #	description
1	plate 2
2	grid 2
3	cathode 2
4,5,9	heater
6	plate 1
7	grid 1
8	cathode 1

Electrical Data	
Heater Voltage, not less than	6.0 or 12.0 V
Heater Voltage, not more than	6.6 or 13.2 V
Plate Voltage, not more than	300 V
Heater to Cathode Voltage:	
positive, V not more than	100 V
negative, V not less than	200 V
Plate Current, not more than	9 mA
Plate Dissipation, each triode, not more than	1.0 watts
Maximum grid circuit resitance:	
fixed bias, not more than	1 Mohm
self bias, not more than	2.2 Mohm
Inter-electrode Capacitances:	
C, grid to plate	1.6 pF (triode 1 and 2)
C, grid to cathode and heater	1.6 pF (triode 1 and 2)
C, plate to cathode and heater	0.44 pF (1) and 0.36 (2)
C, cathode to heater	5.0 nF (nominal)
C, plate to plate	520 pF
Measured Electrical minima:	
Grid reverse current, not more than (see note below)	0.2 uA
Plate current, not less than (see note below)	0.75 mA
Plate current (Eb= 250V, Ec= -4V)	10 uA
Transconductance, not less than (see note below)	1.4 mA/V
Amplification Factor, not less than (see note below)	78
Amplification Factor (nominal)	92
Transconductance (nominal)	1.7 mA/V
Plate Resistance (nominal)	54.1 K OHM
Max Neg. Grid Voltage	55 V
Max Pos. Grid Voltage	0 V
Max Cold Voltage	600 V

NOTE: heater V, 12.6vac; plate V, 250v; grid bias, -2v; grid circuit resistance, 1K ohm

12AX7EH
NEW SENSOR CORP.
tested by jcm
Drawing # GT006
Drawn by: jcm

Figure 8.6 – 12AX7 Datasheet.

8.4 - Preamplifier Comparison Circuit:

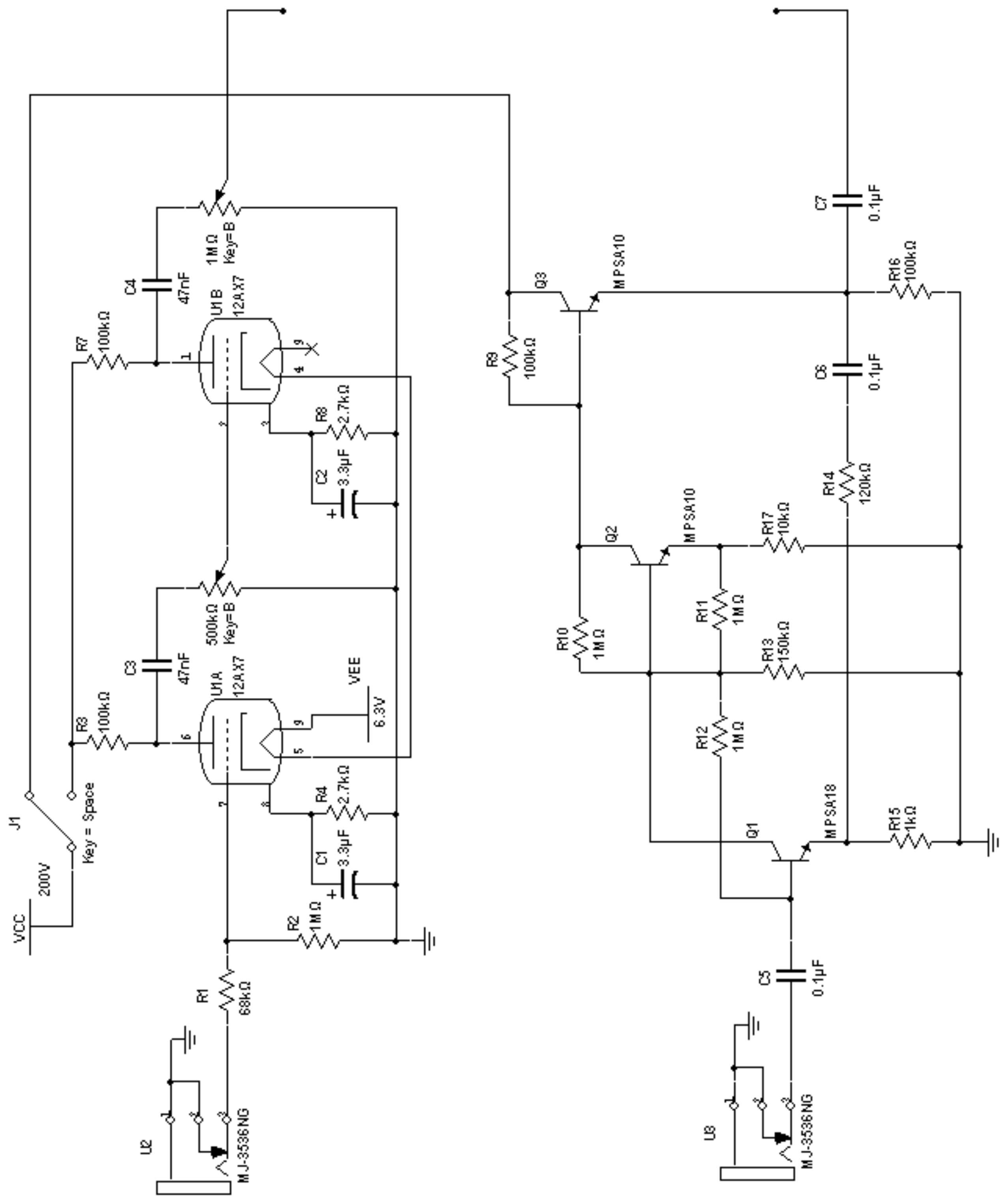


Figure 8.7 – Monteith/Flowers Preamp and comparable ECC83 Preamp.

8.5 - Risk Assessment for Testing:

UNIVERSITY OF HUDDERSFIELD - PROJECT HEALTH AND SAFETY RISK ASSESSMENT FORM

(To be completed for intended and proposed activities)

Brief description of activity: Research into operation of valve amplifiers at under specification voltages.			
Location: T5/02	**Assessment by: Mike Aitchison/Dennis Town/Steve Fenton**	**Assessment date: 11/05/2010**	**Assessment reference:**

SPECIFIC TASK/ASPECT OF ACTIVITY: Work Relating to Actual Project and HIGH VOLTAGES						
Hazards identified	**Risks to health and safety**	**People at risk**	**Measures to manage the risks effectively**	**Action by:**		
				Who	**When**	**Completed**
Electric shock. Unwanted power up of equipment. Tampering of project equipment by other laboratory users.	Maximum Voltages of 400V DC Exposed terminals of Variac. Radiated heat from glass of valves. Power up of equipment while project left unattended.	Self and other users of laboratory while project unattended.	- Good earth contact at all times, all equipment connected to a proper mains earth. - System has a connected earth leakage. - Internal voltages of up to 400V DC are totally enclosed while testing. - Only working on the internal circuit while supply is dead. - Spatial awareness of surroundings, preventing knocks from other persons within the laboratory. - Working with only one hand when using test probes to measure and debug circuit parameters thus preventing any electrical passage across the chest. - Work in an isolated section of the laboratory with no crowding of project area. - Never work on anything while unattended within the laboratory. - No exposed terminals, covered by insulated materials. - All equipment turned off and disconnected from the mains if left unattended. - Sensible work ethic, break for 20 minutes every second hour.	MA / DT / SF	18/05	YES

Version: Mike Aitchison – U0401955 – Masters by Research Project Risk Assessment.

SPECIFIC TASK/ASPECT OF ACTIVITY: General Laboratory Use						
Hazards identified	**Risks to health and safety**	**People at risk**	**Measures to manage the risks effectively**	**Action by:**		
				Who	**When**	**Completed**
Use of laboratory equipment. Workspace usage. Other Electrical Risks	Poor workstation layout and insufficient working space. Obstructions and trailing cables. Use of laboratory electrical equipment. Over loading of electrical sockets.	Anyone within the lab, coming into contact with appliances.	- Desk used to be in the corner of the lab away from main walkway. - Regular desk checks. - Avoidance of trailing cables across walkways. Where unavoidable, cover with cable mats. - Appliances subject to appropriate levels of maintenance that is recorded. - Extension cables used only as a temporary measure and positioned away from areas of potential damage. - Sufficient number and position of fixed socket outlets.	**MA / DT / SF**	**18/05**	**YES**

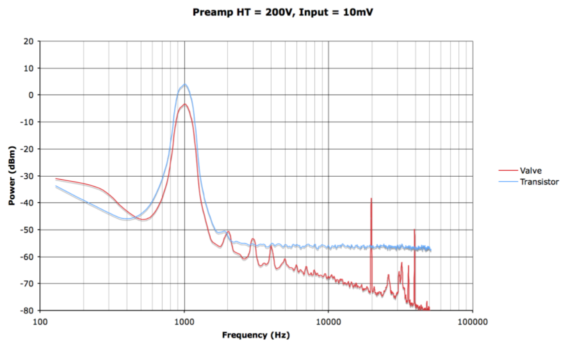
RISK ASSESSMENT REVIEW:

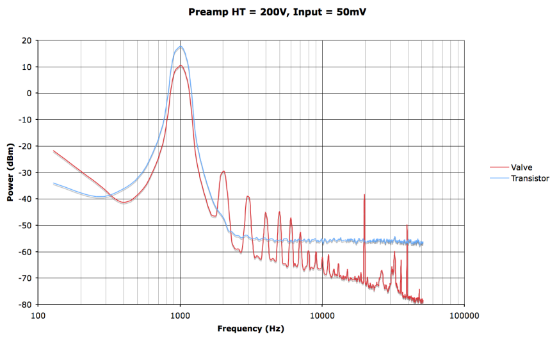
Date to be carried out by: **19/05/2010**

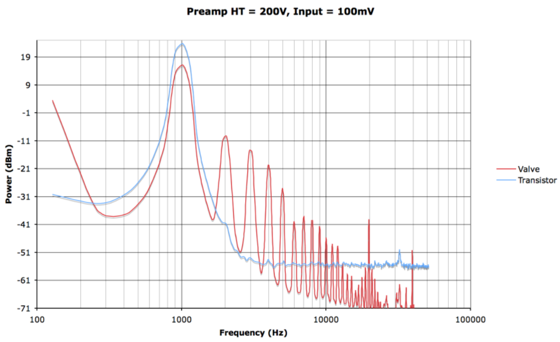
Signed: ....Mike Aitchison / Steve Fenton...................................... Date: ....18/05/2010..........................

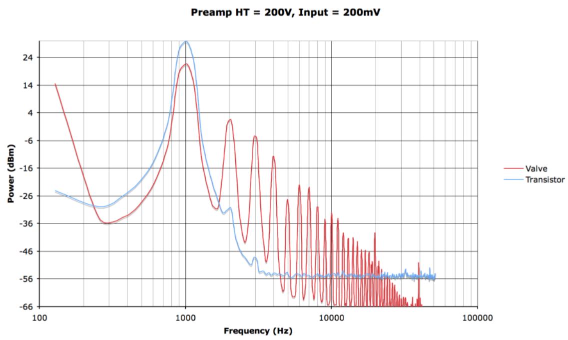
8.6 - Output Effects of Increased Input Voltage:

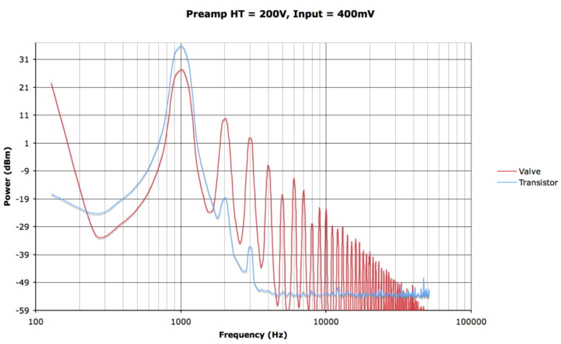
The full range of plots in increments of 10mV from 10mV to 3.00V is available as part of the full data set. These plots have been chosen because they demonstrate the properties discussed in the main text.

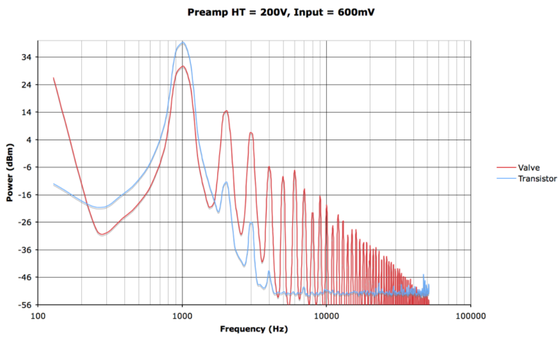


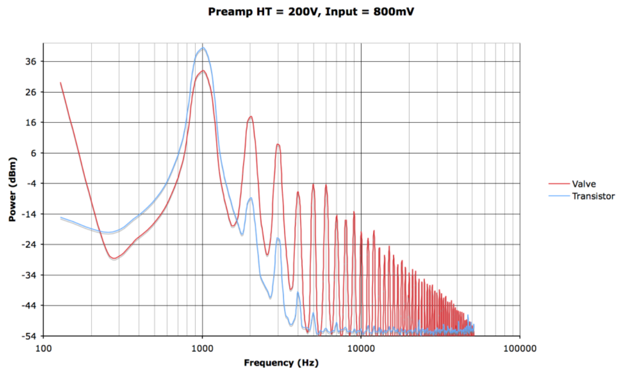


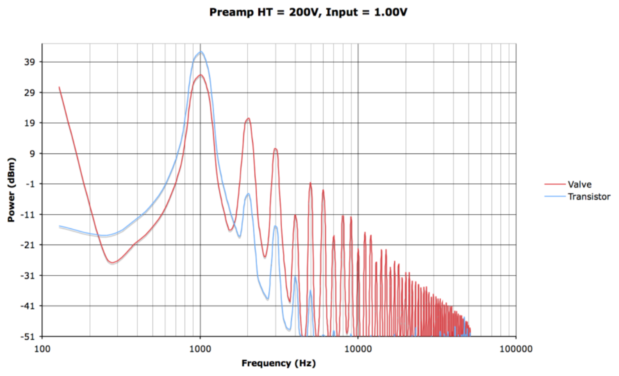


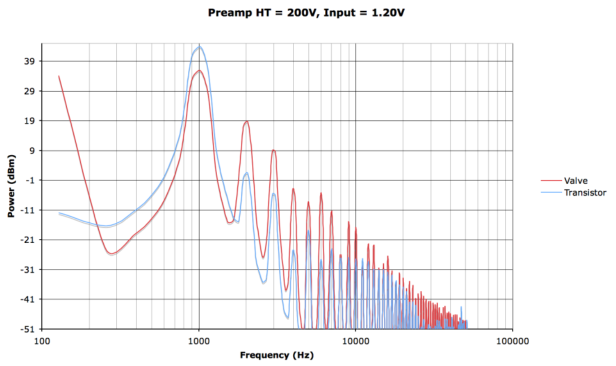


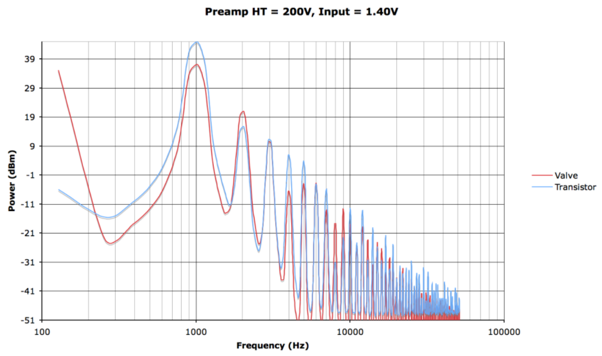


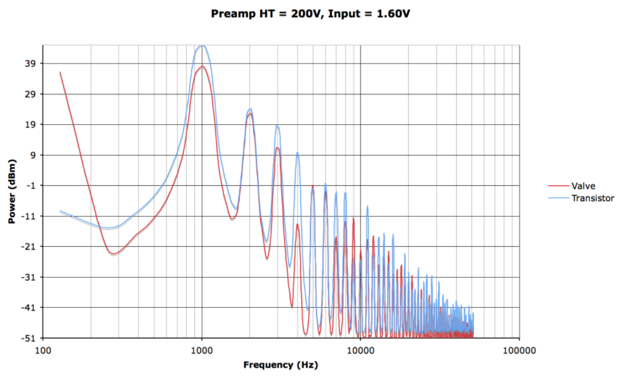


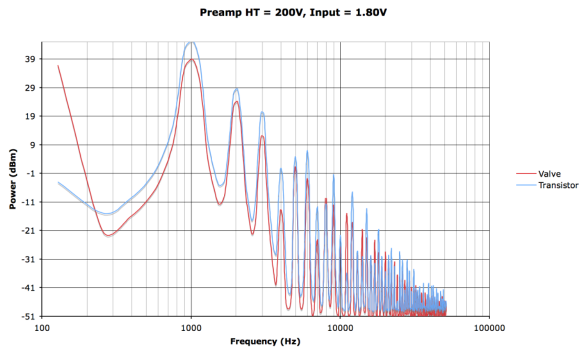

Preamp HT = 200V, Input = 800mV
Power (dBm)
Frequency (Hz)
Valve
Transistor


Preamp HT = 200V, Input = 1.00V
Power (dBm)
Frequency (Hz)
Valve
Transistor

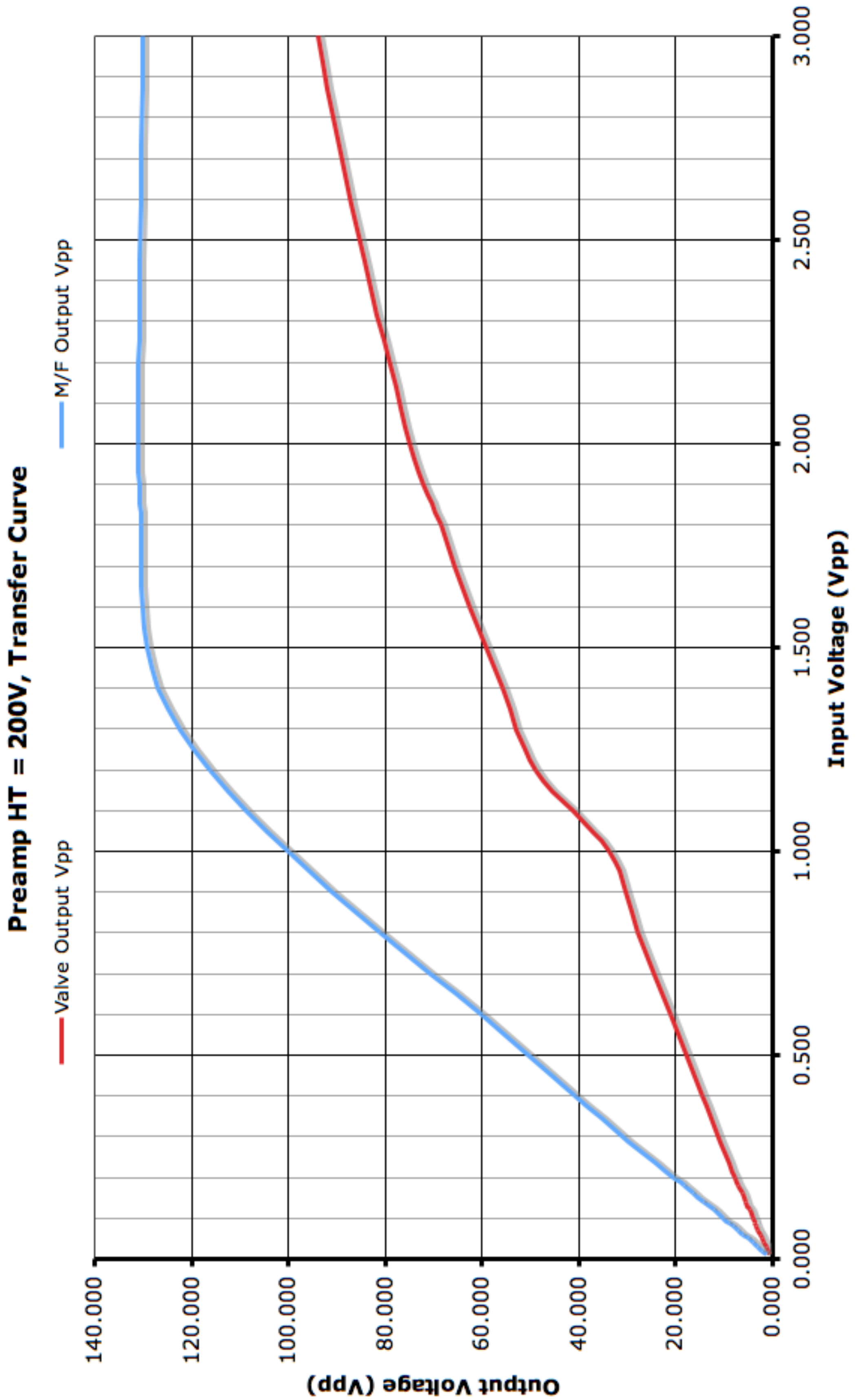

Preamp HT = 200V, Input = 1.20V
Power (dBm)
Frequency (Hz)
Valve
Transistor

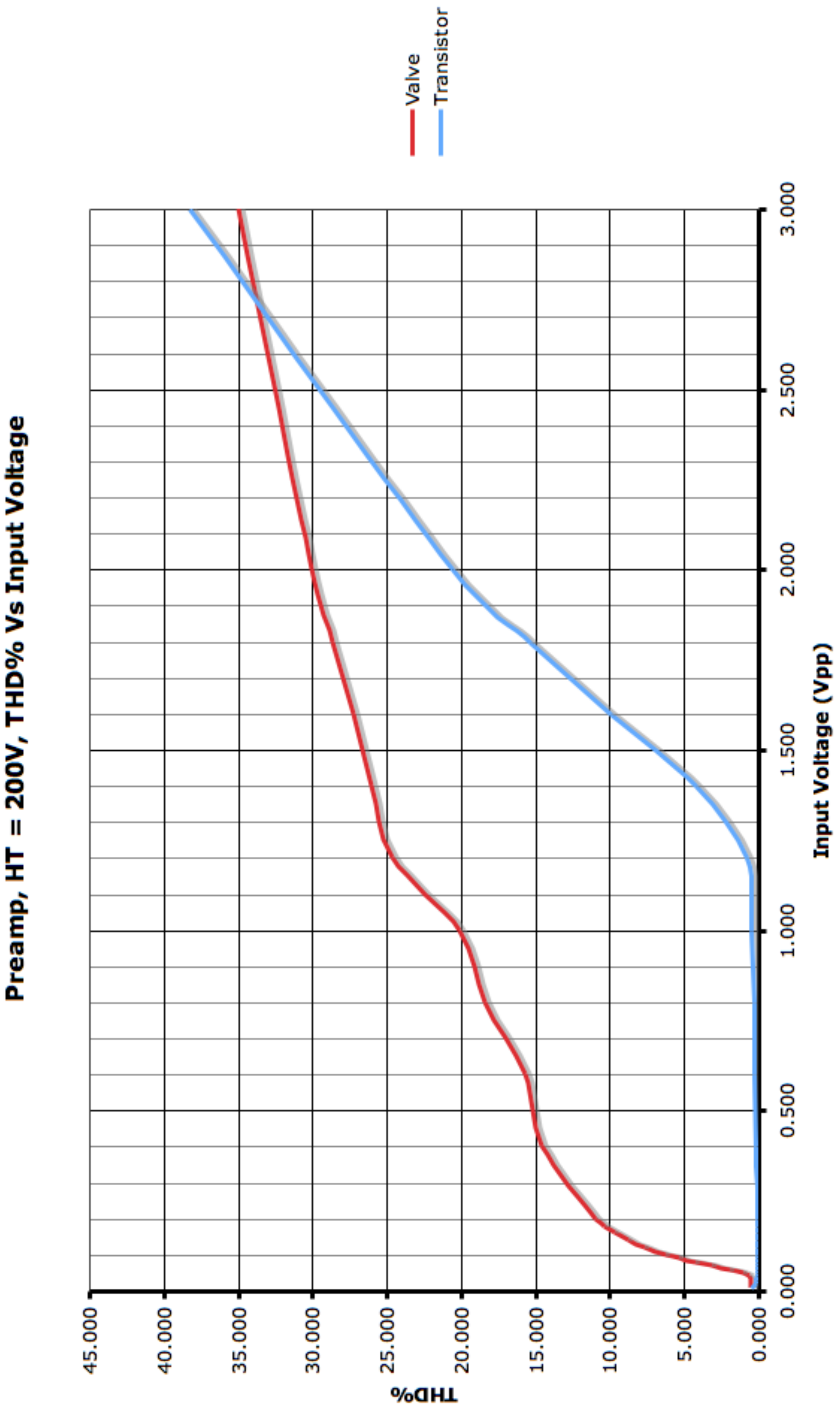

Preamp HT = 200V, Input = 1.40V
Power (dBm)
Frequency (Hz)
Valve
Transistor


Preamp HT = 200V, Input = 1.60V
Power (dBm)
Frequency (Hz)
Valve
Transistor

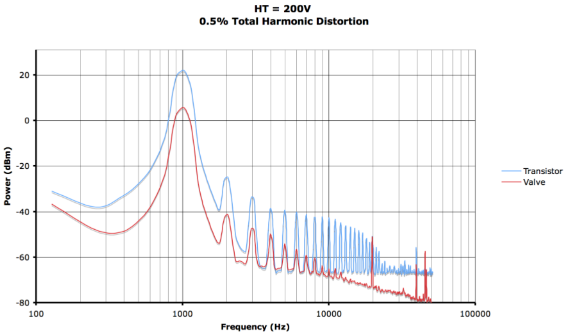

Preamp HT = 200V, Input = 1.80V
Power (dBm)
Frequency (Hz)
Valve
Transistor

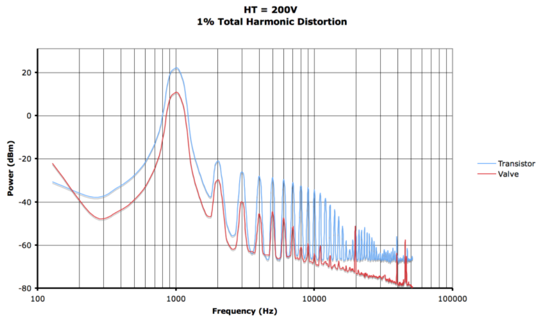
8.7 - 200V HT Transfer Curve and Relative THD%:

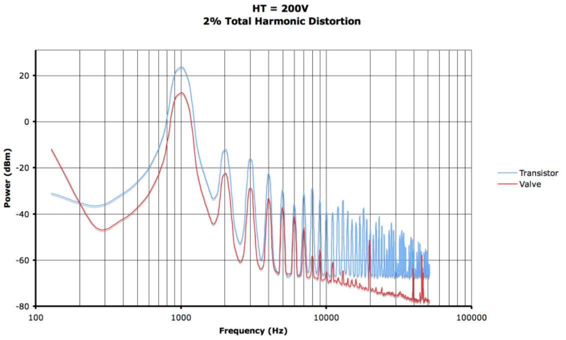


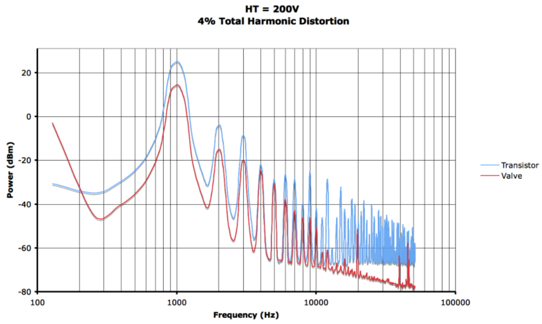

Preamp, HT = 200V, THD% Vs Input Voltage
Valve
Transistor
THD%
45.000
40.000
35.000
30.000
25.000
20.000
15.000
10.000
5.000
0.000
Input Voltage (Vpp)
0.000
0.500
1.000
1.500
2.000
2.500
3.000

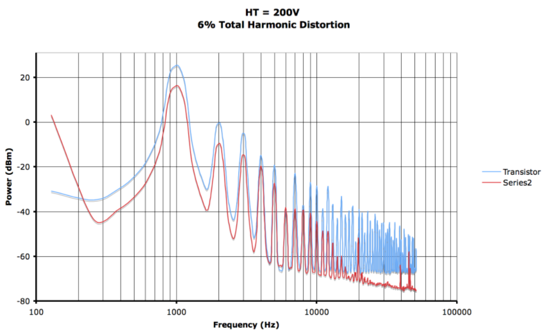
8.8 - Matched Output THD% at HT = 200V:

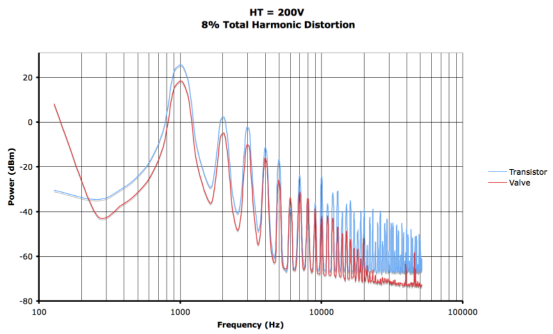


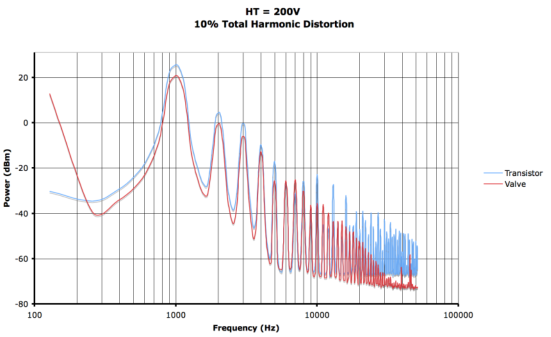


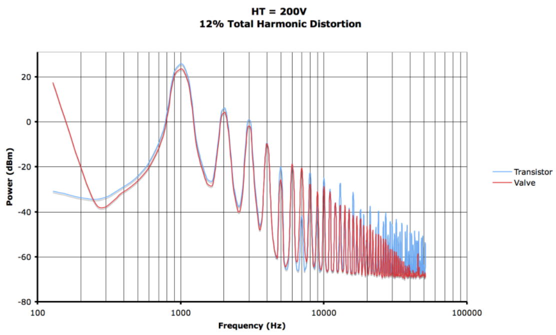


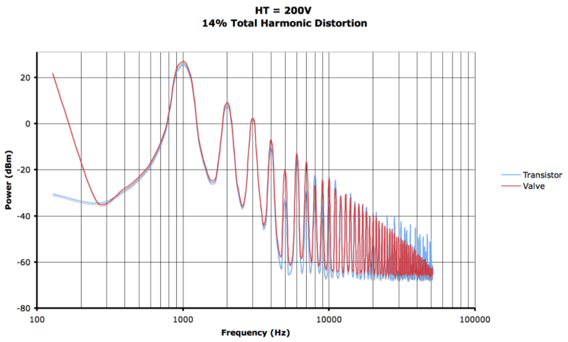


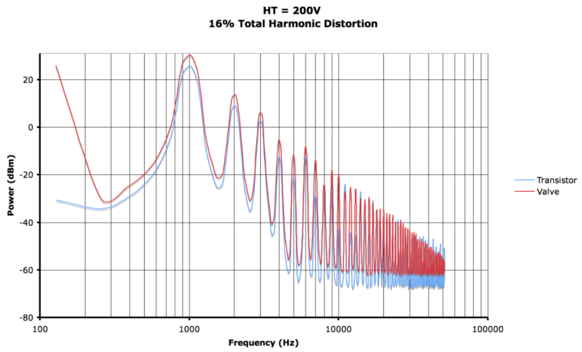




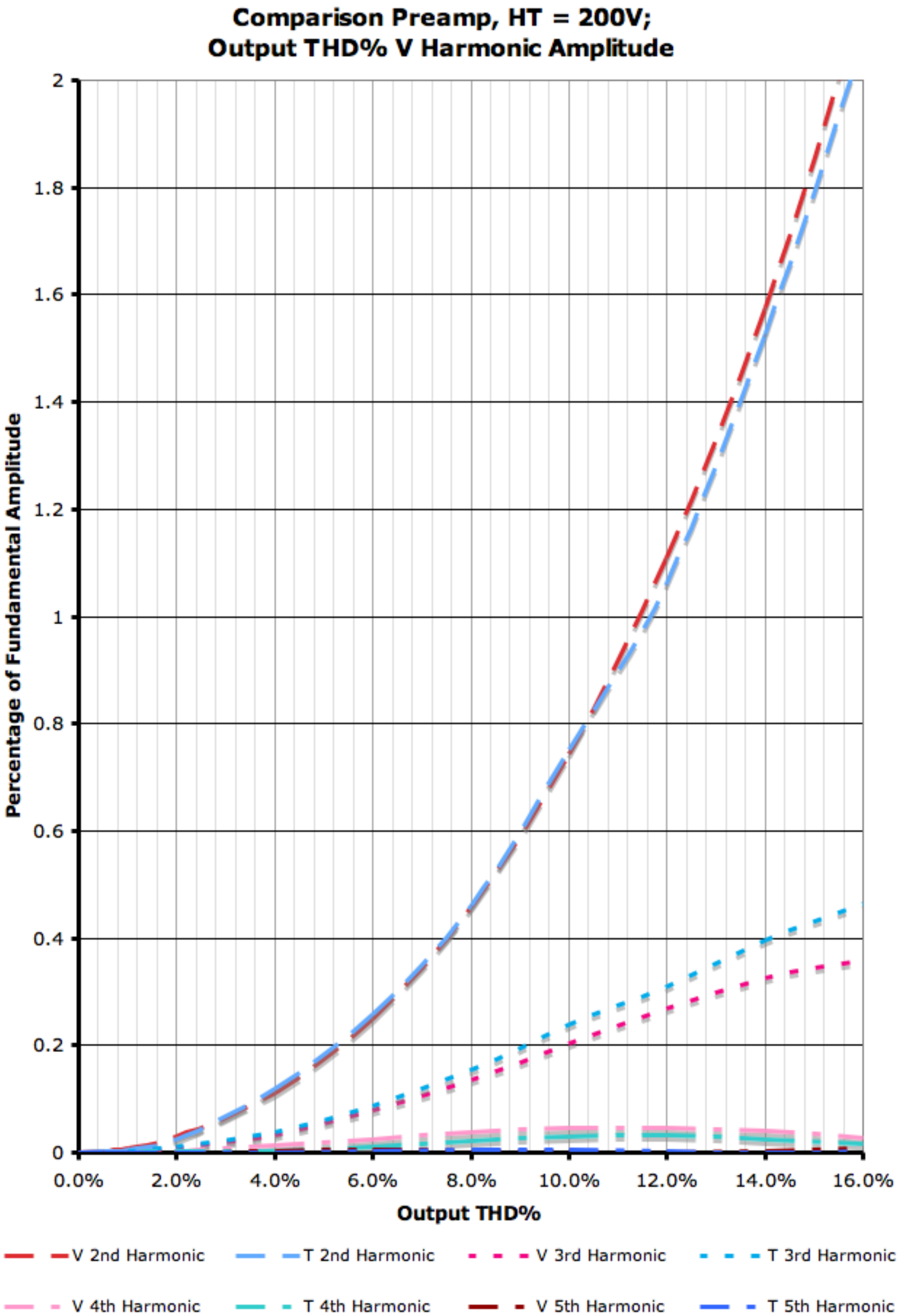


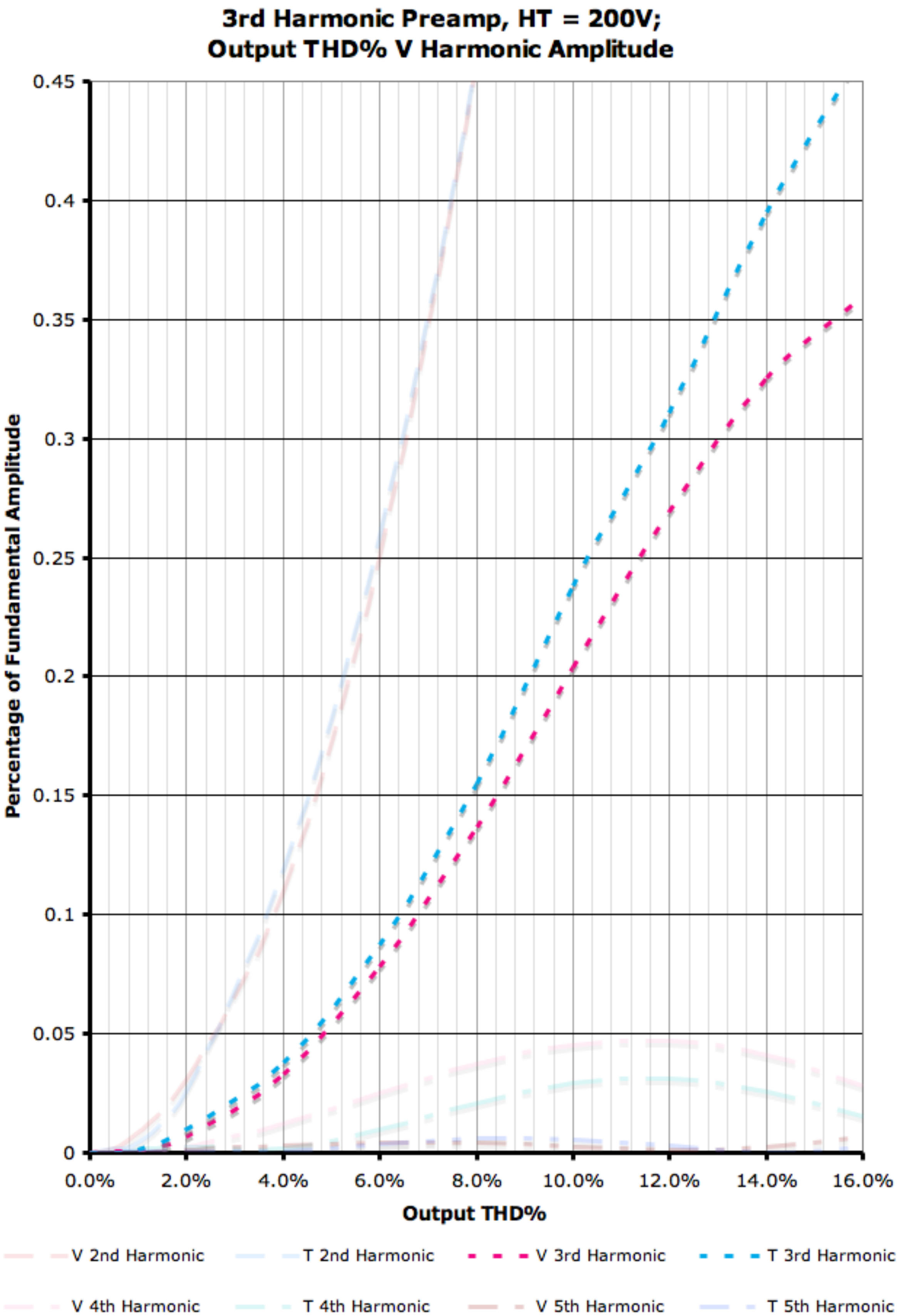


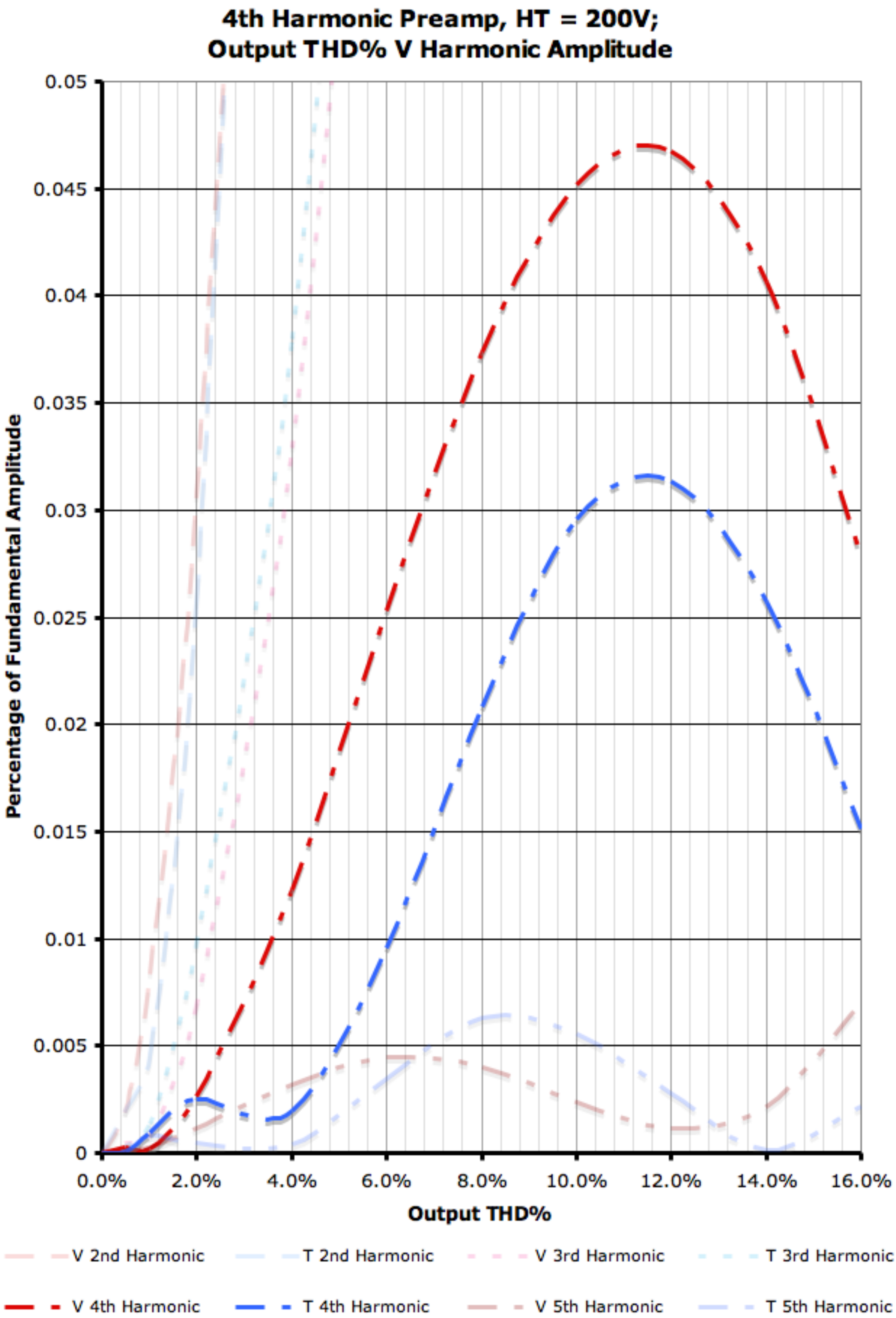

HT = 200V
14% Total Harmonic Distortion
Power (dBm)
Frequency (Hz)
Transistor
Valve
20
0
-20
-40
-60
-80
100
1000
10000
100000

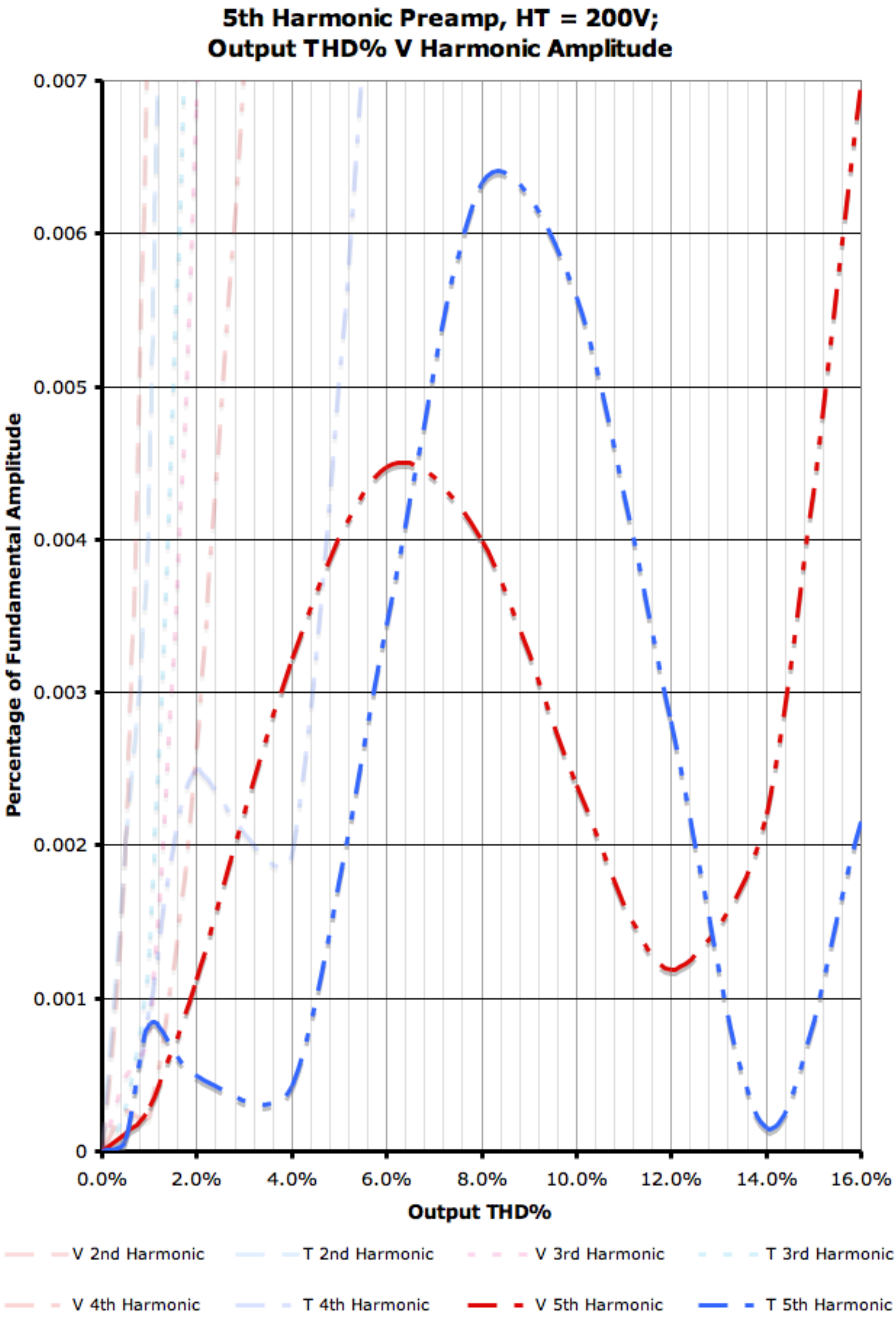

HT = 200V
16% Total Harmonic Distortion
Power (dBm)
Frequency (Hz)
Transistor
Valve
20
0
-20
-40
-60
-80
100
1000
10000
100000

8.9 - Comparison of Low Order Harmonics at 200V:




3rd Harmonic Preamp, HT = 200V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
0.45
0.4
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
V 2nd Harmonic
T 2nd Harmonic
V 3rd Harmonic
T 3rd Harmonic
V 4th Harmonic
T 4th Harmonic
V 5th Harmonic
T 5th Harmonic


4th Harmonic Preamp, HT = 200V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
0.05
0.045
0.04
0.035
0.03
0.025
0.02
0.015
0.01
0.005
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
V 2nd Harmonic
T 2nd Harmonic
V 3rd Harmonic
T 3rd Harmonic
V 4th Harmonic
T 4th Harmonic
V 5th Harmonic
T 5th Harmonic


5th Harmonic Preamp, HT = 200V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
0.007
0.006
0.005
0.004
0.003
0.002
0.001
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
V 2nd Harmonic
T 2nd Harmonic
V 3rd Harmonic
T 3rd Harmonic
V 4th Harmonic
T 4th Harmonic
V 5th Harmonic
T 5th Harmonic

8.10 - Audio Evaluation Subject Instruction:

Distortion Characteristic Preferences

Instructions for the Undertaking of An Audio Evaluation

Thank you for participating in the audio evaluation stage of this research project. Please read through these instructions and take time to understand the research question;

"In relation to the reference, how much do you prefer the samples whilst comparing their distortion characteristics; More, less or the same?"

The audio evaluation should last on average approximately 15-20 minutes. There are six experiments, each with ten samples nine seconds in length. Before beginning the grading phase each subject is asked to familiarize themselves with the media used for testing.

Training Phase:

You will be presented with a grid of all the samples present in the ensuing evaluation. The overall aim of this training phase is to become familiar with the testing process. This should hopefully allow you as the listener to achieve two objectives:

- To become familiar with all the sound excerpts under test and their relative quality level ranges; and

- To learn how to use the testing user interface.

In this phase you will be able to listen to all the sound excerpts that have been selected for the tests in order to illustrate the whole range of possible qualities. Figure 1. Shows the training phase user interface. You may click on any buttons to play the chosen sample including the reference, however once clicked, the sample will play in full. Only once a sample has finished playing will another sample be able to be selected to enable the listener to appreciate the range of differing quality levels and differing distortion characteristics available to hear during the grading phase.

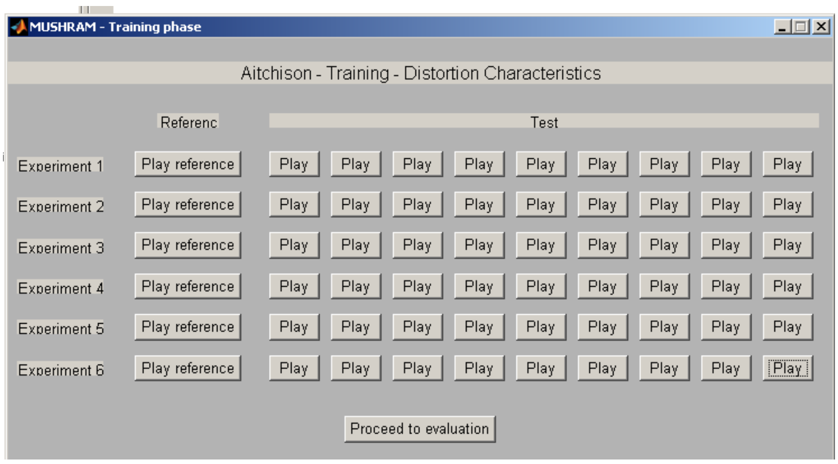


Figure 1. - The Training Phase General User Interface (GUI)

During the training phase you should be able to learn how you, as the individual, interpret the audible distortion characteristics and begin to formulate an idea of personal preference. You should not discuss these preferences or interpretations with any other subjects at any time during the training phase.

It is recommended that you take a few minutes break before beginning the grading phase to prevent ear fatigue during the test.

Grading Phase:

The grading phase is the actual testing phase of the audio evaluation. You are invited to assign your grades using the quality scales provided. Said scale is a scale of preference in relation to the reference sample, positive grading being a more preferable distortion characteristic, negative grading being a less preferable distortion characteristic.

The audio evaluation should last on average approximately 15-20 minutes. There are six experiments, each with ten samples nine seconds in length. You should listen to the reference and all the test samples by clicking on the respective buttons. You may listen to the samples in any order and any number of times. Unlike the training phase each sample will start anew when clicked. This should ease the process of directly comparing the samples with each other to allow grading. If at any point you wish for a few seconds silence, a “Stop All” button is provided.

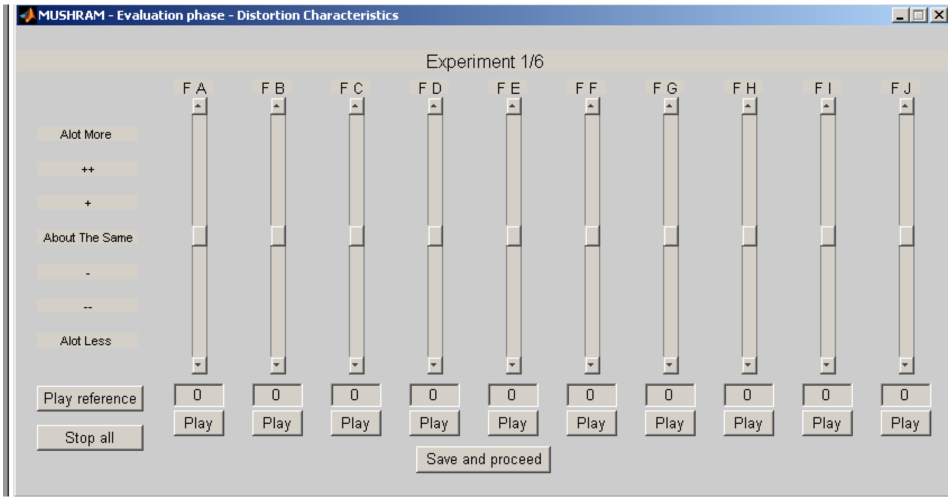


Figure 2. - The Grading Phase General User Interface (GUI)

Figure 2 shows the user interface for the grading stage of testing. Please use the slider for each signal to indicate your opinion of preference. The grading scale is continuous about a zero point, “About the same”, from “A lot more” to “A lot less.” In evaluating the sound excerpts, please note that you should not necessarily give the lowest grade possible to your least preferable sample. However one sample in each experiment must be graded between -5 and +5 because the unprocessed reference signal is included as one of the samples to be graded.

When you are satisfied with your grading of all the samples you should click on the "save and continue" button at the bottom of the screen to move on to the next experiment. Once all six experiments are complete please click the "save and exit" button located in the same position.

Listener Information:

If you could please take the time to fill out the below form the information provided may prove extremely useful in the interpretation of the results.

Name of Participant:_____________________________

Male or Female? _______________________________

Course of Study:________________________________

Year of Study:__________________________________

Any additional comments or thoughts about the samples heard?

Thank you for your participation, your time has been greatly appreciated. If you wish to be contacted about the results of the audio evaluation please tick the box.

☐

8.11 - RECOMMENDATION ITU-R BS.1534-1:

Method for the subjective assessment of intermediate quality
level of coding systems
(Question ITU-R 220/10)
(2001-2003)
The ITU Radiocommunication Assembly,
considering

a) that Recommendations ITU-R BS.1116, ITU-R BS.1284, ITU-R BT.500, ITU-R BT.710 and ITU-R BT.811 as well as ITU-T Recommendations P.800, P.810 and P.830, have established methods for assessing subjective quality of audio, video and speech systems;

b) that new kinds of delivery services such as streaming audio on the Internet or solid state players, digital satellite services, digital short and medium wave systems or mobile multimedia applications may operate at intermediate audio quality;

c) that Recommendation ITU-R BS.1116 is intended for the assessment of small impairments and is not suitable for assessing systems with intermediate audio quality;

d) that Recommendation ITU-R BS.1284 gives no absolute scoring for the assessment of intermediate audio quality;

e) that ITU-T Recommendations P.800, P.810 and P.830 are focused on speech signals in a telephone environment and proved to be not sufficient for the evaluation of audio signals in a broadcasting environment;

f) that the use of standardized subjective test methods is important for the exchange, compatibility and correct evaluation of the test data;

g) that new multimedia services may require combined assessment of audio and video quality,

recommends

1 that the testing and evaluation procedures given in Annex 1 of this Recommendation be used for the subjective assessment of intermediate audio quality.

Annex 1

1 Introduction

This Recommendation describes a new method for the subjective assessment of intermediate audio quality. This method mirrors many aspects of Recommendation ITU-R BS.1116 and uses the same grading scale as is used for the evaluation of picture quality (i.e. Recommendation ITU-R BT.500).

The method, called "MUlti Stimulus test with Hidden Reference and Anchor (MUSHRA)", has been successfully tested. These tests have demonstrated that the MUSHRA method is suitable for evaluation of intermediate audio quality and gives accurate and reliable results, [EBU, 2000a; Soulodre and Lavoie, 1999; EBU, 2000b].

This Recommendation includes the following sections and Appendix:

Section 1: Introduction

Section 2: Scope, test motivation and purpose of new method

Section 3: Experimental design

Section 4: Selection of subjects

Section 5: Test method

Section 6: Attributes

Section 7: Test material

Section 8: Listening conditions

Section 9: Statistical analysis

Section 10: Test report and presentation of results

Appendix 1: Instructions to be given to subjects.

2 Scope, test motivation and purpose of new method

Subjective listening tests are recognized as still being the most reliable way of measuring the quality of audio systems. There are well described and proven methods for assessing audio quality at the top and the bottom quality range.

Recommendation ITU-R BS.1116 – Methods for the subjective assesment of small impairments in audio systems including multichannel sound systems, is used for the evaluation of high quality audio systems having small impairments. However, there are applications where lower quality audio is acceptable or unavoidable. Rapid developments in the use of the Internet for distribution and broadcast of audio material, where the data rate is limited, have led to a compromise in audio quality. Other applications that may contain intermediate audio quality are digital AM (i.e. digital radio mondiale (DRM), digital satellite broadcasting, commentary circuits in radio and TV, audio on demand services and audio on dial-up lines. The test method defined in Recommendation ITU R BS.1116 is not entirely suitable for evaluating these lower quality audio systems [Soulodre and Lavoie, 1999] because it is poor at discriminating between small differences in quality at the bottom of the scale.

Recommendation ITU-R BS.1284 gives only methods which are dedicated either to the high quality audio range or gives no absolute scoring of audio quality.

Other Recommendations, like ITU-T Recommendations P.800, P.810 or P.830, are focused on subjective assessment of speech signals in a telephone environment. The European Broadcasting Union (EBU) Project Group B/AIM has done experiments with typical audio material as used in a broadcasting environment using these ITU-T methods. None of these methods fulfils the requirement for an absolute scale, comparison with a reference signal and small confidence intervals with a reasonable number of subjects at the same time. Therefore the evaluation of audio signals in a broadcasting environment cannot be done properly by using one of these methods.

The new test method described in this Recommendation is intended to give a reliable and repeatable measure of systems having audio quality which would normally fall in the lower half of the impairment scale used by Recommendation ITU R BS.1116 [EBU, 2000a; Soulodre and Lavoie, 1999; EBU, 2000b]. In the MUSHRA test method, a high quality reference signal is used and the systems under test are expected to introduce significant impairments. If the systems under test can improve the subjective quality of a signal then other test methods should be used.

3 Experimental design

Many different kinds of research strategies are used in gathering reliable information in a domain of scientific interest. In the subjective assessment of impairments in audio systems, the most formal experimental methods shall be used. Subjective experiments are characterized firstly by actual control and manipulation of the experimental conditions, and secondly by collection and analysis of statistical data from listeners. Careful experimental design and planning is needed to ensure that uncontrolled factors which can cause ambiguity in test results are minimized. As an example, if the actual sequence of audio items were identical for all the subjects in a listening test, then one could not be sure whether the judgements made by the subjects were due to that sequence rather than to the different levels of impairments that were presented. Accordingly, the test conditions must be arranged in a way that reveals the effects of the independent factors, and only of these factors.

In situations where it can be expected that the potential impairments and other characteristics will be distributed homogeneously throughout the listening test, a true randomization can be applied to the presentation of the test conditions. Where non-homogeneity is expected this must be taken into account in the presentation of the test conditions. For example, where material to be assessed varies in level of difficulty, the order of presentation of stimuli must be distributed randomly, both within and between sessions.

Listening tests need to be designed so that subjects are not overloaded to the point of lessened accuracy of judgement. Except in cases where the relationship between sound and vision is important, it is preferred that the assessment of audio systems is carried out without accompanying pictures. A major consideration is the inclusion of appropriate control conditions. Typically, control conditions include the presentation of unimpaired audio materials, introduced in ways that are unpredictable to the subjects. It is the differences between judgement of these control stimuli and the potentially impaired ones that allows one to conclude that the grades are actual assessments of the impairments.

Some of these considerations will be described later. It should be understood that the topics of experimental design, experimental execution, and statistical analysis are complex, and that not all details can be given in a Recommendation such as this. It is recommended that professionals with expertise in experimental design and statistics should be consulted or brought in at the beginning of the planning for the listening test.

4 Selection of subjects

Data from listening tests assessing small impairments in audio systems, as in Recommendation ITU R BS.1116, should come from subjects who have experience in detecting these small impairments. The higher the quality reached by the systems to be tested, the more important it is to have experienced listeners.

4.1 Criteria for selecting subjects

Although the MUSHRA test method is not intended to be applied to small impairments, it is still recommended that experienced listeners should be used. These listeners should have experience in listening to sound in a critical way. Such listeners will give a more reliable result more quickly than non-experienced listeners. It is also important to note that most non-experienced listeners tend to become more sensitive to the various types of artefacts after frequent exposure.

There is sometimes a reason for introducing a rejection technique either before (pre-screening) or after (post screening) the real test. In some cases both types of rejections might be used. Here, rejection is a process where all judgements from a particular subject are omitted.

Any type of rejection technique, not carefully analysed and applied, may lead to a biased result. It is thus extremely important that, whenever elimination of data has been made, the test report clearly describes the criterion applied.

4.1.1 Pre-screening of subjects

The listening panel should be composed of experienced listeners, in other words, people who understand and have been properly trained in the described method of subjective quality evaluation. These listeners should:

- have experience in listening to sound in a critical way;
- have normal hearing (ISO Standard 389 should be used as a guideline).

The training procedure might be used as a tool for pre-screening.
The major argument for introducing a pre-screening technique is to increase the efficiency of the listening test. This must however be balanced against the risk of limiting the relevance of the result too much.

4.1.2 Post-screening of subjects

Post-screening methods can be roughly separated into at least two classes:

- one is based on the ability of the subject to make consistent repeated gradings;
- the other relies on inconsistencies of an individual grading compared with the mean result of all subjects for a given item.

It is recommended to look to the individual spread and to the deviation from the mean grading of all subjects.
The aim of this is to get a fair assessment of the quality of the test items.
If few subjects use either extreme end of the scale (excellent, bad) and the majority are concentrated at another point on the scale, these subjects could be recognized as outliers and might be rejected.

Due to the fact that "intermediate quality" is tested, a subject should be able to identify the coded version very easily and therefore find a grade which is in the range of the majority of the subjects. Subjects with grades at the upper end of the scale are likely to be less critical and subjects who have grades only at the lowest end of the scale are likely to be too critical. By rejecting these extreme subjects a more realistic quality assessment is expected.
The methods are primarily used to eliminate subjects who cannot make the appropriate discriminations. The application of a post-screening method may clarify the tendencies in a test result. However, bearing in mind the variability of subjects' sensitivities to different artefacts, caution should be exercised. By increasing the size of the listening panel, the effects of any individual subject's grades will be reduced and so the need to reject a subject's data is greatly diminished.

4.2 Size of listening panel

The adequate size for a listening panel can be determined if the variance of grades given by different subjects can be estimated and the required resolution of the experiment is known.
Where the conditions of a listening test are tightly controlled on both the technical and behavioural side, experience has shown that data from no more than 20 subjects are often sufficient for drawing appropriate conclusions from the test. If analysis can be carried out as

the test proceeds, then no further subjects need to be processed when an adequate level of statistical significance for drawing appropriate conclusions from the test has been reached.

If, for any reason, tight experimental control cannot be achieved, then larger numbers of subjects might be needed to attain the required resolution.

The size of a listening panel is not solely a consideration of the desired resolution. The result from the type of experiment dealt with in this Recommendation is, in principle, only valid for precisely that group of experienced listeners actually involved in the test. Thus, by increasing the size of the listening panel the result can be claimed to hold for a more general group of experienced listeners and may therefore sometimes be considered more convincing. The size of the listening panel may also need to be increased to allow for the probability that subjects vary in their sensitivity to different artefacts.

5 Test method

The MUSHRA test method uses the original unprocessed programme material with full bandwidth as the reference signal (which is also used as a hidden reference) as well as at least one hidden anchor. Additional well-defined anchors can be used as described in Section 5.1.

5.1 Description of test signals

The length of the sequences should typically not exceed 20 s to avoid fatiguing of listeners and to reduce the total duration of the listening test.

The set of processed signals consists of all the signals under test and at least one additional signal (anchor) being a low-pass filtered version of the unprocessed signal. The bandwidth of this additional signal should be 3.5 kHz. Depending on the context of the test, additional anchors can be

used optionally. Other types of anchors showing similar types of impairments as the systems under test can be used. These types of impairments can include:

- bandwidth limitation of 7 kHz or 10 kHz;
- reduced stereo image;
- additional noise;
- drop outs;
- packet losses;
- and others.

NOTE 1 – The bandwidths of the anchors correspond to the Recommendations for control circuits (3.5 kHz), used for supervision and coordination purpose in broadcasting, commentary circuits (7 kHz) and occasional circuits (10 kHz), according to ITU T Recommendations G.711, G.712, G.722 and J.21, respectively. The characteristic of the 3.5 kHz low-pass filter should be as follows:

f_c = 3.5 kHz

Maximum passband ripple = ±0.1 dB
Minimum attenuation at 4 kHz = 25 dB
Minimum attenuation at 4.5 kHz = 50 dB.
The additional anchors are intended to provide an indication of how the systems under test compare to well-known audio quality levels and should not be used for rescaling results between different tests.

5.2 **Training phase**

In order to get reliable results, it is mandatory to train the subjects in special training sessions in advance of the test. This training has been found to be important for obtaining reliable results. The training should at least expose the subject to the full range and nature of impairments and all test signals that will be experienced during the test. This may be achieved using several methods: a simple tape playback system or an interactive computer-controlled system. Instructions are given in Appendix 1.

5.3 **Presentation of stimuli**

MUSHRA is a double-blind multi-stimulus test method with hidden reference and hidden anchor(s), whereas Recommendation ITU-R BS.1116 uses a "double blind triple-stimulus with hidden reference" test method. The MUSHRA approach is felt to be more appropriate for evaluating medium and large impairments [Soulodre and Lavoie, 1999].
In a test involving small impairments, the difficult task for the subject is to detect any artefacts which might be present in the signal. In this situation a hidden reference signal is necessary in the test in order to allow the experimenter to evaluate the subject's ability to successfully detect these artefacts. Conversely, in a test with medium and large impairments, the subject has no difficulty in detecting the artefacts and therefore a hidden reference is not necessary for this purpose. Rather, the difficulty arises when the subject must grade the relative annoyances of the various artefacts. Here the subject must weigh his preference for one type of artefact versus some other type of artefact.

The use of a high quality reference introduces an interesting problem. Since the new methodology is to be used for evaluating medium and large impairments, the perceptual difference from the reference signal to the test items is expected to be relatively large. Conversely, the perceptual differences between the test items belonging to different systems may be quite small. As a result, if a multi-trial test method (such as is used in Recommendation ITU-R BS.1116) is used, it may be very difficult for subjects to accurately discriminate between the various impaired signals. For example, in a direct paired comparison test subjects might agree that System A is better than System B. However, in a situation where each system is only compared with the reference signal (i.e. System A and System B are not directly compared to each other), the differences between the two systems may be lost.

To overcome this difficulty, in the MUSHRA test method, the subject can switch at will between the reference signal and any of the systems under test, typically using a computer-controlled replay system, although other mechanisms using multiple CD or tape machines can be used. The subject is presented with a sequence of trials. In each trial the subject is presented with the reference version as well as all versions of the test signal processed by the systems under test. For example, if a test contains 8 audio systems, then the subject is allowed to switch instantly among the 11 signals (1 reference + 8 impaired + 1 hidden reference + 1 hidden anchor).

Because the subject can directly compare the impaired signals, this method provides the benefits of a full paired comparison test in that the subject can more easily detect differences between the impaired signals and grade them accordingly. This feature permits a high degree of resolution in the grades given to the systems. It is important to note however, that subjects will derive their grade for a given system by comparing that system to the reference signal, as well as to the other signals in each trial.

It is recommended that no more than 15 signals (e.g. 12 systems under test, 1 known reference, 1 hidden anchor, and 1 hidden reference) should be included in any trial.

In a Recommendation ITU-R BS.1116 test, subjects tend to approach a given trial by starting with a detection process, followed by a grading process. The experience from conducting tests according to the MUSHRA method shows, that subjects tend to begin a session with a rough estimation of the quality. This is followed by a sorting or ranking process. After that the subject performs the grading process. Since the ranking is done in a direct fashion, the results for intermediate audio quality are likely to be more consistent and reliable than if the Recommendation ITU-R BS.1116 method had been used.

5.4 **Grading process**

The subjects are required to score the stimuli according to the continuous quality scale (CQS). The CQS consists of identical graphical scales (typically 10 cm long or more) which are divided into five equal intervals with the adjectives as given in Fig. 1 from top to bottom.

This scale is also used for evaluation of picture quality (Recommendation ITU-R BT.500 – Methodology for the subjective assessment of the quality at television pictures).

The listener records his/her assessment of the quality in a suitable form, for example, with the use of sliders on an electronic display (see Fig. 2), or using a pen and paper scale. Using a set up similar to that shown in Fig. 2 the subject should be constrained, to be able only to adjust the score assigned to the item he or she is currently listening to. Some guidance about interface design can be found in Appendix 2 to Annex 1. The subject is asked to assess the quality of all stimuli, according to the five-interval CQS.

Compared to Recommendation ITU-R BS.1116, the MUSHRA method has the advantage of displaying all stimuli at the same time so that the subject is able to carry out any comparison between them directly. The results become more consistent, leading to smaller confidence intervals. The time taken to perform the test using the MUSHRA method can be significantly less than when using the Recommendation ITU-R BS.1116 method.

5.5 **Recording of test sessions**

In the event that something anomalous is observed when processing assigned scores, it is very useful to have a record of the events that produced the scores. A relatively simple way of achieving this is to make video and audio recordings of the whole test. In the case where an anomalous grade is found in a set of results, the tape recording can be inspected to try to establish whether the reason was human error or equipment malfunction.

6 **Attributes**

Listed below are attributes specific to monophonic, stereophonic and multichannel evaluations. It is preferred that the attribute "basic audio quality" be evaluated in each case. Experimenters may choose to define and evaluate other attributes.

Only one attribute should be graded during one trial. When subjects are asked to assess more than one attribute in each trial they can become overburdened or confused, or both, by trying to answer multiple questions about a given stimulus. This might produce unreliable gradings for all the questions.

6.1 Monophonic system

Basic audio quality: This single, global attribute is used to judge any and all detected differences between the reference and the object.

6.2 Stereophonic system

Basic audio quality: This single, global attribute is used to judge any and all detected differences between the reference and the object. The following additional attribute may be of interest:

Stereophonic image quality: This attribute is related to differences between the reference and the object in terms of sound image locations and sensations of depth and reality of the audio event. Although some studies have shown that stereophonic image quality can be impaired, sufficient research has not yet been done to indicate whether a separate rating for stereophonic image quality as distinct from basic audio quality is warranted.

NOTE 1 – Up to 1993, most small impairment subjective evaluation studies of stereophonic systems have used the attribute basic audio quality exclusively. Thus the attribute stereophonic image quality was either implicitly or explicitly included within basic audio quality as a global attribute in those studies.

6.3 Multichannel system

Basic audio quality: This single, global attribute is used to judge any and all detected differences between the reference and the object.

The following additional attributes may be of interest:

Front image quality: This attribute is related to the localization of the frontal sound sources. It includes stereophonic image quality and losses of definition.

Impression of surround quality: This attribute is related to spatial impression, ambience, or special directional surround effects.

7 Test material

Critical material which represents the typical broadcast programme for the desired application shall be used in order to reveal differences among systems under test. Critical material is that which stresses the systems under test. There is no universally suitable programme material that can be used to assess all systems under all conditions. Accordingly, critical programme material must be sought explicitly for each system to be tested in each experiment. The search for suitable material is usually time-consuming; however, unless truly critical material is found for each system, experiments will fail to reveal differences among systems and will be inconclusive.

It must be empirically and statistically shown that any failure to find differences among systems is not due to experimental insensitivity which may be caused by poor choices of audio material, or any other weak aspects of the experiment. Otherwise this “null” finding cannot be accepted as valid.

In the search for critical material, any stimulus that can be considered as potential broadcast material shall be allowed. Synthetic signals deliberately designed to break a specific system should not be included. The artistic or intellectual content of a programme sequence should be neither so attractive nor so disagreeable or wearisome that the subject is distracted from focusing on the detection of impairments. The expected frequency of occurrence of each type of programme material in actual broadcasts should be taken into account. However, it should be understood that the nature of broadcast material might change in time with future changes in musical styles and preferences.

When selecting the programme material, it is important that the attributes which are to be assessed are precisely defined. The responsibility of selecting material shall be delegated to a group of skilled subjects with a basic knowledge of the impairments to be expected. Their starting point shall be based on a very broad range of material. The range can be extended by dedicated recordings.

For the purpose of preparing for the formal subjective test, the loudness of each excerpt needs to be adjusted subjectively by the group of skilled subjects prior to recording it on the test media. This will allow subsequent use of the test media at a fixed gain setting for all programme items within a test trial.

For all test sequences the group of skilled subjects shall convene and come to a consensus on the relative sound levels of the individual test excerpts. In addition, the experts should come to a consensus on the absolute reproduced sound pressure level for the sequence as a whole relative to the alignment level.

A tone burst (for example 1 kHz, 300 ms, −18 dBFS) at alignment signal level may be included at the head of each recording to enable its output alignment level to be adjusted to the input alignment level required by the reproduction channel, according to EBU Recommendation R68 (see Recommendation ITU-R BS.1116, § 8.4.1). The tone burst is only for alignment purposes: it should not be replayed during the test. The sound programme signal should be controlled so that the amplitudes of the peaks only rarely exceed the peak amplitude of the permitted maximum signal defined in Recommendation ITU-R BS.645 (a sine wave 9 dB above the alignment level).

The feasible number of excerpts to include in a test varies: it shall be equal for each system under test. A reasonable estimate is 1.5 times the number of systems under test, subject to a minimum value of 5 excerpts. Audio excerpts will be typically 10 s to 20 s long. Due to the complexity of the task, the systems under test should be available. A successful selection can only be achieved if an appropriate time schedule is defined.

The performance of a multichannel system under the conditions of two-channel playback shall be tested using a reference down-mix. Although the use of a fixed down-mix may be considered to be restricting in some circumstances, it is undoubtedly the most sensible option for use by broadcasters in the long run. The equations for the reference down-mix (see Recommendation ITU-R BS.775) are:

The pre-selection of suitable test excerpts for the critical evaluation of the performance of reference two-channel down-mix should be based on the reproduction of two-channel down-mixed programme material.

8 Listening conditions

Methods for the subjective assessment of small impairments in audio systems including multichannel sound systems are defined in Recommendation ITU-R BS.1116. For evaluating audio systems having intermediate quality the listening conditions outlined in Sections 7 and 8 of Recommendation ITU-R BS.1116 shall be used.

Either headphones or loudspeakers may be used in the test. The use of both within one test session is not permitted: all subjects must use the same type of transducer.

For a measuring signal with an r.m.s. voltage equal to the "alignment signal level" (0 dBu0s according to Recommendation ITU-R BS.645; –18 dB below the clipping level of a digital tape recording, according to EBU Recommendation R68) fed in turn to the input of each reproduction channel (i.e. a power amplifier and its associated loudspeaker), the gain of the amplifier shall be adjusted to give the reference sound pressure level (IEC/A-weighted, slow):

$$L_{ref} = 85 - 10 \log n \pm 0.25 \quad \text{dBA}$$

where n is the number of reproduction channels in the total setup.

Individual adjustment of listening level by a subject is allowed within a session and should be limited in the range of ±4 dB relative to the reference level defined in Recommendation ITU R BS.1116. The balance between the test items in one test should be provided by the selection panel in such a way that the subjects would normally not need to perform individual adjustments for each item.

Level adjustments inside one item should be not allowed.

9 Statistical analysis

The assessments for each test condition are converted linearly from measurements of length on the score sheet to normalized scores in the range 0 to 100, where 0 corresponds to the bottom of the scale (bad quality). Then, the absolute scores are calculated as follows.
The calculation of the averages of normalized scores of all listeners remaining after post-screening will result in the mean subjective scores.
The first step of the analysis of the results is the calculation of the mean score, for each of the presentations:

(1)

where:

ui : score of observer i for a given test condition j and audio sequence k

N : number of observers.

Similarly, overall mean scores, and could be calculated for each test condition and each test sequence.
When presenting the results of a test all mean scores should have an associated confidence interval which is derived from the standard deviation and size of each sample.
It is proposed to use the 95% confidence interval which is given by:

where:

(2)

and t0.05 is the t value for a significance level of 95%.
The standard deviation for each presentation, Sjk, is given by:

(3)

With a probability of 95%, the absolute value of the difference between the experimental mean score and the true mean score (for a very high number of observers) is smaller than the 95% confidence interval, on condition that the distribution of the individual scores meets certain requirements.

Similarly, a standard deviation Sj could be calculated for each test condition. It is noted however that this standard deviation will, in cases where a small number of test sequences are used, be influenced more by differences between the test sequences used than by variations between the assessors participating in the assessment.
Experience has shown that the scores obtained for different test sequences are dependent on the criticality of the test material used. A more complete understanding of system performance can be obtained by presenting results for different test sequences separately, rather than only as aggregated averages across all the test sequences used in the assessment.

10 Test report and presentation of results

10.1 General

The presentation of the results should be made in a user friendly way such that any reader, either a naïve one or an expert, is able to get the relevant information. Initially any reader wants to see the overall experimental outcome, preferably in a graphical form. Such a presentation may be supported by more detailed quantitative information, although full detailed numerical analyses should be in appendices.

10.2 Contents of the test report

The test report should convey, as clearly as possible, the rationale for the study, the methods used and conclusions drawn. Sufficient detail should be presented so that a knowledgeable person could, in principle, replicate the study in order to check empirically on the outcome. However, it is not necessary that the report contains all individual results. An informed reader ought to be able to understand and develop a critique for the major details of the test, such as the underlying reasons for the study, the experimental design methods and execution, and the analyses and conclusions.

Special attention should be given to the following:

- a graphical presentation of the results;
- the specification and selection of subjects (see Note 1);
- the specification and selection of test material;
- general information about the system used to process the test material;
- details of the test configuration;
- the physical details of the listening environment and equipment, including the room dimensions and acoustic characteristics, the transducer types and placements, electrical equipment specification (see Note 2);
- the experimental design, training, instructions, experimental sequences, test procedures, data generation;
- the processing of data, including the details of descriptive and analytic inferential statistics;
- the detailed basis of all the conclusions that are drawn.

NOTE 1 – There is evidence that variations in the skill level of listening panels can influence the results of listening assessments. To facilitate further study of this factor experimenters are requested to report as much of the characteristics of their listening panels as possible. Relevant factors might include the age and gender composition of the panel.

NOTE 2 – Because there is some evidence that listening conditions, for example loudspeaker versus headphone reproduction, may influence the results of subjective assessments, experimenters are requested to explicitly report the listening conditions, and the type of reproduction equipment used in the experiments. If a combined statistical analysis of different transducer types is intended, it has to be checked whether such a combination of the results is possible (for example using ANOVA).

10.3 Presentation of the results

For each test parameter, the mean and 95% confidence interval of the statistical distribution of the assessment grades must be given.

The results must be given together with the following information:

− description of the test materials;

− number of assessors;

− the overall mean score for all test items used in the experiment;

− only the mean scores and 95% confidence interval after post-screening of the observers, i.e. after eliminating those results according to the procedure given in Section 4.1.2 (post screening).

Additionally, the results could also be presented in appropriate forms like histograms, medians or other.

10.4 Absolute grades

A presentation of the absolute mean grades, for the systems under test, the hidden reference, and anchor gives a good overview of the result. One should however keep in mind that this does not provide any information of the detailed statistical analysis. Consequently the observations are not independent and statistical analysis of these absolute grades will not lead to meaningful information and should not be done.

10.4 Significance level and confidence interval

The test report should provide the reader with information about the inherently statistical nature of all subjective data. Significance levels should be stated, as well as other details about statistical methods and outcomes, which will facilitate the understanding by the reader. Such details might include confidence intervals or error bars in graphs.

There is of course no "correct" significance level. However, the value 0.05 is traditionally chosen. It is, in principle, possible to use either a one tailed or a two tailed test depending on the hypothesis being tested.

References

EBU [2000a] MUSHRA – Method for Subjective Listening Tests of Intermediate Audio Quality. Draft EBU Recommendation, B/AIM 022 (Rev.8)/BMC 607rev, January.

EBU [2000b] EBU Report on the subjective listening tests of some commercial internet audio codecs. Document BPN 029, June.
SOULODRE, G. A. and LAVOIE, M. C. [September 1999] Subjective evaluation of large and small impairments in audio codecs, AES 17th International Conference, Florence, pp. 329-336.

Appendix 1

to Annex 1

Instructions to be given to subjects

The following is an example of the type of instructions that should be given to or read to the subjects in order to instruct them on how to perform the test.

1 Familiarization or training phase

The first step in the listening tests is to become familiar with the testing process. This phase is called a training phase and it precedes the formal evaluation phase.

The purpose of the training phase is to allow you, as an evaluator, to achieve two objectives as follows:

– Part A: to become familiar with all the sound excerpts under test and their quality level ranges; and

– Part B: to learn how to use the test equipment and the grading scale.

In Part A of the training phase you will be able to listen to all sound excerpts that have been selected for the tests in order to illustrate the whole range of possible qualities. The sound items, which you will listen to, will be more or less critical depending on the bit rate and other "conditions" used. Figure 3 shows the user interface. You may click on different buttons to listen to different sound excerpts including the reference excerpts. In this way you can learn to appreciate a range of different levels of quality for different programme items. The excerpts are grouped on the basis of common conditions. Three such groups are identified in this case. Each group includes 4 processed signals.

In Part B of the training phase you will learn to use the available playback and scoring equipment that will be used to evaluate the quality of the sound excerpts.

During the training phase you should be able to learn how you, as an individual, interpret the audible impairments in terms of the grading scale. You should not discuss your personal interpretation of the scale with the other subjects at any time during the training phase. However, you are encouraged to explain artefacts to other subjects.

No grades given during the training phase will be taken into account in the true tests.

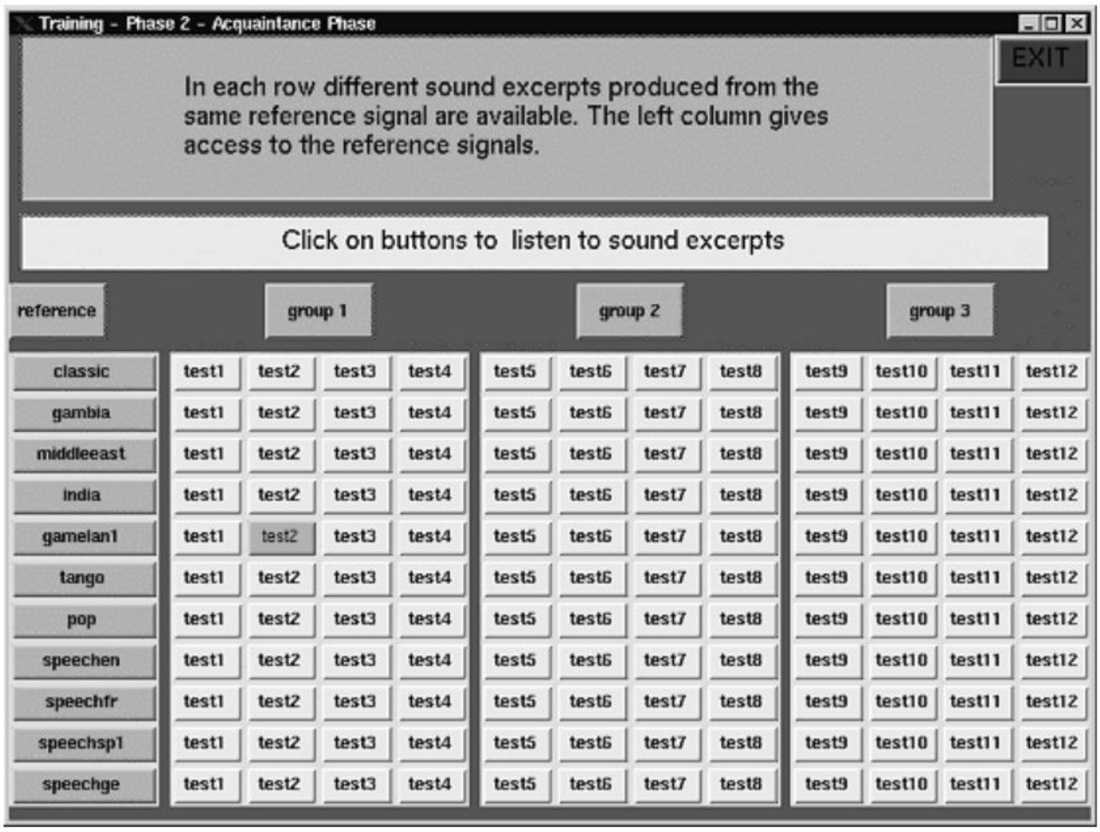
2 Blind grading phase

The purpose of the blind grading phase is to invite you to assign your grades using the quality scale. Your grades should reflect your subjective judgement of the quality level for each of the sound excerpts presented to you. Each trial will contain 11 signals to be graded. Each of the items is approximately 10 to 20 s long. You should listen to the reference and all the test conditions by clicking on the respective buttons. You may listen to the signals in any order, any number of times.

Use the slider for each signal to indicate your opinion of its quality. When you are satisfied with your grading of all signals you should click on the "register scores" button at the bottom of the screen.

FIGURE 3

Picture showing an example of a user interface for Part A of the training phase



1534-03

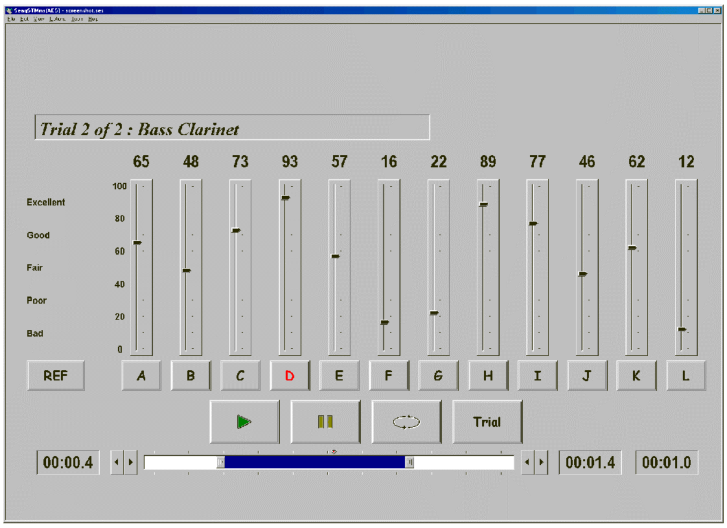
You will use the quality scale as given in Fig. 1 when assigning your grades.

The grading scale is continuous from "excellent" to "bad". A grade of 0 corresponds to the bottom of the "bad" category, while a grade of 100 corresponds to the top of the "excellent" category.

In evaluating the sound excerpts, please note that you should not necessarily give a grade in the "bad" category to the sound excerpt with the lowest quality in the test. However one or more excerpts must be given a grade of 100 because the unprocessed reference signal is included as one of the excerpts to be graded

FIGURE 4

An example of the user interface used in the blind grading phase



1534-04

Appendix 2

to Annex 1

Guidance notes on user interface design

The following suggestions are made for those who might be considering:

a) producing systems for performing subjective tests according to the MUSHRA method,

b) performing such tests.

These suggestions are intended to increase the reliability of the results of tests and to facilitate the analysis of any irregularities that might be found during the processing of test scores.

The design of the user interface should be such that the chance of a subject assigning a score which does not accord their true intent is minimized. To this end, steps should be taken to ensure that it is clear from the user interface to which of the processed versions of a test item the subject is listening at a given time. This can be aided by careful choice of colours and brightness of on-screen indicators (clickable buttons, for example) to avoid potential difficulties should a subject not be sensitive to some colours.

It should also be ensured that the subject is only able to adjust the score assigned to the item currently being listened to. It has been observed that some subjects listen to two processed versions of an item, in succession, in order to assign a score to the first, not the last, that they hear. In this circumstance, it is possible that a mistake might be made (especially when a large number of on screen controls are presented) and the score might be assigned to a signal other than the intended one. To try to reduce this possibility, it is suggested that the only control that is enabled at any one time is the one related to the signal currently being heard. Controls to assign scores to other signals, not currently being heard, should be disabled.

8.12 - Average RMS of Samples for Required THD% Output:

HT = 50V **Monteith/Flowers**

0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	0.000	-595.735	0.000	0.000	0.0000	0.000
2.000	4.000	-63.694	5.657	21.417	0.0007	0.014
6.000	11.500	-54.521	16.263	61.574	0.0019	0.041
8.000	15.500	-51.928	21.920	82.991	0.0025	0.055
10.000	19.500	-49.934	27.577	104.408	0.0032	0.068
16.000	31.800	-45.686	44.972	170.265	0.0052	0.111

HT = 50V **Valve Stage**

0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	0.000	-595.735	0.000	0.000	0.0000	0.000
2.000	2.200	-68.887	3.111	11.779	0.0004	0.104
6.000	7.300	-58.469	10.324	39.086	0.0012	0.204
8.000	11.000	-54.907	15.556	58.897	0.0018	0.286
10.000	16.700	-51.281	23.617	89.416	0.0027	0.427
16.000	66.500	-39.279	94.045	356.058	0.0109	1.629

Figure 8.8 – Conversion of Sample Level for HT = 50V.

HT = 100V **Monteith/Flowers**

0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	48.000	-42.110	67.882	257.004	0.0078	0.421
2.000	126.000	-33.728	178.191	674.635	0.0206	1.096
6.000	145.000	-32.508	205.061	776.366	0.0237	1.214
8.000	156.000	-31.873	220.617	835.263	0.0255	1.285
10.000	191.000	-30.114	270.115	1022.661	0.0312	1.533
16.000	279.000	-26.823	394.566	1493.835	0.0456	2.067

HT = 100V **Valve Stage**

0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
1.000	5.030	-61.704	7.113	26.932	0.0008	0.174
2.000	8.000	-57.673	11.314	42.834	0.0013	0.278
6.000	17.200	-51.024	24.324	92.093	0.0028	0.595
8.000	23.200	-48.425	32.810	124.219	0.0038	0.815
10.000	31.600	-45.741	44.689	169.194	0.0052	1.305
16.000	88.500	-36.796	125.158	473.851	0.0145	3.392

Figure 8.9 – Conversion of Sample Level for HT = 100V.

HT = 150V Monteith/Flowers

0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	179.000	-30.678	253.144	958.410	0.029	1.658
2.000	224.000	-28.730	316.784	1199.352	0.037	2.051
6.000	359.000	-24.633	507.703	1922.175	0.059	3.113
8.000	376.000	-24.231	531.744	2013.197	0.061	3.176
10.000	390.000	-23.914	551.543	2088.157	0.064	3.210
16.000	433.000	-23.005	612.354	2318.390	0.071	3.299

HT = 150V Valve Stage

0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	7.000	-58.833	9.899	37.480	0.0011	0.282
2.000	15.100	-52.155	21.355	80.849	0.0025	0.602
6.000	26.400	-47.303	37.335	141.352	0.0043	1.052
8.000	34.500	-44.979	48.790	184.722	0.0056	1.398
10.000	46.200	-42.442	65.337	247.366	0.0075	1.932
16.000	133.500	-33.225	188.798	714.792	0.0218	5.980

Figure 8.10 - Conversion of Sample Level for HT = 150V.

HT = 200V Monteith/Flowers

0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	306.000	-26.021	432.749	1638.400	0.050	2.970
1.000	315.000	-25.769	445.477	1686.588	0.051	2.965
2.000	380.000	-24.139	537.401	2034.614	0.062	3.520
4.000	466.500	-22.358	659.731	2497.757	0.076	4.172
6.000	493.000	-21.878	697.207	2639.644	0.081	4.297
8.000	513.000	-21.533	725.492	2746.729	0.084	4.387
10.000	531.000	-21.233	750.947	2843.106	0.087	4.429
12.000	548.000	-20.959	774.989	2934.128	0.090	4.467
14.000	568.000	-20.648	803.273	3041.213	0.093	4.497
16.000	587.000	-20.362	830.143	3142.944	0.096	4.518

HT = 200V Valve Stage

0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	10.000	-55.735	14.142	53.542	0.0016	0.437
1.000	18.500	-50.392	26.163	99.054	0.0030	0.824
2.000	22.700	-48.615	32.103	121.541	0.0037	0.973
4.000	28.800	-46.547	40.729	154.202	0.0047	1.226
6.000	36.300	-44.537	51.336	194.359	0.0059	1.572
8.000	46.700	-42.349	66.044	250.043	0.0076	2.066
10.000	62.800	-39.776	88.813	336.247	0.0103	2.853
12.000	89.700	-36.679	126.855	480.276	0.0147	4.252
14.000	134.400	-33.167	190.070	719.611	0.0220	6.530
16.000	199.600	-29.732	282.277	1068.708	0.0326	9.840

Figure 8.11 - Conversion of Sample Level for HT = 200V.

HT = 250V Monteith/Flowers
0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	404.000	-23.607	571.342	2163.116	0.066	3.806
2.000	519.000	-21.432	733.977	2778.855	0.085	4.756
6.000	620.000	-19.887	876.812	3319.634	0.101	5.402
8.000	641.000	-19.598	906.511	3432.073	0.105	5.466
10.000	663.000	-19.305	937.624	3549.867	0.108	5.522
16.000	733.000	-18.433	1036.619	3924.664	0.120	5.641

HT = 250V Valve Stage
0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	11.000	-54.907	15.556	58.897	0.0018	0.487
2.000	28.100	-46.761	39.739	150.454	0.0046	1.247
6.000	43.500	-42.965	61.518	232.910	0.0071	1.960
8.000	56.000	-40.771	79.196	299.838	0.0092	2.581
10.000	76.000	-38.119	107.480	406.923	0.0124	3.641
16.000	260.000	-27.436	367.696	1392.105	0.0425	13.481

Figure 8.12 - Conversion of Sample Level for HT = 250V.

HT = 300V Monteith/Flowers
0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	487.000	-21.984	688.722	2607.519	0.080	4.595
2.000	647.000	-19.517	914.996	3464.199	0.106	5.958
6.000	743.000	-18.315	1050.761	3978.207	0.121	6.566
8.000	771.000	-17.994	1090.359	4128.125	0.126	6.661
10.000	798.000	-17.695	1128.542	4272.690	0.130	6.747
16.000	884.000	-16.806	1250.165	4733.156	0.144	6.920

HT = 300V Valve Stage
0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	19.000	-50.160	26.870	0.003	0.0031	0.874
2.000	34.200	-45.055	48.366	0.006	0.0056	1.568
6.000	51.100	-41.567	72.266	0.008	0.0083	2.347
8.000	65.100	-39.463	92.065	0.011	0.0106	3.067
10.000	88.000	-36.845	124.451	0.014	0.0144	4.338
16.000	321.000	-25.605	453.963	0.052	0.0525	17.178

Figure 8.13 - Conversion of Sample Level for HT = 300V.

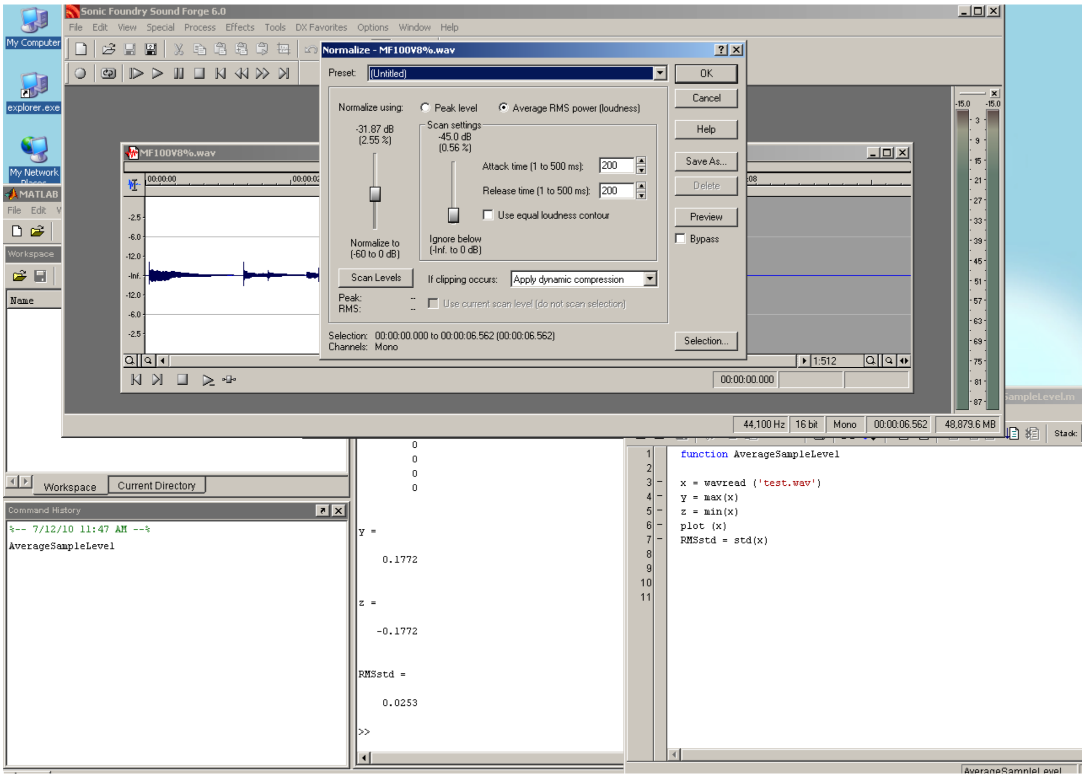


Figure 8.14 – Conversion of 8%THD M/F sample at 100V HT.

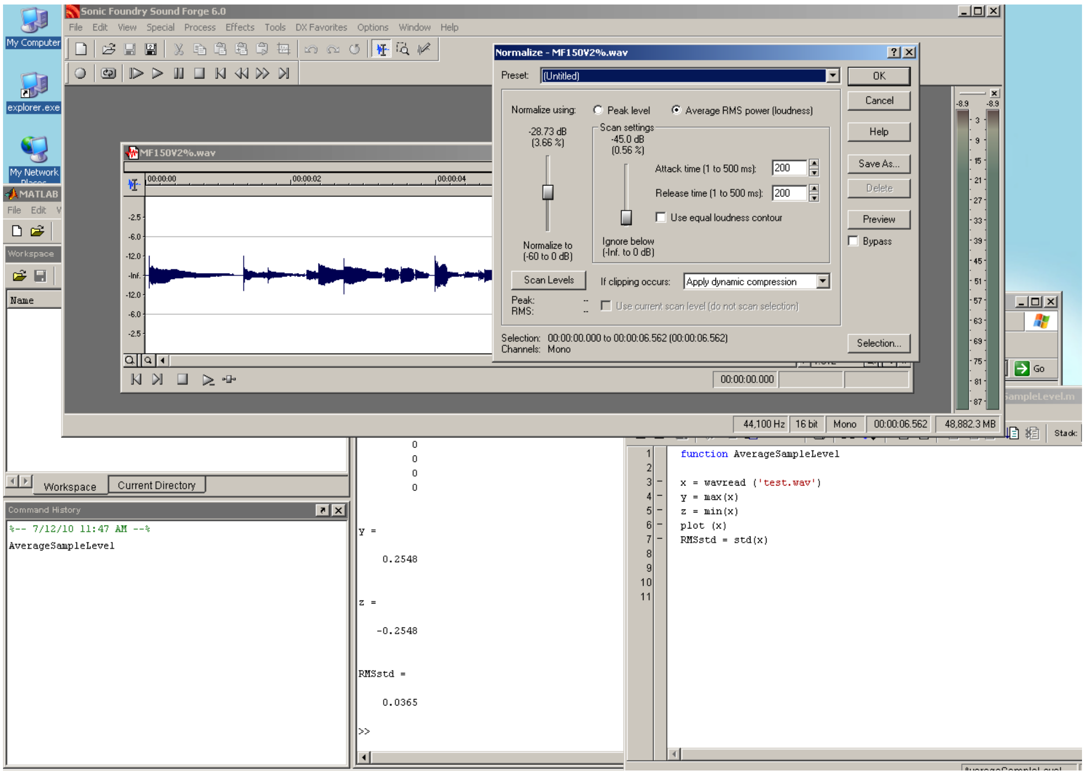


Figure 8.15 – Conversion of 2%THD M/F sample at 150V HT.

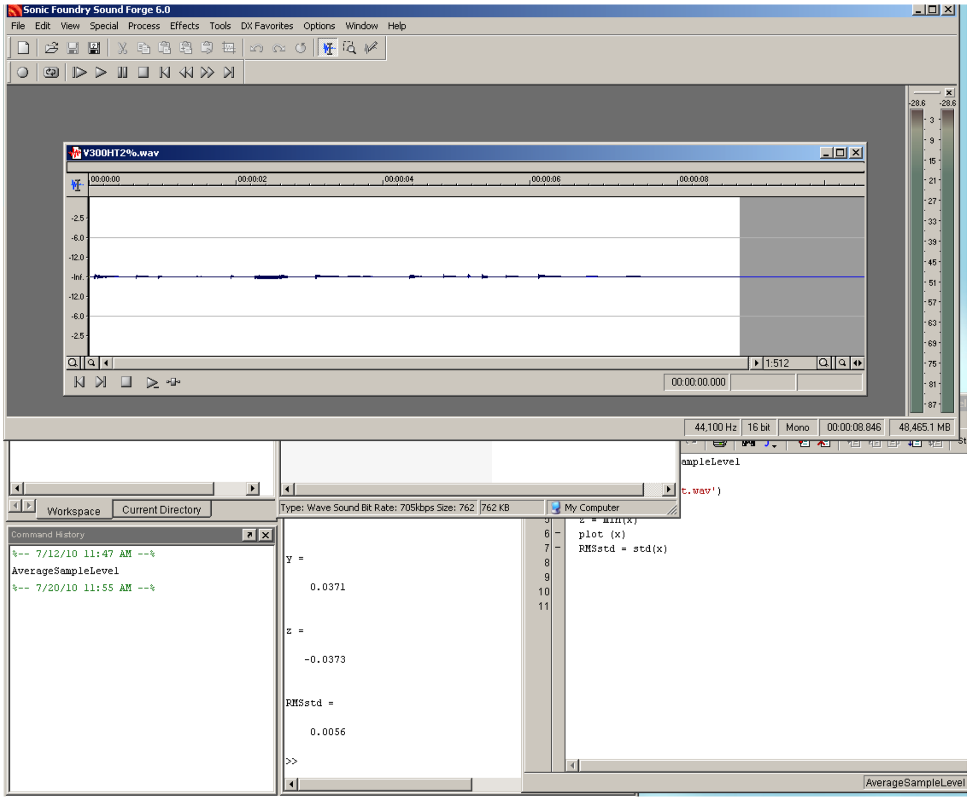


Figure 8.16 – Conversion of 2%THD valve sample at 300V HT.

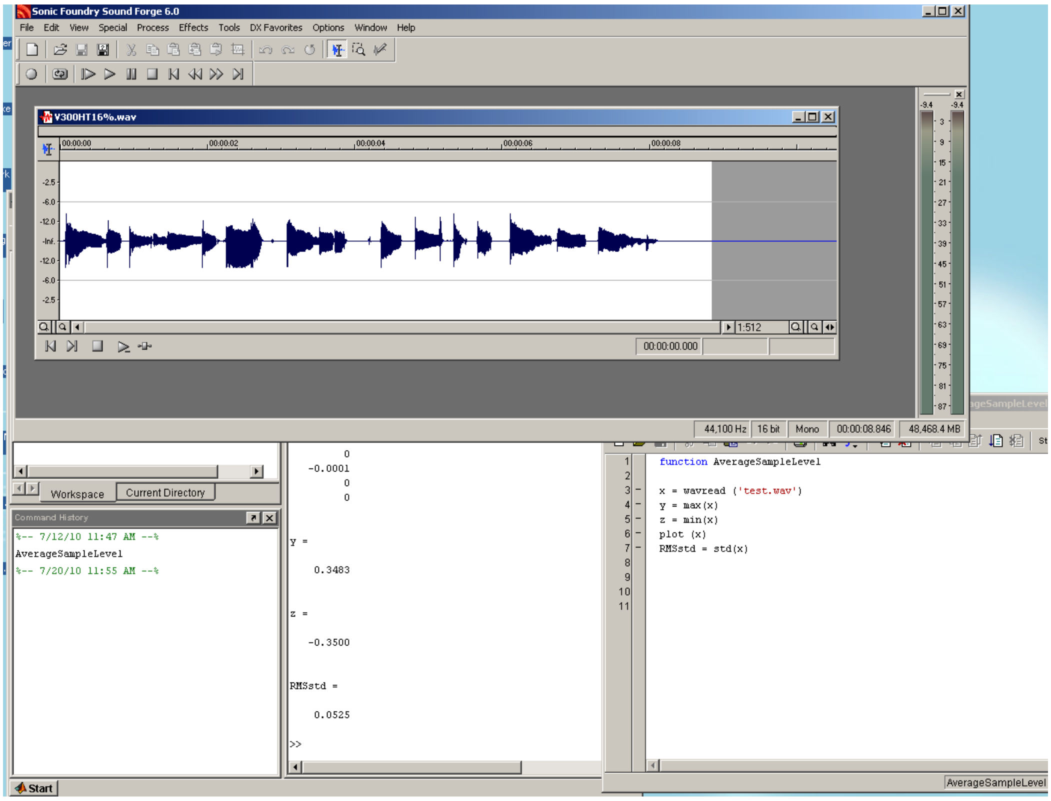
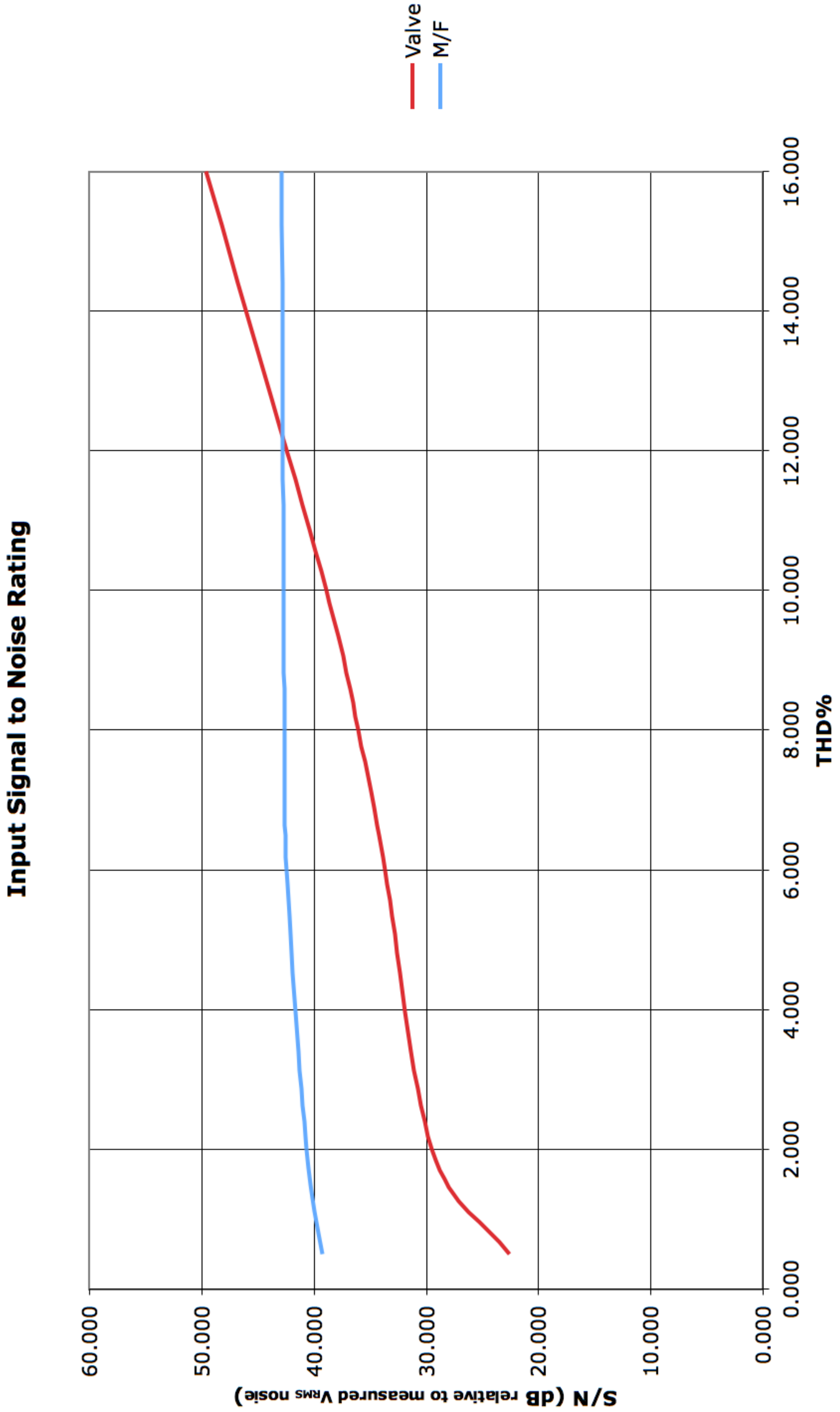


Figure 8.17 – Conversion of 16%THD valve sample at 300V HT.

8.13 - Output 003/Input to Preamplifier Signal/Noise:



8.14 - Paper Submission to the AES:

CAN TRANSISTORS SOUND BETTER THAN VALVES?

M. J. K. Aitchison – Studying MSc by Research. Steve Fenton – Supervising Tutor
University of Huddersfield, Queensgate, Huddersfield HD1 3DH, UK

ABSTRACT

This paper describes an objective and subjective comparison between a referenced high-voltage solid-state preamplifier with acclaims of large signal capabilities comparable to a valve amplifier, and an ECC83 based preamplifier topology. By analyzing the interaction of individual harmonic amplitudes throughout the amplifiers overload regions it is shown that there is little correlation between the two systems signal outputs. The paper describes the properties of the valve sound as having a dominant second order harmonic with an array of higher order harmonics producing the popular warmth of distortion often used for guitar amplification. The resulting dominance of 2nd and 4th harmonic components in the solid state system demonstrates the objective similarities, however differences are shown by the presence of prevalent higher order harmonics in contrast to the attenuated higher harmonics of the valve stage. Subjective measurements are made through a controlled audio evaluation comparing the two systems at varying matched levels of output THD%. The conclusion is drawn that with 95% certainty this transistor pre-amplifier does not sound like the ECC83 preamplifier circuit.

Keywords – Audio, amplification, valves, thermionic, tubes, guitar, harmonics.

1 – The Valve Sound

Monteith and Flowers [1] have designed "a low-noise microphone preamplifier transistor circuit which has the same large signal capability as tube designs." Fig. 1. They specifically mention that "this design exhibits the desirable overload characteristics of tubes" in accordance with the investigations of Hamm [2]. Hamm ultimately denotes that tube amplifiers react differently to transistor amplifiers in their regions of overload, particularly in the output signals interchanging dominance of odd and even harmonics when pushed up to 12dB into overload.

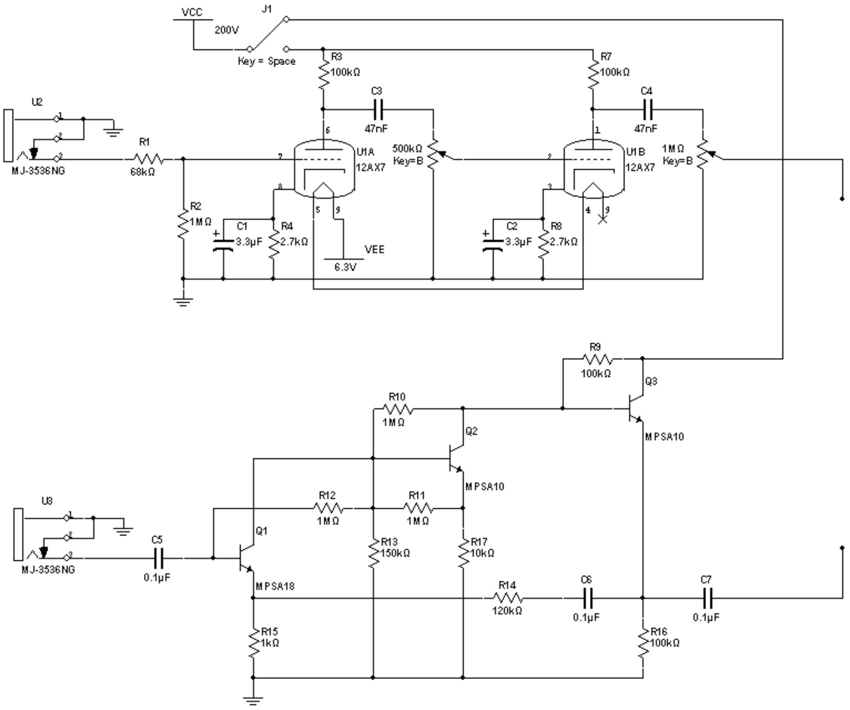
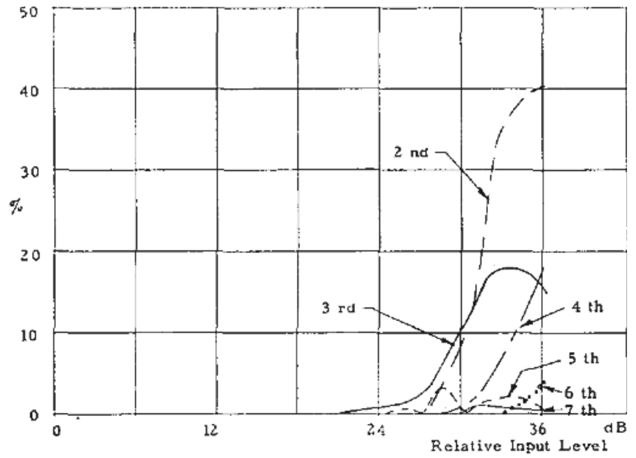


Figure 1 – Circuit diagrams for the DUT. [1]

The pretension of the Montieth/Flowers paper demonstrates the harmonic content of their systems performance approaching and passing through the point of output clipping showing the dominance of a 2nd order harmonic. It states that the circuit "produces harmonic distortion components which are comparable to, and perhaps more pleasing than, tube preamplifiers.'" [1]

Hamms developed measurement technique [2] using Fourier analysis to study the percentages of ensuing harmonics in relation to the fundamental showed conclusively, Fig. 2, that triode preamplifiers "outstanding characteristic" was a particularly dominant 2nd harmonic in tandem with an initially dominant 3rd harmonic and an increasing 4th harmonic further into overload.

A comparison with the resulting plot of the Monteith/Flowers circuit, Fig. 3, shows that although there are some "harmonic distortion components which are comparable to" [1] Hamms plots, there are also some apparent differences; An ultimately more dominant 3rd harmonic increasing over 20% of the fundamental approaching maximum overload; A diminishing 4th harmonic falling to less than 5% of the fundamental as opposed to Hamms steadily increasing one; And a prominent 5th harmonic peaking at approximately 11% of the fundamental in contrast to Hamms where the 5th, 6th and 7th harmonics all remain under the 5% line throughout clipping.



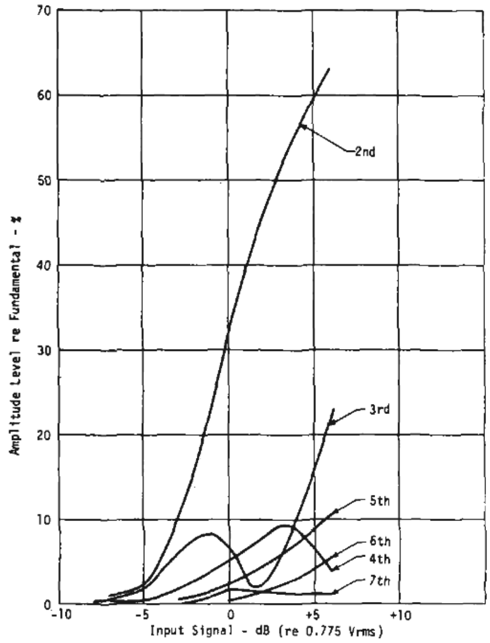


Figure 2. **Figure 3.**

Figure 2. Distortion components for two-stage triode amplifier. [2]

Figure 3. Distortion components as a function of input level for high-voltage preamplifier. [1]

Hamms conclusion that "inaudible harmonics in the early overload condition might very well be causing the difference in sound coloration between tubes and transistors" [2] would suggest that these differences in harmonic content could well belie the fact that the Monteith/Flowers circuit can indeed sound better than a tube amplifier.

The other main characteristic identified by Montieth/Flowers as "tubelike" is an output waveform that displays asymmetrical clipping. This too was documented by Hamm, as is demonstrated by comparing the oscilloscope shots within each paper. It is also well documented [4] that using Fourier analysis on complex signals can demonstrate that any waveform with a strong presence of 2nd harmonic will result in an asymmetric output, and as Mintz [5] implies, "a particular 'sound' may be incurred or avoided at the designer's pleasure no matter what active devices he uses.'" [5]

It is the authors' belief that no conclusions should be made as to whether an amplifier system sounds "tubelike" or not by noting an asymmetrical waveform on an oscilloscope.

2 – GUITAR AMPLIFICATION VERSUS HIGH FIDELITY AMPLIFICATION

The aforementioned papers were fundamentally focused towards high fidelity reproduction of sounds where accurate reproduction of signals is of utmost importance. This paper however is concentrated particularly on the use of valves in guitar amplification. Bussey and Haiglers paper [6] identifies the crucial difference between audio reproduction and guitar amplification in that, ”In this application the amplifier becomes part of the musical instrument, and is frequently used to radically alter the signal from the guitar.” [6]

Rutt [7] further studied into the use of valves for guitar amplification, paying particular attention to the preamplifier stage, almost exclusively triode nonlinearity, suggesting that the common ECC83 triode stage was perhaps the most commonly used stage in guitar preamplifier design. In concert with the usual test methods using single frequency test sources Rutt also based his research on the more complex guitar waveform constructed of many frequencies of sinusoidal wave.

Bussey and Haigler pointed out the difference in the way an amplifier responds to single plucked strings and chords specifically mentioning, ”one subject felt that the difference between amps was an order of magnitude below the difference in striking the strings.” [6] Rutt determines that a valves pleasing soft-limiting of signals is due to ‘an induced voltage drop across the grid circuit source resistance’ as a result of grid current, and that it is this soft/grid limiting that allows the small transient nuances produced by the higher frequency guitar strings to still be present at the output, giving greater harmonic detail to the sound.

With the emphasis of all the researched papers being on the harmonic content of signals it was proposed to investigate into the differing harmonics incited by both the Montieth/Flowers and an ECC83 based preamplifier, Fig. 1. Particular interest was to be paid to the region most exploited by guitarists in search of the fabled “warmth” of valve amplification, the overload region.

3 – Objective Testing Methodology

In keeping with the original paper [54] testing of the Montieth/Flowers preamp was carried out at a HT of 200VDC. The primary indicator that the circuit was performing as recorded by the original paper was to monitor its output waveforms in the overload region.

Once satisfied that the circuit reacted to an input signal as expected, a systematic approach to increasing the input voltages from 10mV to 3.00V of a 1kHz sine wave was undertaken. At each incremental increase of 10mV the output was monitored and recorded both for its waveform shape and for its harmonic content using an Agilent 35670A signal analyser with its internal signal generator, thus reducing error and containing all testing within one piece of equipment. At each increment the HT supply was switched to the valve stage and adjusted using a variac at source to maintain 200VDC and the same measurements were taken.

All files were converted from .DAT to .csv and placed in to Excel for analysis. Readings of output voltage and total harmonic distortion were also taken giving a wealth of data from which to derive plots.

As much of the researched literature suggested the ‘valve sound’ becomes most pleasurable when operating in the region of overload and as such it was the authors opinion that the preamplifiers regions of overload should be closer scrutinised. The transfer characteristics of the two systems needed to be compared, and the variables between the systems needed to be reduced to generate a more specific research question for an audio evaluation.

It was decided that the only fair way to ascertain if there were indeed similarities between the two preamplifiers was to focus upon their region of overload. Further sets of measurements were taken to determine what value of input signal produced a set value of output THD%. Using the internal signal generator of the Agilent 35670A spectrum analyzer, input signal amplitude was gradually adjusted, giving time for each preamplifier to adjust and “settle” its operating parameters, until the required output total harmonic distortion was matched. This process was repeated for both systems to produce a set of data as shown in figure 3.

THD%	Input to M/F mVrms	Input to Valve mVrms
0.500	306.000	10.000
1.000	315.000	18.500
2.000	380.000	22.700
4.000	466.500	28.800
6.000	493.000	36.300
8.000	513.000	46.700
10.000	531.000	62.800
12.000	548.000	89.700
14.000	568.000	134.400
16.000	587.000	199.600

Figure 3 – Required input amplitude to achieve set THD%.

4 – Objective Analysis

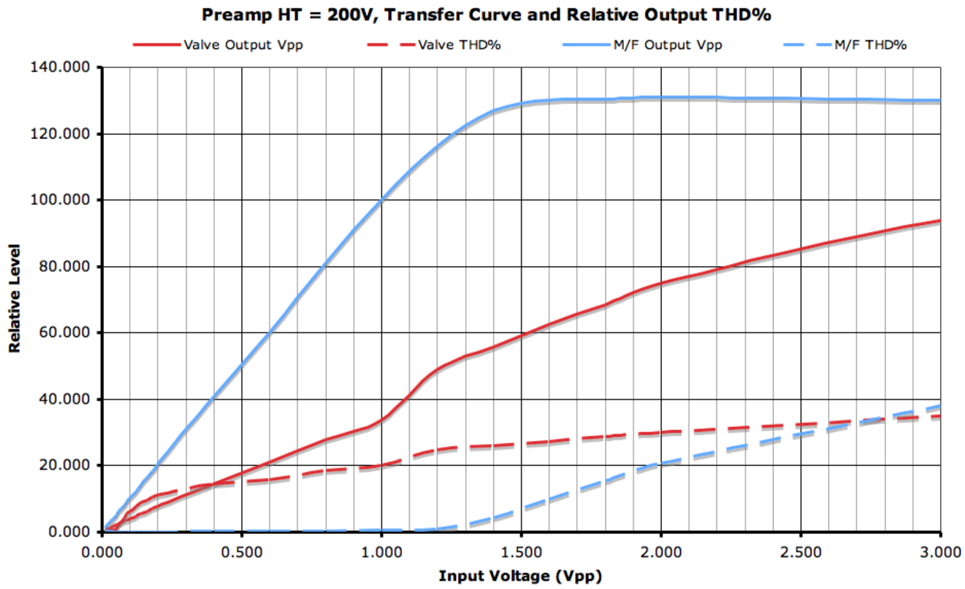
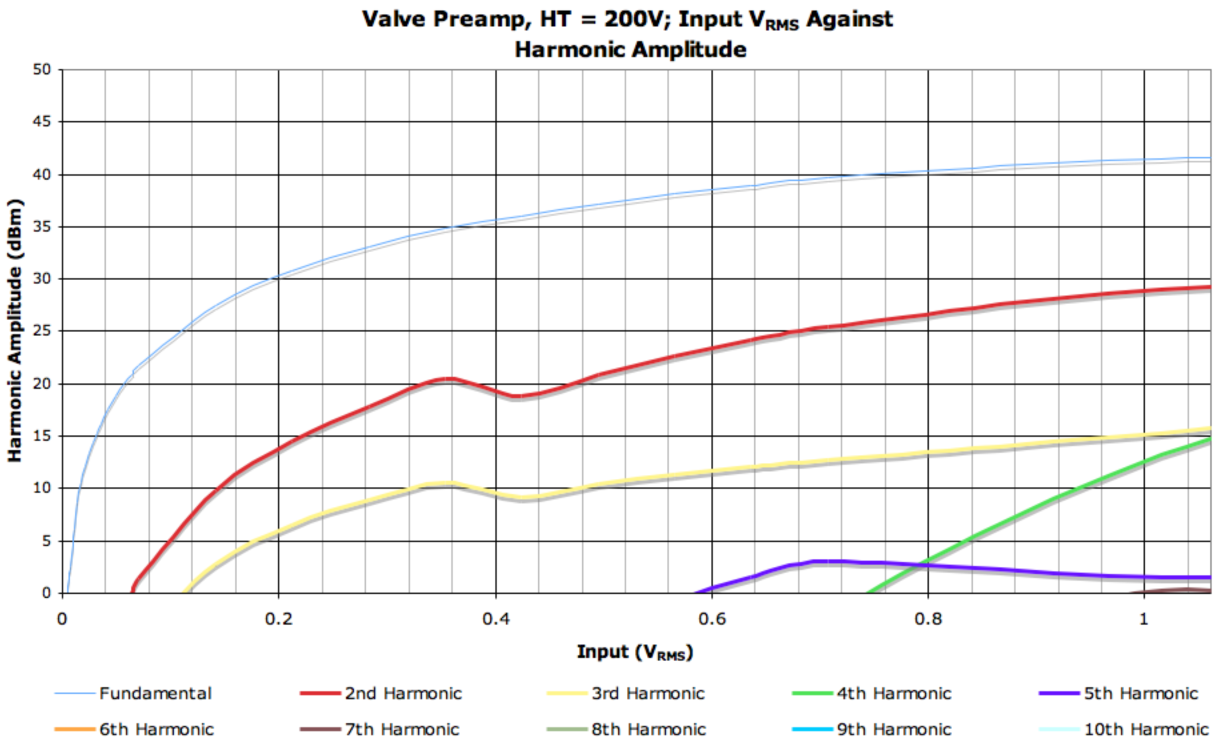


Figure 4 – Comparative transfer curves and relative THD%.

As figure 4 demonstrates, the M/F circuit reaches saturation long before the valve circuit as a result of its greater amplification factor of 100 compared to the valves 40. This produces a sharp knee, or a "hard limit", of any input signals greater than 1.5 V peak-peak (0.53VRMS). The effect of this "hard limit" is demonstrated by figure 5. The M/F preamps linear region, from 0 – 1Vp-p, (relating to 0 – 0.35VRMS in figure 5) clearly produces very undistorted output signal with very little generation of harmonic content.



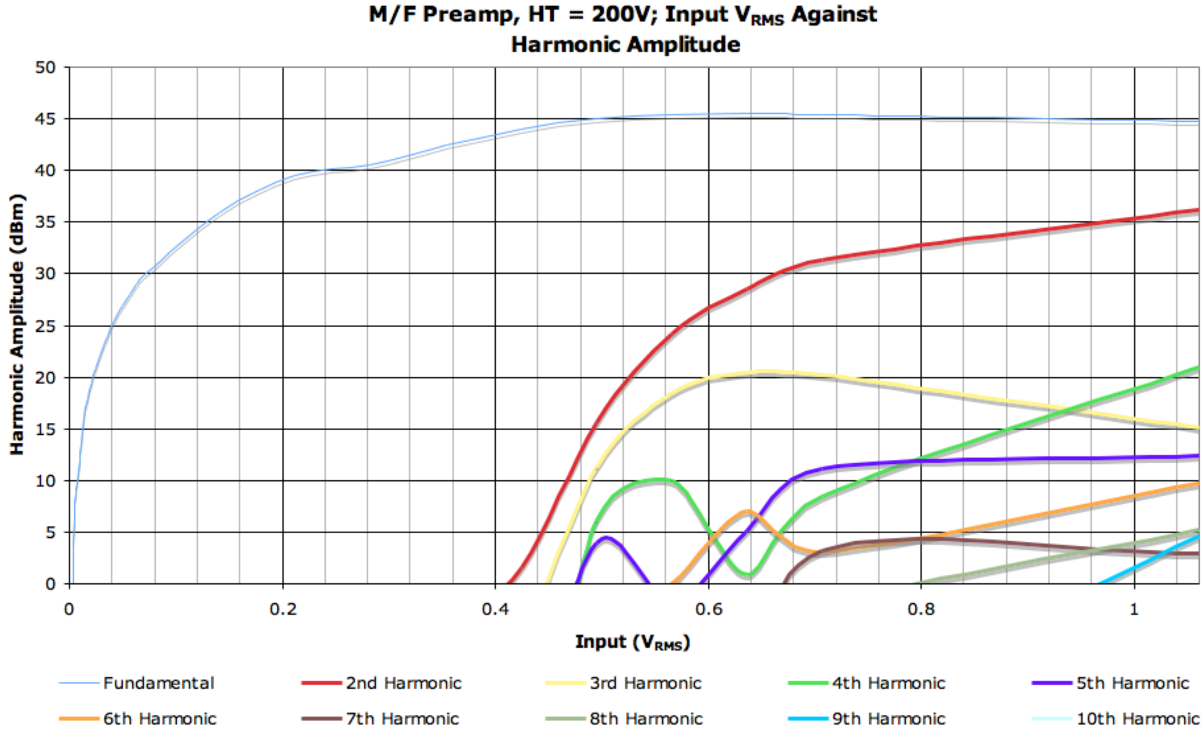


Figure 5 – Comparison of Output Harmonic Content at 200VDC.

In contrast to the M/F output, the valve displays the expected trait of low order harmonic generation throughout the range of input signals, with 2nd order being the most predominant. Further into overload and the M/F output shows a greater overall harmonic component density and it is this difference that requires closer inspection.

Figure 6 immediately shows the similarity between the two preamplifiers output Fourier Transforms when their THD%s had been matched. The low order harmonics can be seen as having closely correlating amplitudes and bandwidths. Figure 7 demonstrates the resulting harmonic amplitude plots.

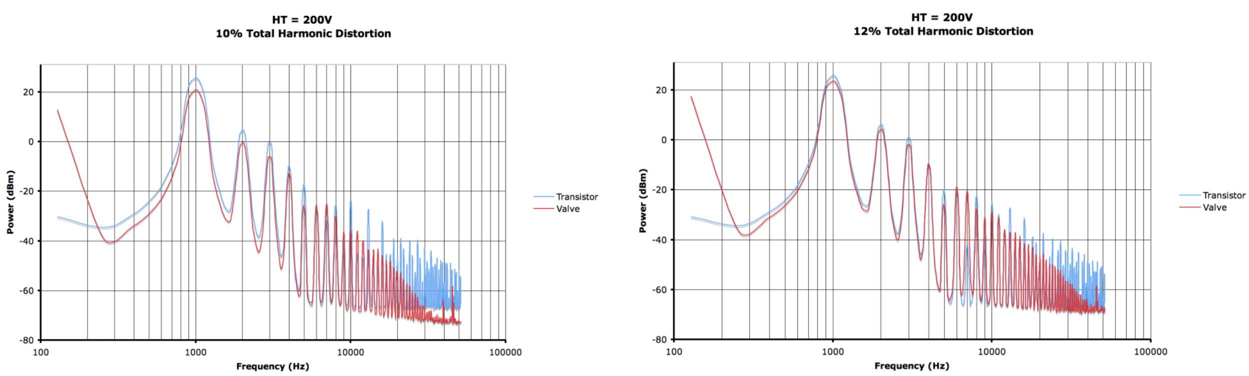
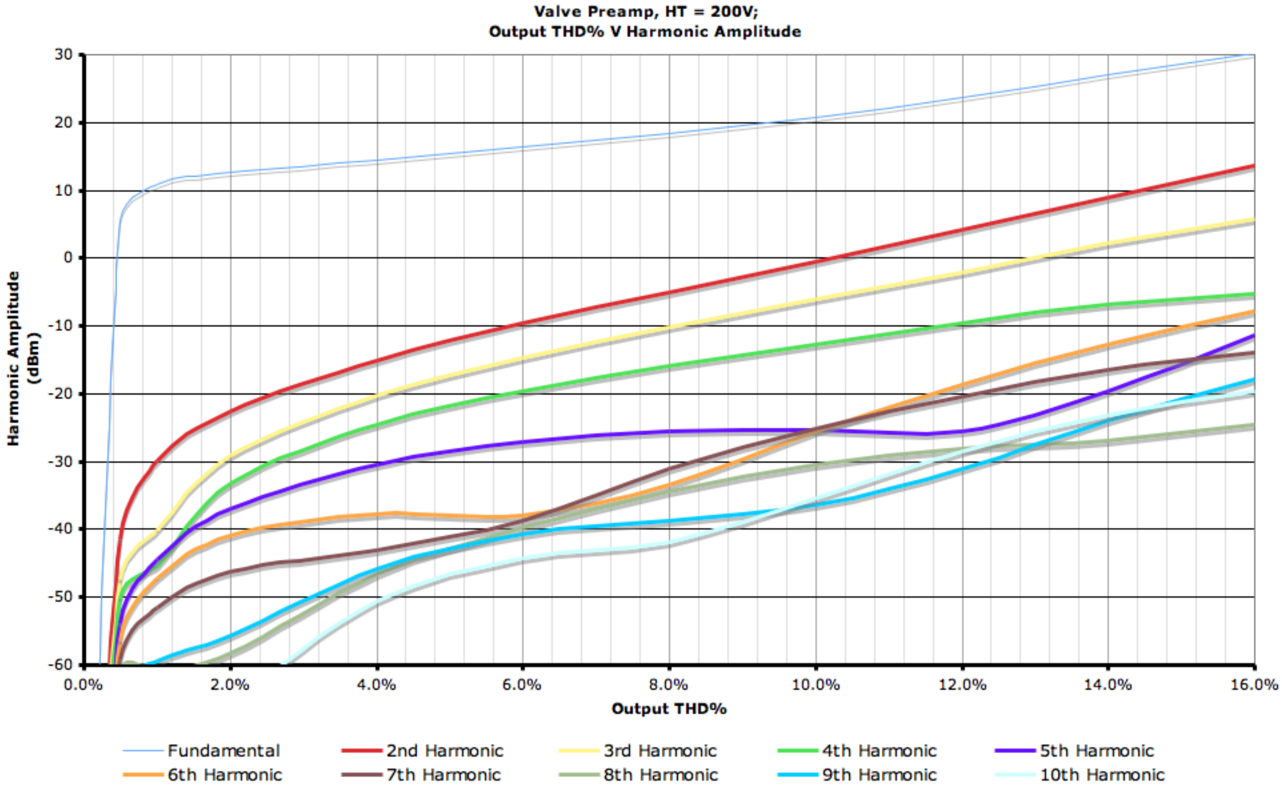


Figure 6 – FFT of both systems outputs at 10% and 12% THD.



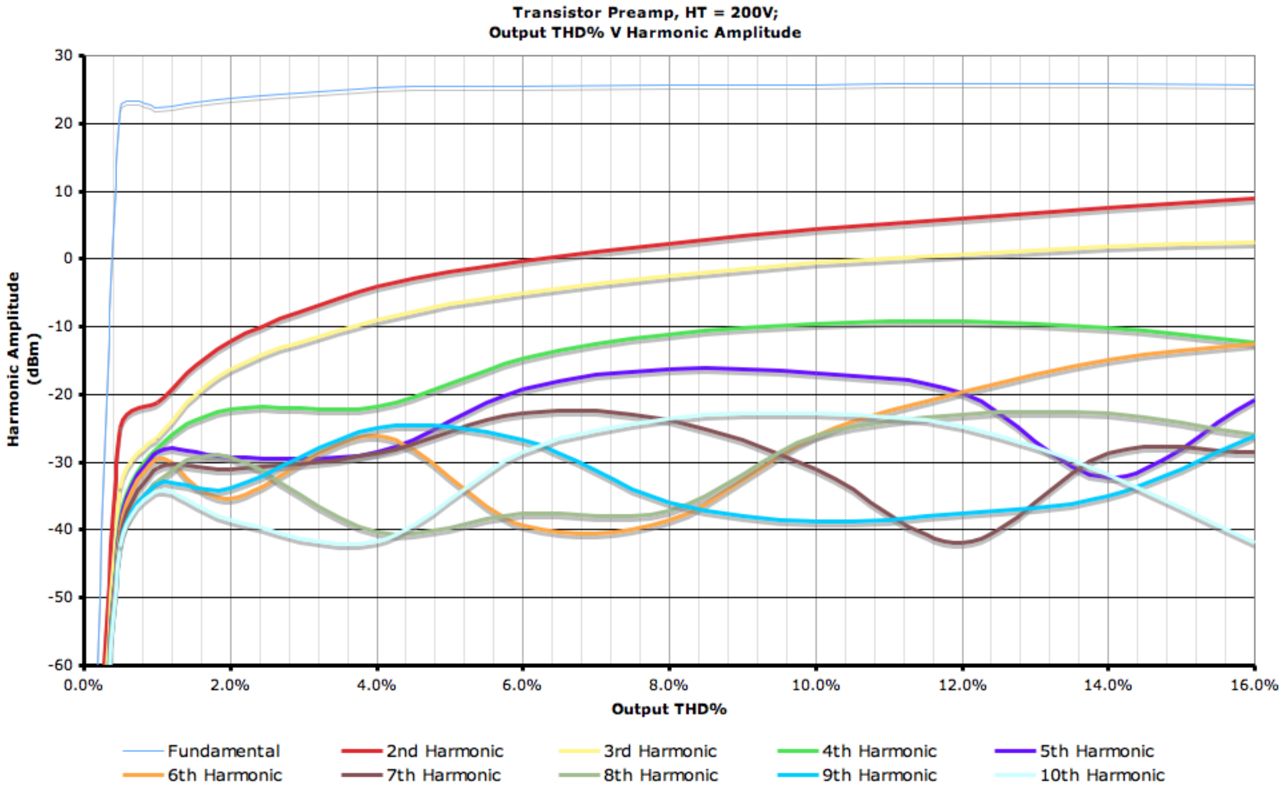


Figure 7 – Matched THD% Comparison of Output Harmonic Content at 200VDC.

The amplitude of the fundamental depicts the matching of the transfer curve nicely, in that both systems were now being tested at the point when the preamplifiers were saturating and the harmonic distortion was being produced. The harder limit of the M/F topology is however still apparent from the rate at which its harmonic density increases. For example the M/F 2nd harmonic amplitude has an increase of 40dBm per 1%THD, whereas the ECC83 has a slower increase of 40dBm per 2%THD. This greater speed of increase in amplitude is apparent in all the M/F harmonics.

Hamm articulated; “the primary colour characteristic of an instrument is determined by the strength of the first few harmonics,” and also explained that “the ear seems very sensitive to the edge harmonics.” [3] This suggested that closer inspection was required of the low order harmonics, but that the difference in the weighting of the higher order harmonics was likely to be a perceivable audible difference. The valves relatively uniform increase of approximately 26dBm per 10%THD across harmonics order 1-10 could be speculated as aiding in the perceived “smooth” sound of a valve. Whereas the much less M/F average of 10dBm per 10%THD, along with the interchanging dominance of its harmonics could produces a greater perceived “raspy dissonant quality.” [3]

The individual harmonic amplitudes in dBm, (relative to 1mW into 600Ω) were converted into relative amplitudes in mW, (dBm to watts = 10(dBm/10)) and then calculated as a percentage of the fundamental amplitude to produce plots with comparable axis to those used by Hamm and Monteith and Flowers. These plots display the correlation of the “first few harmonics” between the two topologies, figure 8. The full-scale version along with magnified comparisons of the 2nd, 3rd, 4th, and 5th harmonics can be found in figures 16-19 at the end of the paper.

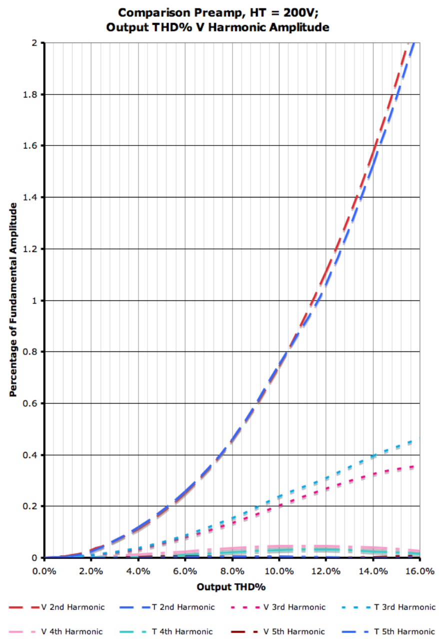


Figure 8 – Comparative “Hamm” plots of low order harmonics as percentage of fundamental amplitude.

Initially the 2nd harmonic of the valve amplifier is more dominant than the Monteith/Flowers, then between the region of 3-8%THD the Monteith circuit has a slight increase in 2nd harmonic until 10.4%THD and upwards where the valve 2nd harmonic is the stronger presence again. However the shape of the two plots are very well correlated.

The Monteith/Flowers shows some relation to expected transistor technology harmonic production, with the 3rd harmonic being the more prominent of the two topologies through the whole range from 0-16%. Both stages show signs of tailing off from 14%THD with the valve stage demonstrating a more severe roll off resulting in a 22.35% difference between the two harmonics amplitudes at 16%THD, the valve being the comparative lesser of the two.

The shape of the curves for the 4th harmonic again demonstrate similar characteristics between the two pre-amplifiers, but ultimately the valve stage has a greater amplitude in 4th harmonic than the Monteith/Flower. With the knowledge that even harmonics produce “choral” or “singing” sounds it could be predetermined that the valve stages more prevalent even order hamonics, 2nd and 4th, would result in the valve sound being preferable in this instance.

However, when analyzing the 5th harmonic components, despite there being a dominance of the Monteith/Flowers 5th from 6.7% to 12.8%THD it is the valve stage whose 5th harmonic at 16%THD is 69.1% larger than that of its emulator possibly “covering” or “masking” any “pleasant” perceived sound as a result of its 4th and 2nd harmonic.

With close inspection of the objective measurements complete, and some speculative hypothesise as to possible perceived subjective impressions of the two amplifiers made, an audio evaluation was undertaken to produce subjective results.

5 – SUBJECTIVE AUDIO EVALUATION

Research Question: ""In relation to the reference, how much do you prefer the samples whilst comparing their distortion characteristics; More, less or the same?"

Hypothesis: The perceived distortion characteristics of an ECC83 valve preamplifier will be more pleasing to the listener than that of the Monteith and Flowers topology when run at 200VDC.

It must be stated from the outset that the aim of conducting this auditory evaluation was not to produce objective results but more to produce a set of data that is reproducible in another laboratory. This set of tests is required in order to provide some subjective backing to objective results already obtained from both preamplifier stages using scientific evaluation. It must also be stated that only eleven people gave their time to take the test and as such the audio evaluation can only be considered a 'pilot'.

It is the authors' belief that the attributes as outlined in ITU-R Recommendation BS.1116-1 [6] provide grounds for a suitable perceptual model for analysing basic audio quality. "This single, global attribute is used to judge any and all detected differences between the reference and the object." [6] However, as suggested by Bech [7] "This recommendation is intended for use in the assessment of systems that introduce impairments so small as to be undetectable without rigorous control of the experimental conditions and appropriate statistical analysis. If used for systems that introduce relatively large and easily detectable impairments, it leads to excessive expenditure of time and effort and may also lead to less reliable results."

In the case of this audio evaluation, initial objective tests showed that the two systems exhibited "intermediate levels of quality" [9] and as such ITU-R BS.1534-1, more commonly known as MUSHRA (Multiple Stimulus Hidden References and Anchor) was regarded as more appropriate. "The MUSHRA test method uses the original unprocessed programme material with full bandwidth as the reference signal (which is also used as a hidden reference) as well as at least one hidden anchor." [9] By having a hidden reference the validity of each subject's responses could be determined, and using a hidden anchor should have helped in the statistical evaluation of the results by reducing possible contraction bias [7] of listeners being afraid to use the extremities of the scales provided, figure 9 demonstrates.

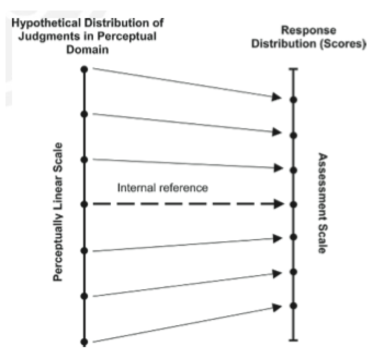


Figure 9 – Contraction bias Model. [61]

Despite there being numerous scales stated in various ITU standards, it was decided to utilise a "polar scale", and idea proposed by Watson [10], using "a vertical line 20mm long with no labels other than a + sign at the top and a – sign at the bottom, to indicate the polarity of the scale." In this case using a positive and negative labeled axis to demonstrate whether the sample had more or less preferable distortion characteristics to the reference.

6 –Audio Evaluation Methodology

Two differing samples were recorded of a dry guitar signal using a DI (direct input) box. The first sample consisted of a series or riff of single picked notes; the second sample combined single picked notes with some chordal work.

The recorded samples were then measured and adjusted in level accordingly using Matlab and SoundForge to provide average RMS levels at equivalent dBFS (relative to full scale of a digital signal) to the 1kHz sine waves used to achieve set levels of THD% at output in the objective measurements. The conversion spreadsheet can be seen in the below figure:

HT = 200V Monteith/Flowers
0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	306.000	-26.021	432.749	1638.400	0.050	2.970
1.000	315.000	-25.769	445.477	1686.588	0.051	2.965
2.000	380.000	-24.139	537.401	2034.614	0.062	3.520
4.000	466.500	-22.358	659.731	2497.757	0.076	4.172
6.000	493.000	-21.878	697.207	2639.644	0.081	4.297
8.000	513.000	-21.533	725.492	2746.729	0.084	4.387
10.000	531.000	-21.233	750.947	2843.106	0.087	4.429
12.000	548.000	-20.959	774.989	2934.128	0.090	4.467
14.000	568.000	-20.648	803.273	3041.213	0.093	4.497
16.000	587.000	-20.362	830.143	3142.944	0.096	4.518

HT = 200V Valve Stage
0dB reference = digital full scale = 6.120 Vrms = 8.65 Vp

THD%	Input mVrms	Input dB	Input mV peak	Input Sample Level	Input double' std	Output Vrms
0.500	10.000	-55.735	14.142	53.542	0.0016	0.437
1.000	18.500	-50.392	26.163	99.054	0.0030	0.824
2.000	22.700	-48.615	32.103	121.541	0.0037	0.973
4.000	28.800	-46.547	40.729	154.202	0.0047	1.226
6.000	36.300	-44.537	51.336	194.359	0.0059	1.572
8.000	46.700	-42.349	66.044	250.043	0.0076	2.066
10.000	62.800	-39.776	88.813	336.247	0.0103	2.853
12.000	89.700	-36.679	126.855	480.276	0.0147	4.252
14.000	134.400	-33.167	190.070	719.611	0.0220	6.530
16.000	199.600	-29.732	282.277	1068.708	0.0326	9.840

Figure 10 – Conversion Calculation for Sample Level dBFS at HT = 200V.

Each sample was adjusted to the calculated input dB value in SoundForge, then run through a standard deviation program in Matlab to obtain the average full-scale (digital) sample level of the excerpt. These samples then became the signal source to the two preamplifiers

The level adjusted samples, one of length 7.00 Seconds, and one of length 9.00 seconds, were 're-amped' out of a Digidesign 003 rack to the inputs of the valve stage and the M/F stage. There was inherent measured noise of amplitude 32.7mVRMS from the output of the 003. Due to the low levels of signal amplitude required for the lower percentage harmonic distortions this noise, seen at the input to the two preamplifiers, was far from ideal. The signal to noise ratios were calculated and can be found plotted in figure 12. The outputs of each system were then recorded at 16 bit 44.1KHz as mono wav files.

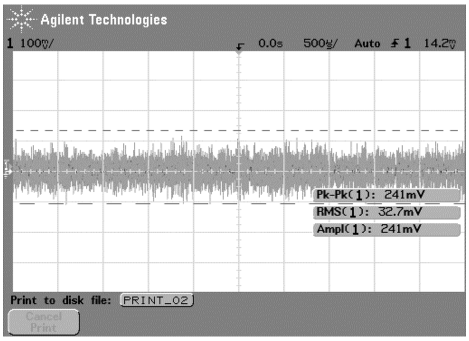


Figure 11 – Measured noise from the 003 output.

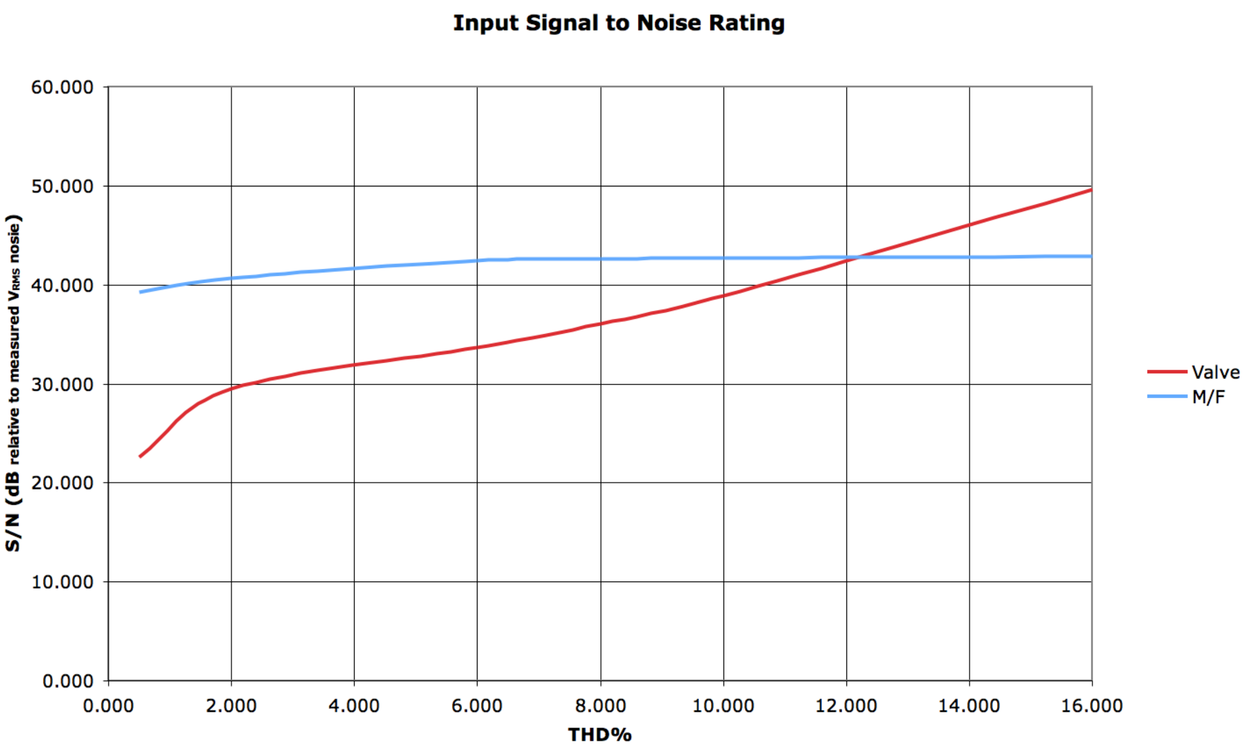


Figure 12 – Measured signal to noise ratings for input signals.

The resulting recorded samples of varying amplitude were monitored for equal loudness using the Orban level meter, adhering to ITU-R BS.1770 and the levels changed accordingly. This prevented any preference of sample as a result of listening level. It was decided to reduce the level of the higher percentage THD recordings down to the lower loudness levels of the lower THD% recordings to avoid amplification of the noise floor as a result of lower input signals used at source.

These leveled recordings were then reproduced at a reference listening level of 75dBA* [12], calculate using a binaural head,, through Sennheiser HD650 headphones with a known SPL of 103dB at 1kHz 1VRMS and nominal impedance of 300Ω.

The recorded and levelled (to equal loudness) samples were then placed into a modified MUSHRAM.m Matlab file producing nine experiments testing in bands of 0-4%THD, 4-10%THD, and 10-16%THD. The reference for each band being the clean input signal used for the recordings, the anchor for each experiment being the reference sample low-pass filtered at 3.5kHz with a roll off of 25dB per 500Hz, both samples leveled accordingly.

A previous study with loudspeakers addressed the issue of whether or not “the preferences of trained listeners can be extrapolated to a larger population of naïve, untrained listeners.” [14] This study concluded, “Trained listeners have the same preferences as untrained listeners.” For this reason it was suggested that although it would be advantageous for the all of the subjects used for this test to be experienced trained listeners, the possible impracticalities of this meant that some untrained but still experienced listeners could also be used.

Because the objective of this auditory evaluation was to provide evidence as to whether the Monteith/Flowers transistor preamplifier sounds “better” or more preferable than a valve based stage then it seems reasonable to suggest that using untrained experienced listeners would not have an adverse effect upon the resulting data.

In accordance with ITU-R BS. 1534-1 a training stage was utilised before the grading stage to allow all listeners to become accustomed and comfortable with the GUI, (general user interface) programmed using Matlab script.

All testing took place in a controlled environment to minimise any distractions and create an “even playing field” for all test subjects. While the MUSHRA test was under way, all on-screen objects that did “not correspond to the currently auditioned stimulus” [8] were disabled.

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*dBA calculation using the A-weighted filter responses determined by Fletcher-Munson curves of human ear sensitivity. This filter is needed “to obtain objective measurements which correlate well with human response.” [13]

7 – AUDIO EVALUATION ANALYSIS

Of the eleven subjects to partake in the audio evaluation, two sets of results were declared unusable. One subject demonstrated contraction bias, “a conservative tendency in using the grading scale,” [8], scoring all samples on a reduced scale of -1 to 1. The other suggested in his comments about the test; “I love distortion, it all sounds good to me” subsequently scoring all results on a shifted and reduced scale from +18 to +70.

The remaining ten subjects results were arranged in to useable data format in excel. The collated data for both stages was then run through a two-way ANOVA test, figure 13, the results of which allowed four statements to be drawn:

- Interaction between sample and THD% is not significant.
- THD% is not a significant factor for the transistor stage.
- THD% is a significant factor for the valve stage.
- Sample selection is not significant for either transistor or valve stages.

Because the choice of sample had no significant effect upon the overall result, the scores from sample 1 and sample 2 could be combined.

ANOVA - Transistor

Source of Variation	SS	df	MS	F	P-value	F crit
THD%	4391.67222	9	487.96358	1.27823276	0.25250584	1.93881904
Sample 1 or 2	301.605556	1	301.605556	0.79006327	0.37541536	3.90023603
Interaction	923.338889	9	102.59321	0.26874547	0.98212697	1.93881904
Within	61079.7778	160	381.748611			
Total	66696.3944	179				

ANOVA - Valve

Source of Variation	SS	df	MS	F	P-value	F crit
THD%	2949.97917	9	327.775463	2.48495809	0.01105092	1.93881904
Sample 1 or 2	258.001389	1	258.001389	1.95598118	0.16387909	3.90023603
Interaction	1292.29028	9	143.587809	1.08857961	0.37406479	1.93881904
Within	21104.6111	160	131.903819			
Total	25604.8819	179				

Figure 13 – Two-way ANOVA test for Valve and M/F stage.

The combined averages were plotted and 95% confidence levels calculated for each result. Because the sample size was less than thirty, z-values could not be used, instead they were calculated using t-values and a degrees of freedom chart. The results of this combined plot, figure 14, show relatively conclusively that all through the region of overload, defined in this case as 0.5 – 16% total harmonic distortion, it can be stated with 95% confidence that the valves distortion characteristics were preferred to those of the Monteith and Flowers pre-amplifier.

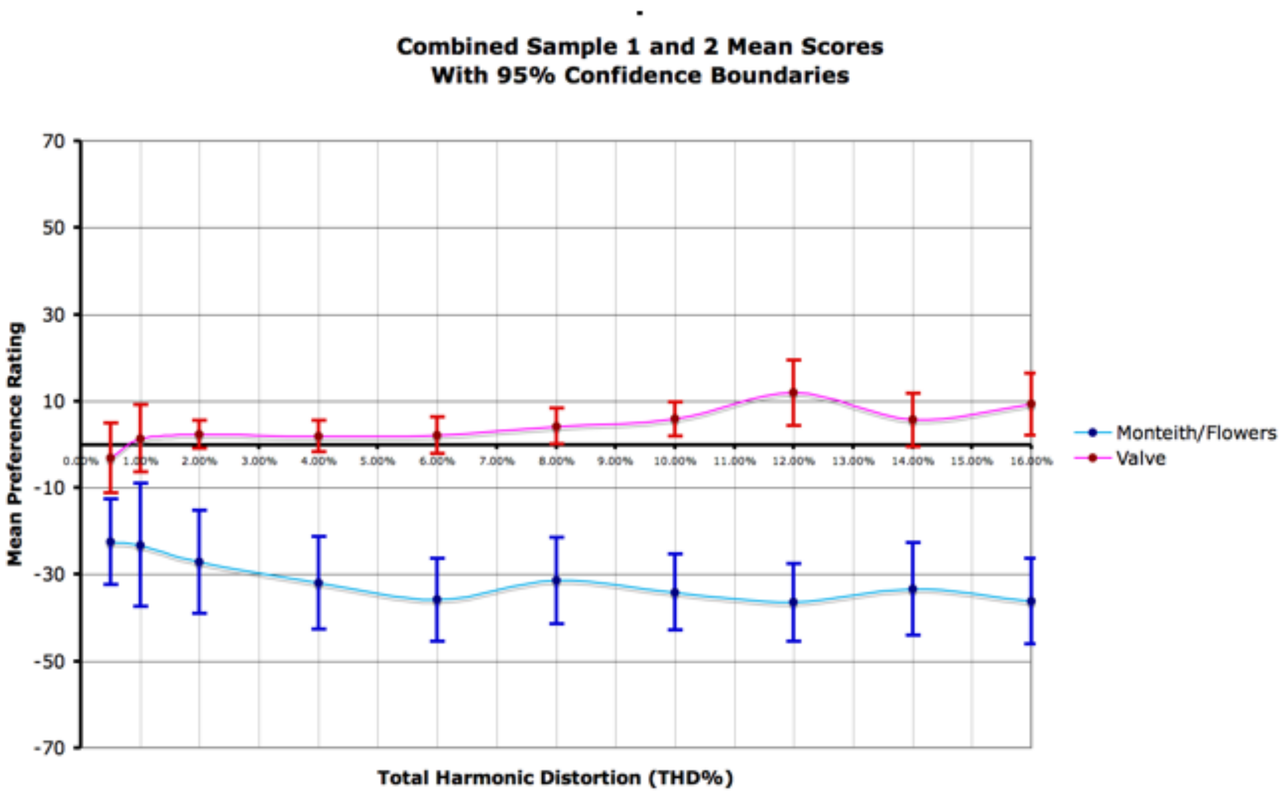

Figure 14 – Combined Preference Ratings with 95% Confidence Boundaries.

One interesting observation is the comparison between the signal to noise plot, figure 12, and the preference plot for the valve stage. The correlation between the two lines could well suggest that the slight trend in valve preference of higher THD%s in comparison to low THD%s could be as a result of the noise on the sample recordings as a result of the noisy 003 output.

Further to the two-way ANOVA, analysing the variance within the M/F and Valve results separately, an n-way ANOVA was completed on the entire group of results, figure 15, allowing further conclusions to be drawn.

ANOVA

Source of Variation	*SS*	*df*	*MS*	*F*	*P-value*	*F crit*
Transistor Vs Valve	56347.9587	1	56347.9587	296.739024	2.8026E-38	3.90023603
THD% significance	429.350347	9	47.7055941	0.2512267	0.98592052	1.93881904
Interaction	3241.47535	9	360.163927	1.89669146	0.05590139	1.93881904
Within	30382.5	160	189.890625			
Total	90401.2844	179				

Figure 15 – N-way ANOVA analysis of combined audio evaluation results.

8 – Conclusions and Further Work

The objective results of this paper allow the conclusion to be drawn that despite the even order harmonic dominance of the output of Monteith and Flowers' high voltage pre-amplifier, and the notable similarities in 2nd order harmonic dominance, the overall harmonic component density, particularly in the upper octave ranges, differs quite significantly to that produced by the valve pre-amplifier under test. There is therefore a marked difference between the two pre-amplifier topologies with particular respect to THD% vs. higher order harmonic amplitudes.

Hamms theory that the valve sound lies in the subtle differences of upper order harmonics may well be correct, subjective analysis of the two systems provides some grounds to suggest that the over crowded higher order harmonics of the Monteith/Flowers in comparison to the valve stage does prove detrimental to its sound as perceived by the listener.

From the ANOVA analysis, an F-statistic of 296.54 in conjunction with a very small p value of 2.80262E-38 allows conclusion that with 99% certainty the valve sound in this scenario is preferable to the Monteith and Flowers design.

It can also be said that with 98.6% confidence the total harmonic distortion percentage is not a significant factor when directly comparing distortion characteristics of a system, and as such it would seem fair to conclude that the use of matched output THD percentages is a reasonable way of producing comparative results between these two systems distortion characteristics.

It must be duly noted that there are indeed many other factors that may determine a subjective preference of one system to the other. In this paper the question was directly asked of subjects to grade the distortion properties. Indeed slew rate, and transient response are two key factors among many that could well prove to have been detrimental to the Monteith and Flowers circuits perceived output and as such there are grounds for further investigation in to these areas. It can not be denied that objectively the Monteith and Flowers circuit does demonstrate a close correlation of output harmonic content, but ultimately it is the conclusion of this paper that transistors used in this circuit topology do not sound "better" than valves.

9 – References

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10 – FIGURES

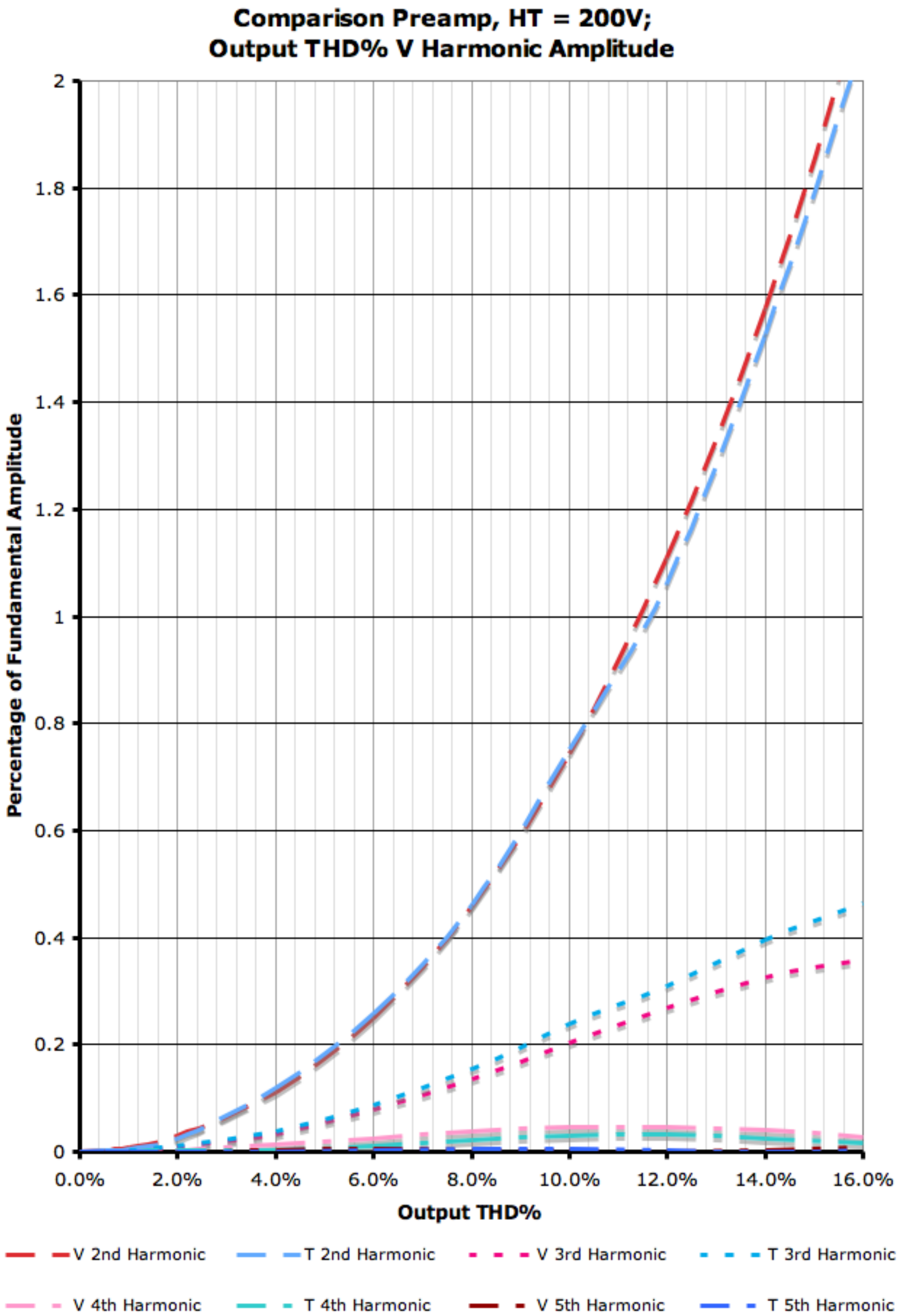


Figure 16 - Equivalent Hamm plots from 0.5% to 16% THD for both systems, showing the percentage of each harmonic in relation to the fundamental.

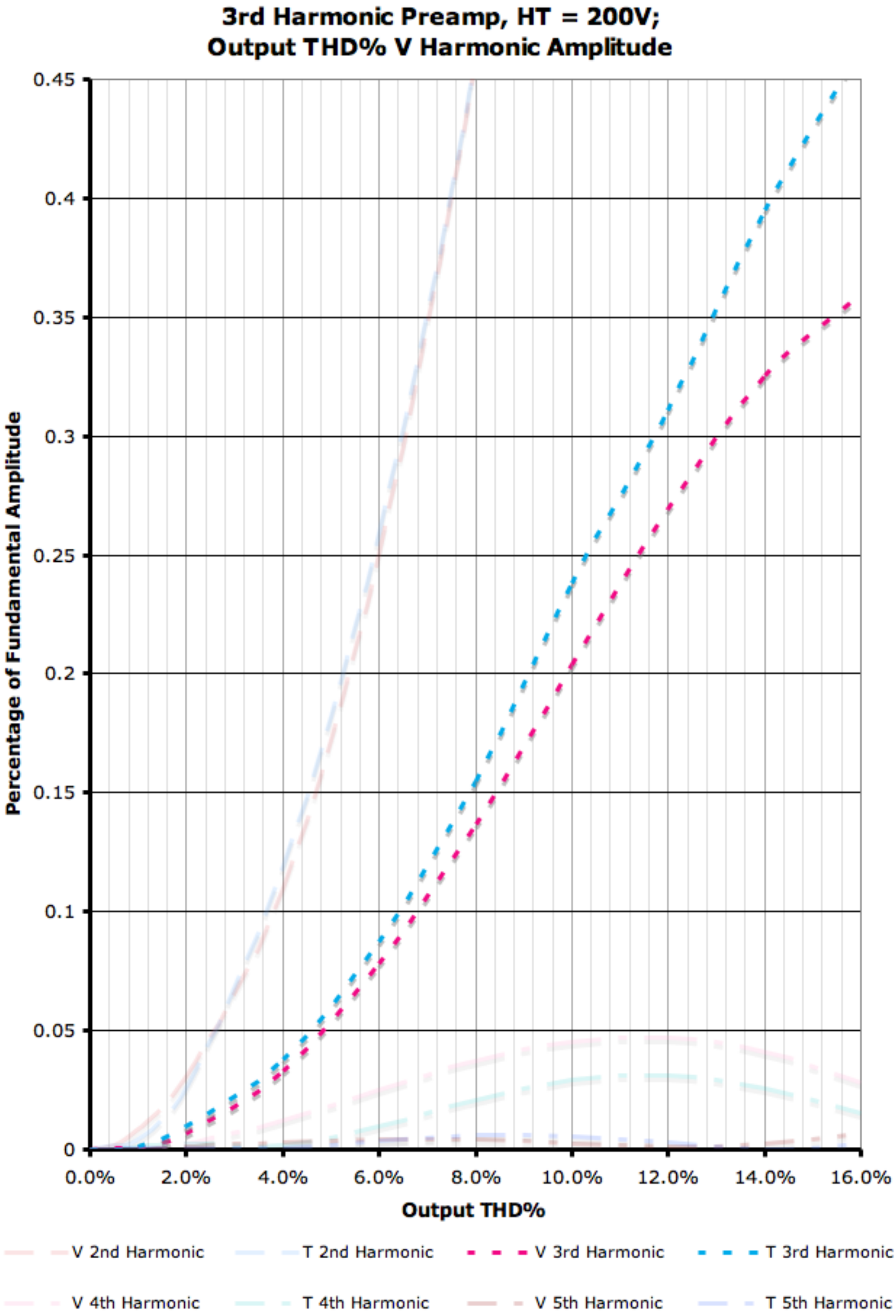


Figure 17 - 3rd harmonic comparison as a percentage of the fundamental.

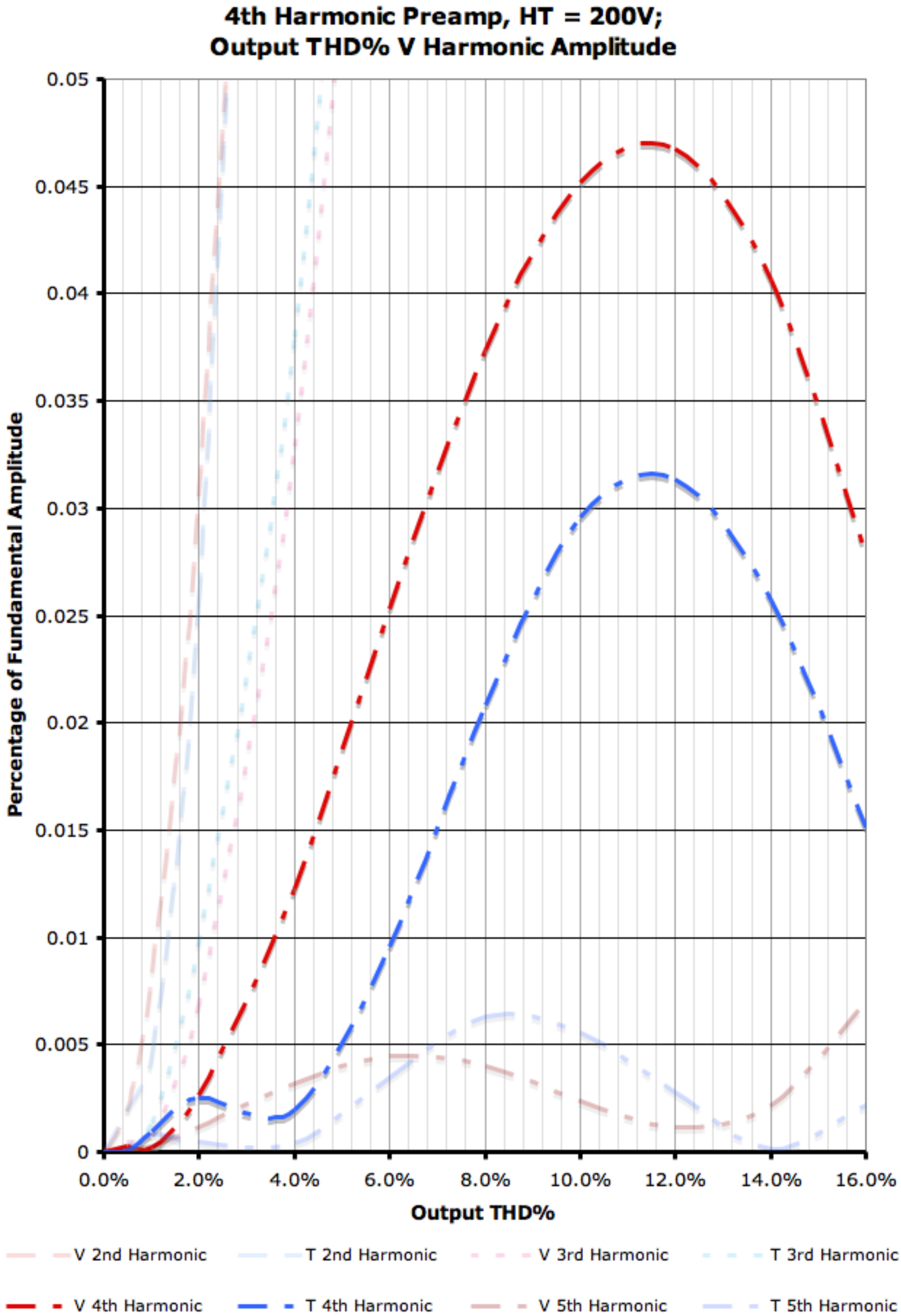


Figure 18 - 4th harmonic comparison as a percentage of the fundamental.

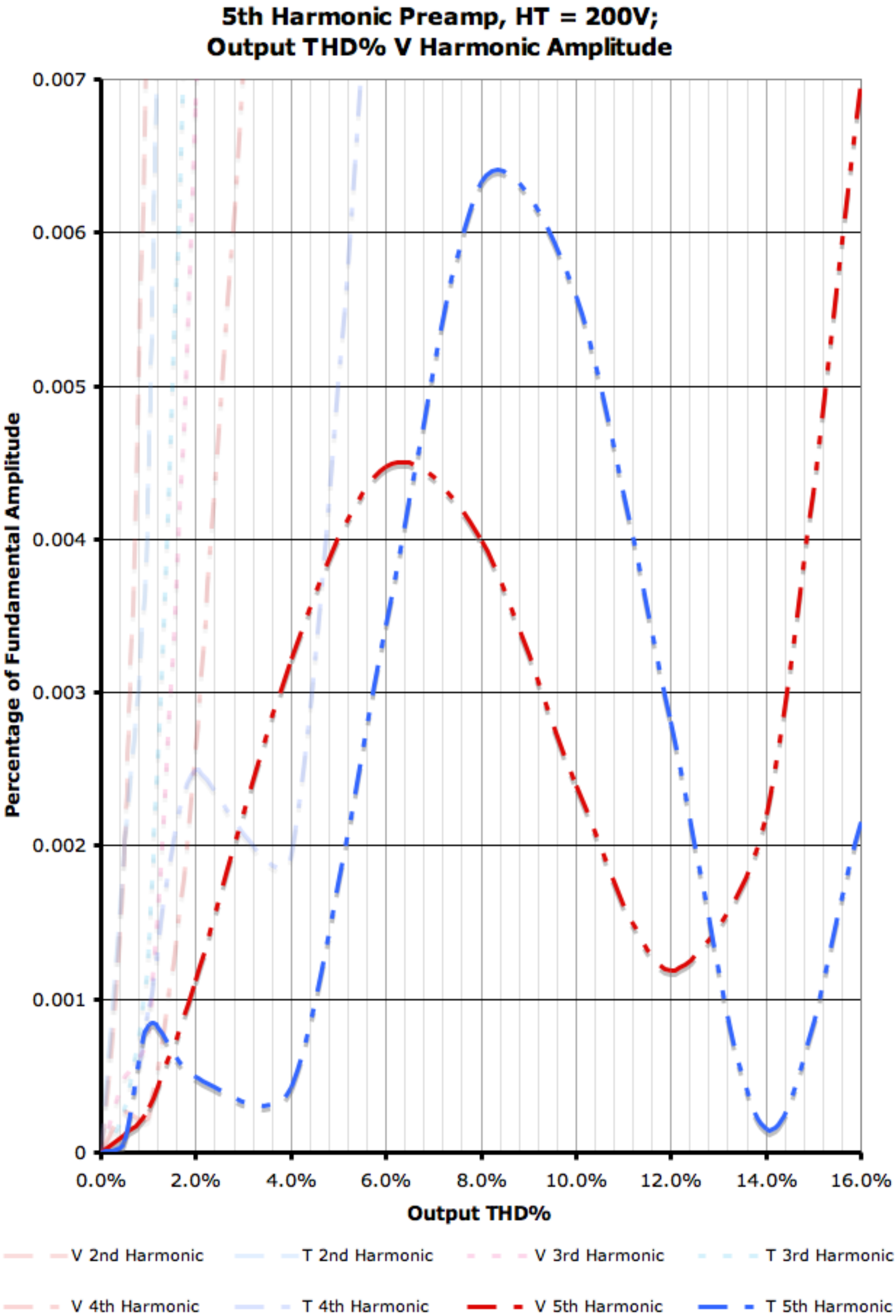
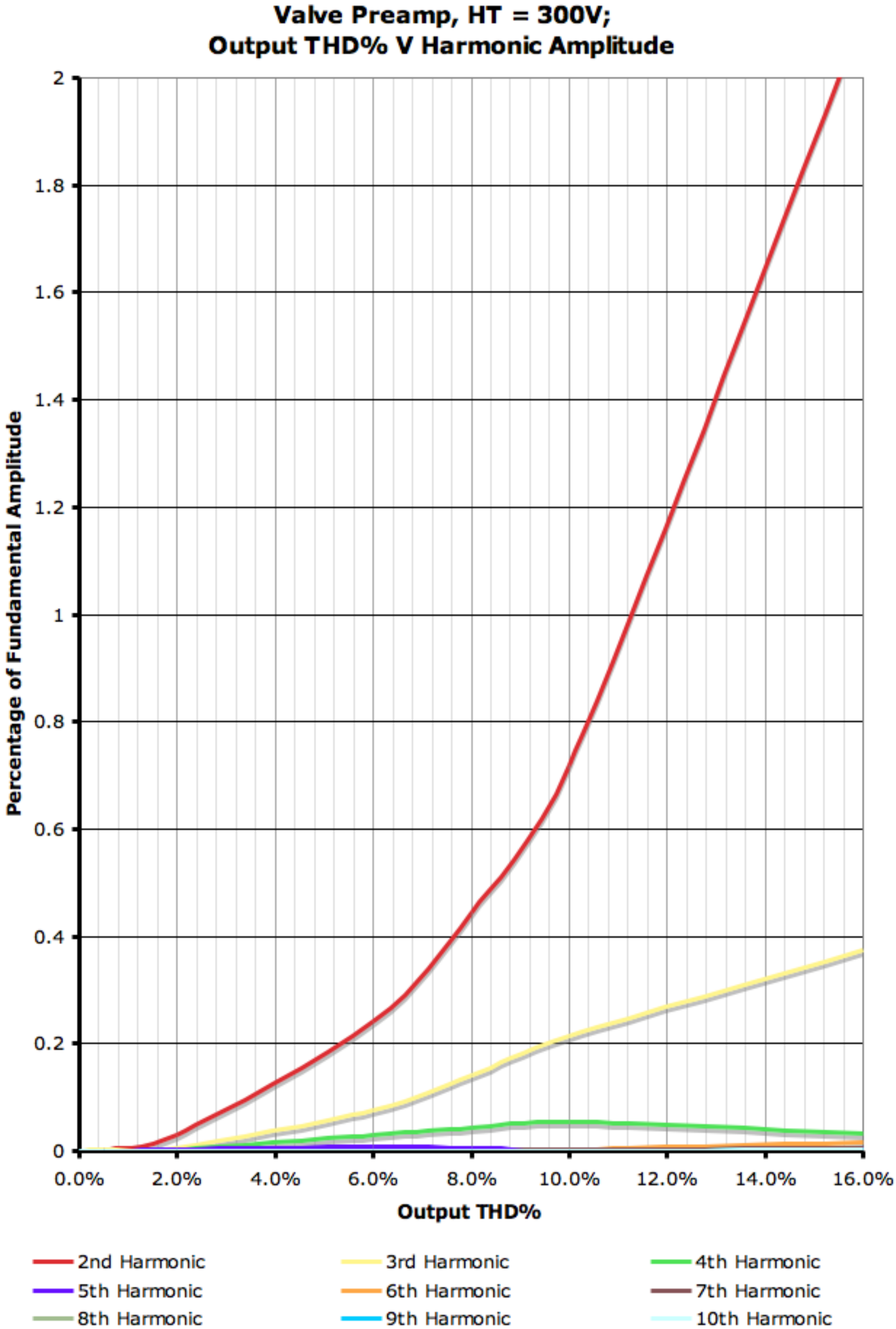
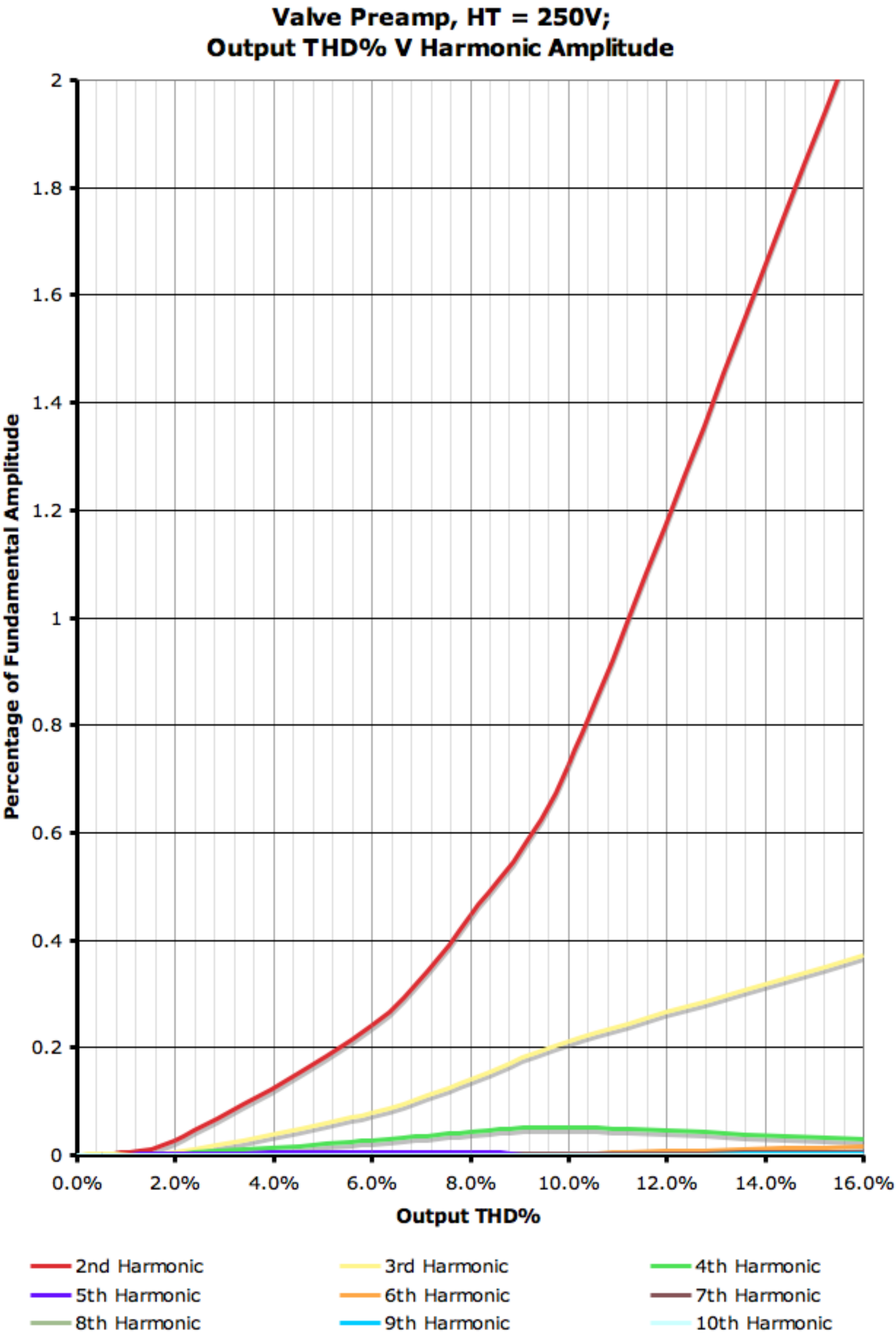
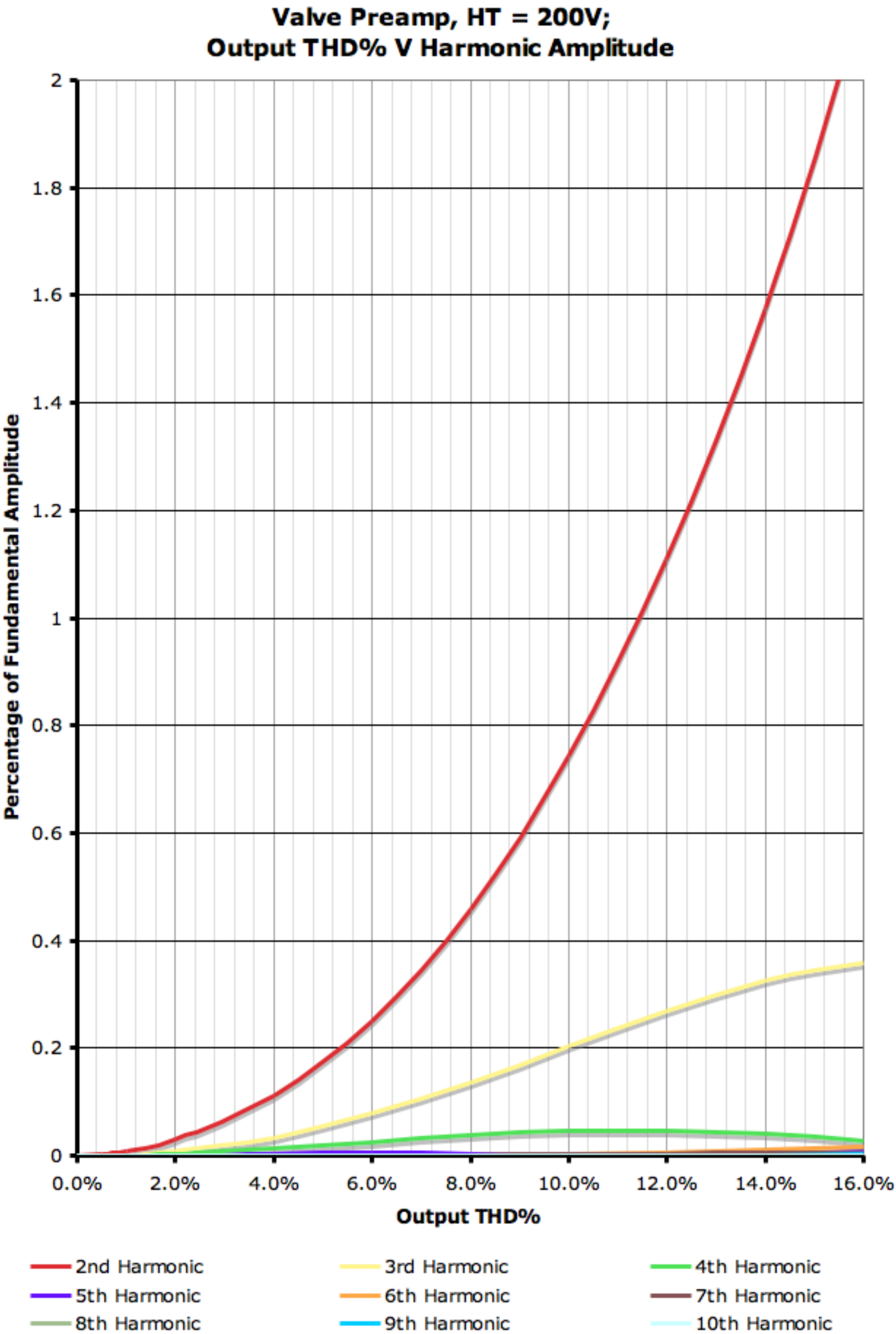


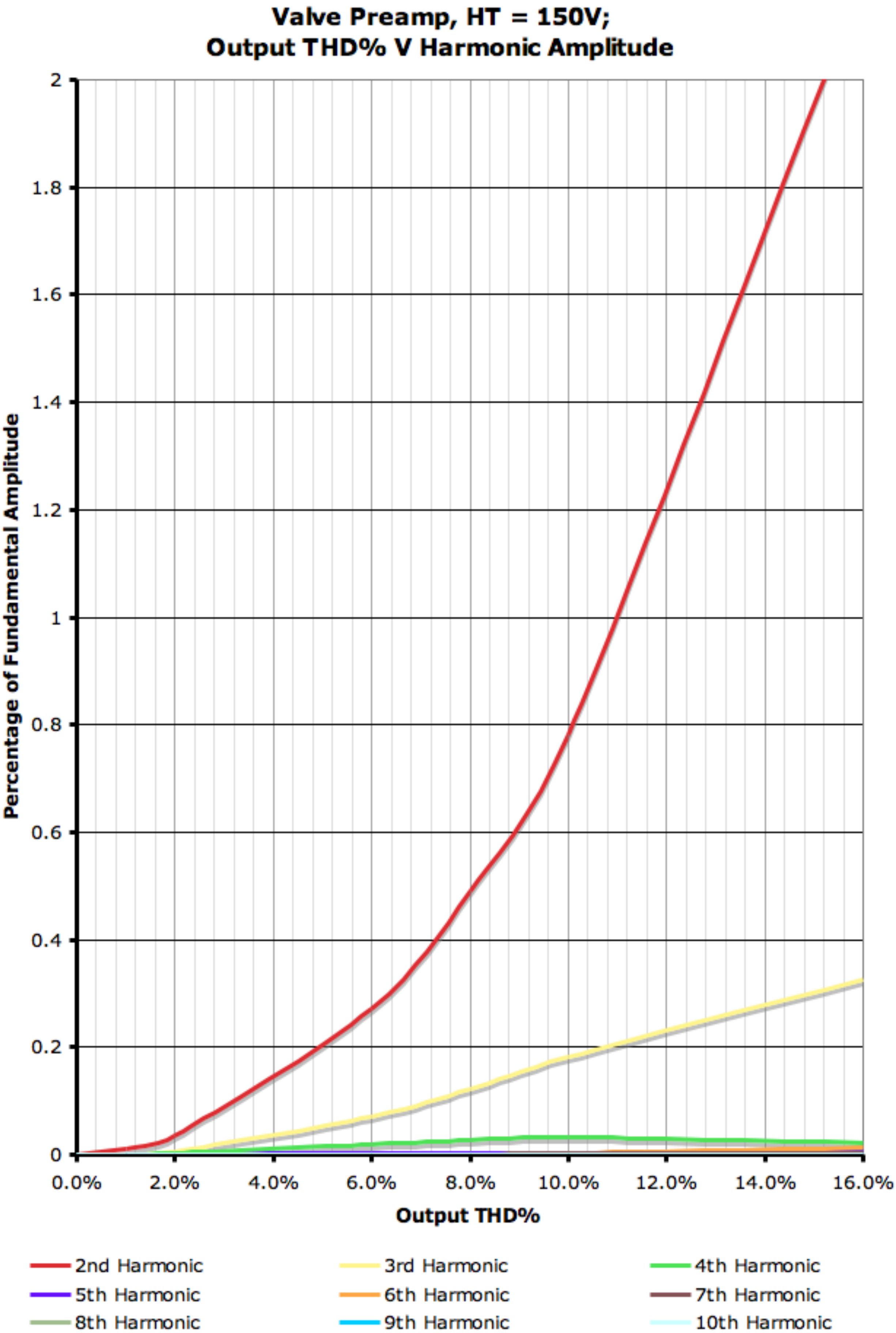
Figure 19 - 5th harmonic comparison as a percentage of the fundamental.

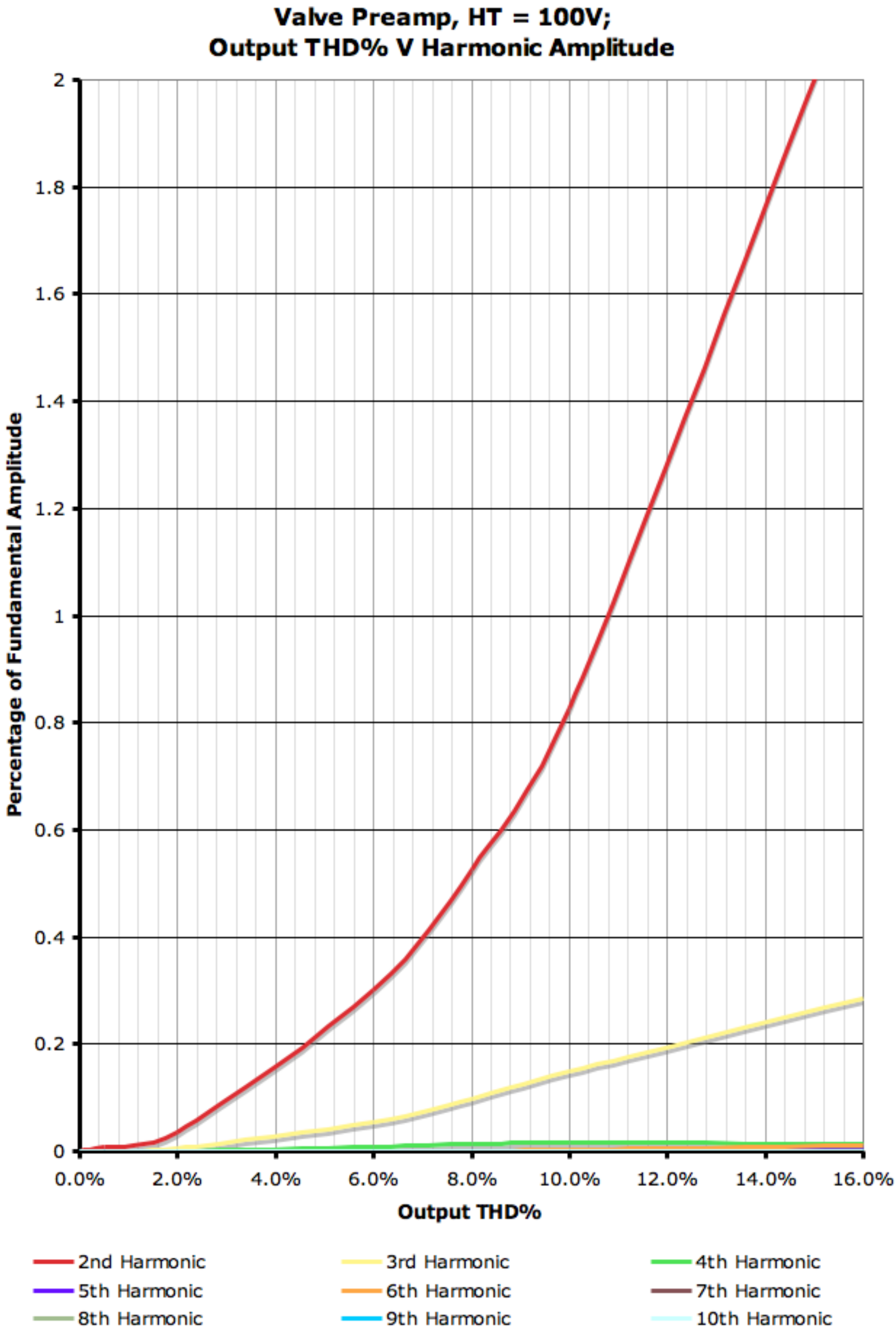
8.15 - Equivalent Hamm Plots for Valve Anode Reduction:

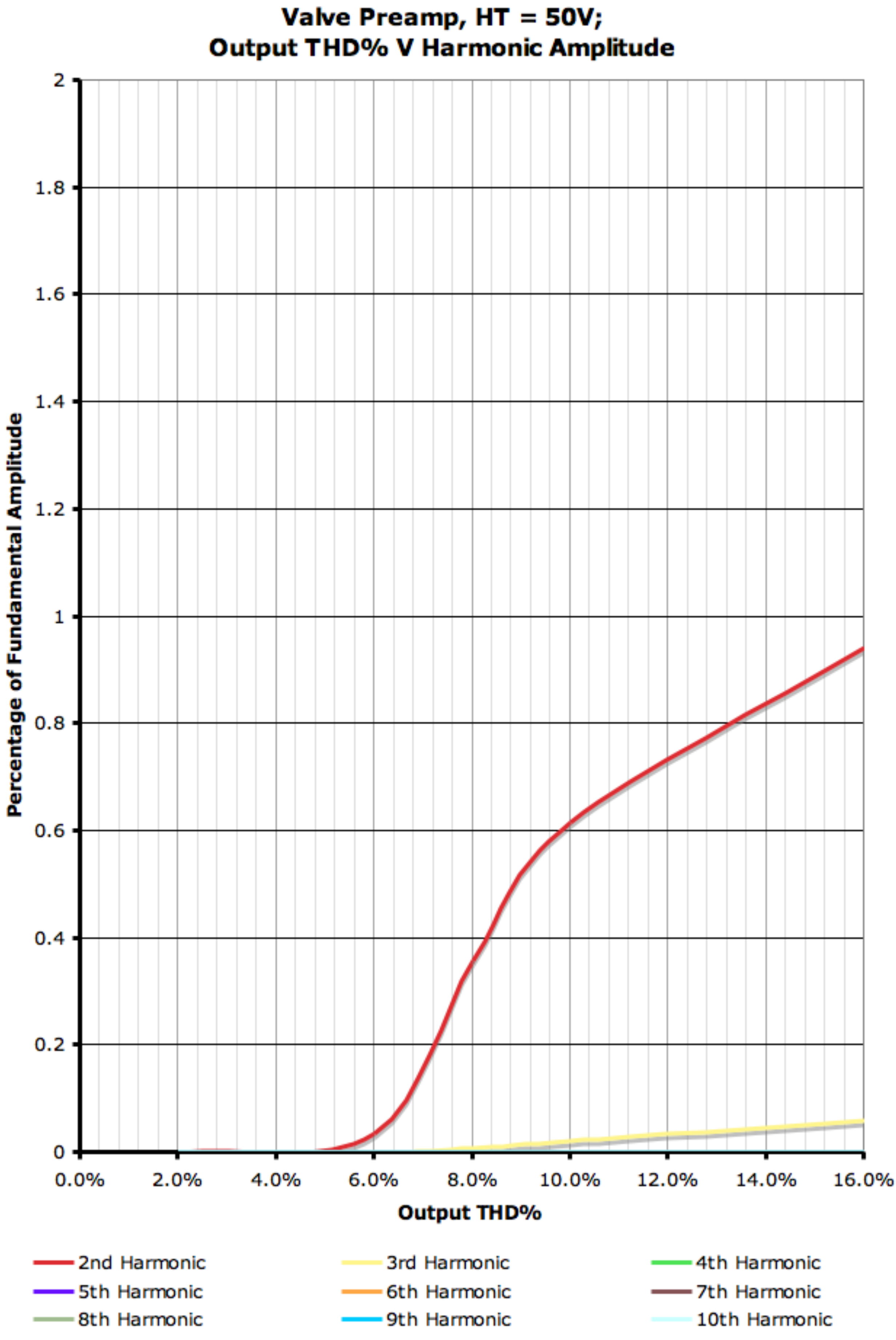



Valve Preamp, HT = 250V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
0
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
2
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
2nd Harmonic
3rd Harmonic
4th Harmonic
5th Harmonic
6th Harmonic
7th Harmonic
8th Harmonic
9th Harmonic
10th Harmonic

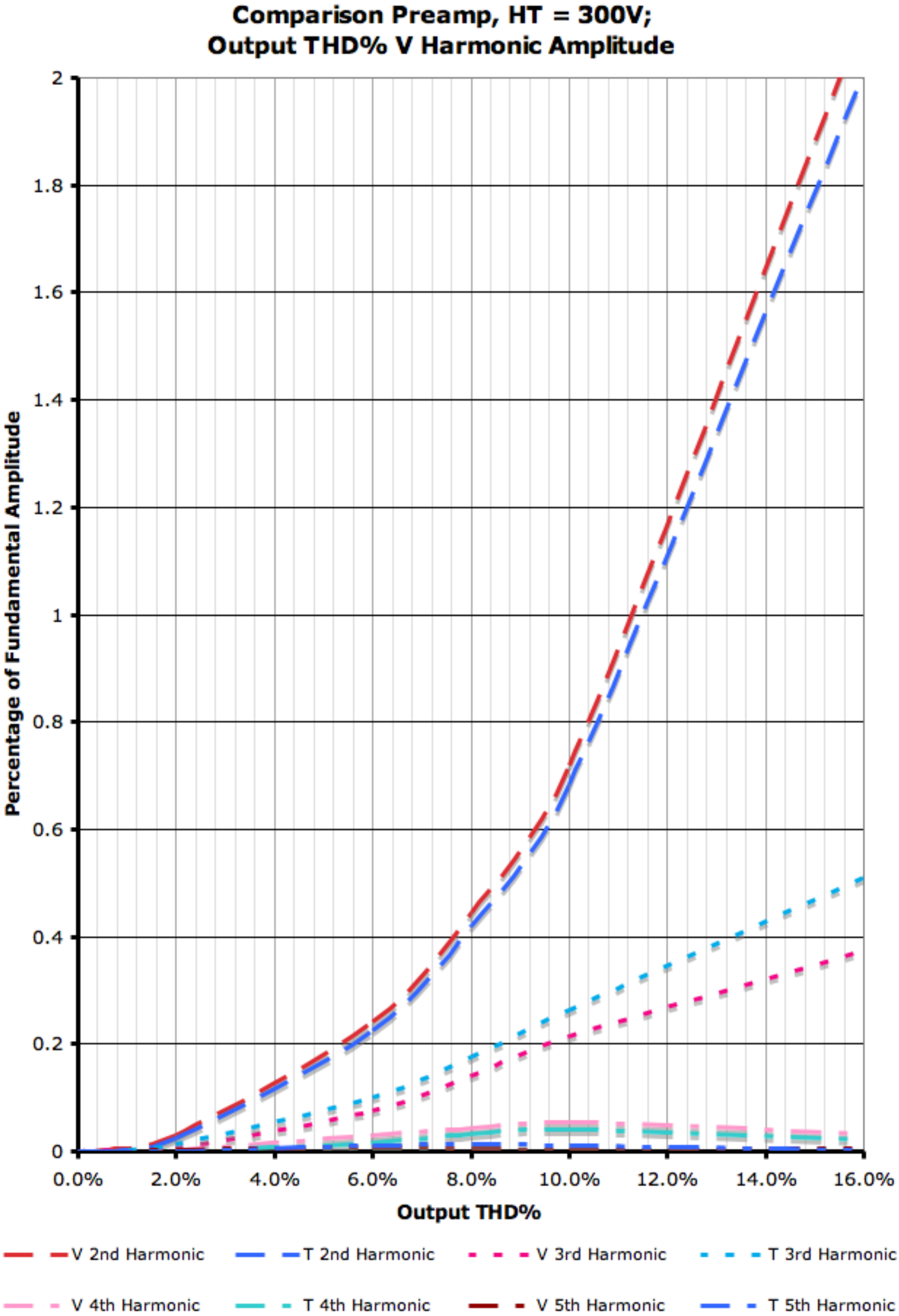

Valve Preamp, HT = 200V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
2nd Harmonic
3rd Harmonic
4th Harmonic
5th Harmonic
6th Harmonic
7th Harmonic
8th Harmonic
9th Harmonic
10th Harmonic

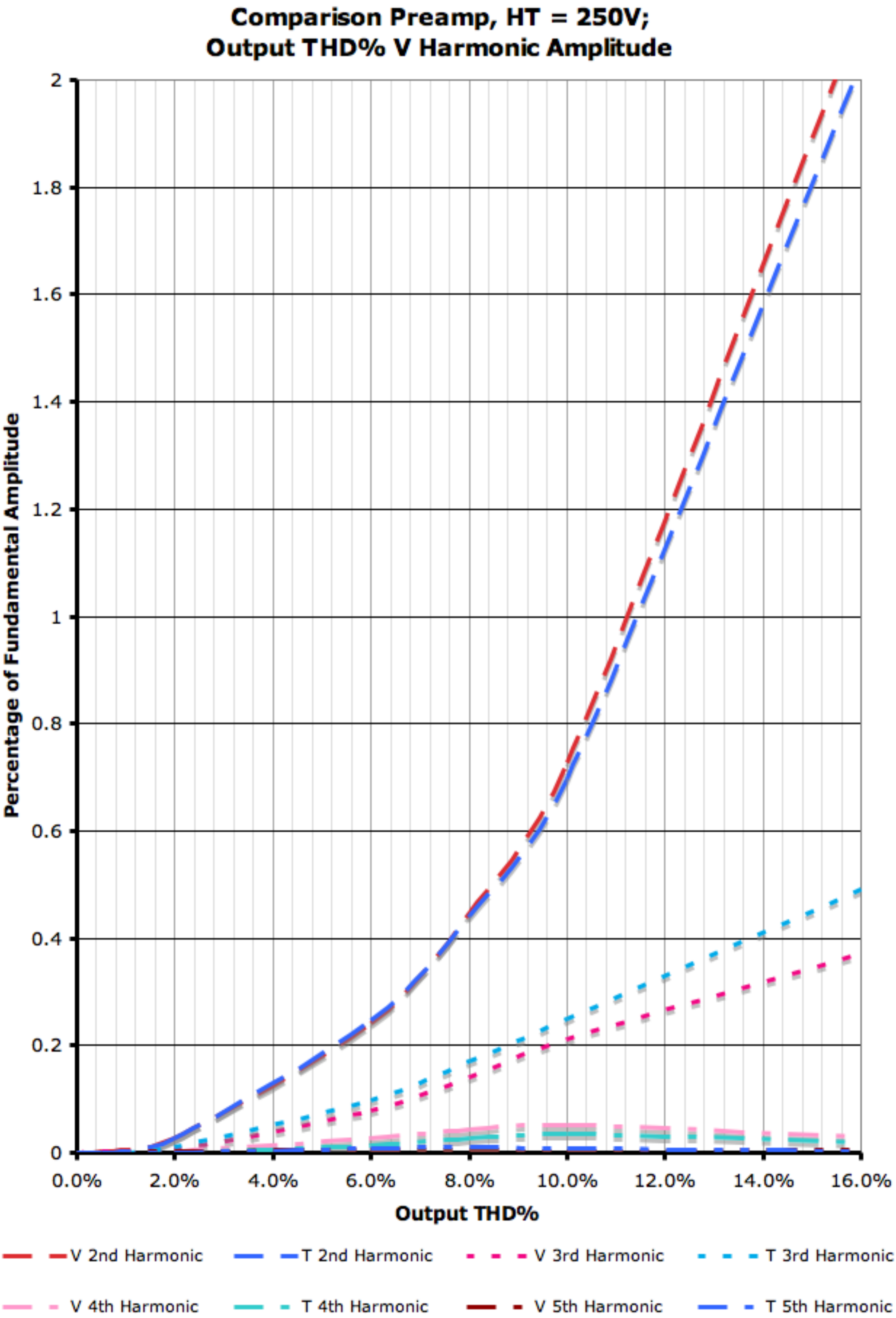

Valve Preamp, HT = 150V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
2nd Harmonic
3rd Harmonic
4th Harmonic
5th Harmonic
6th Harmonic
7th Harmonic
8th Harmonic
9th Harmonic
10th Harmonic

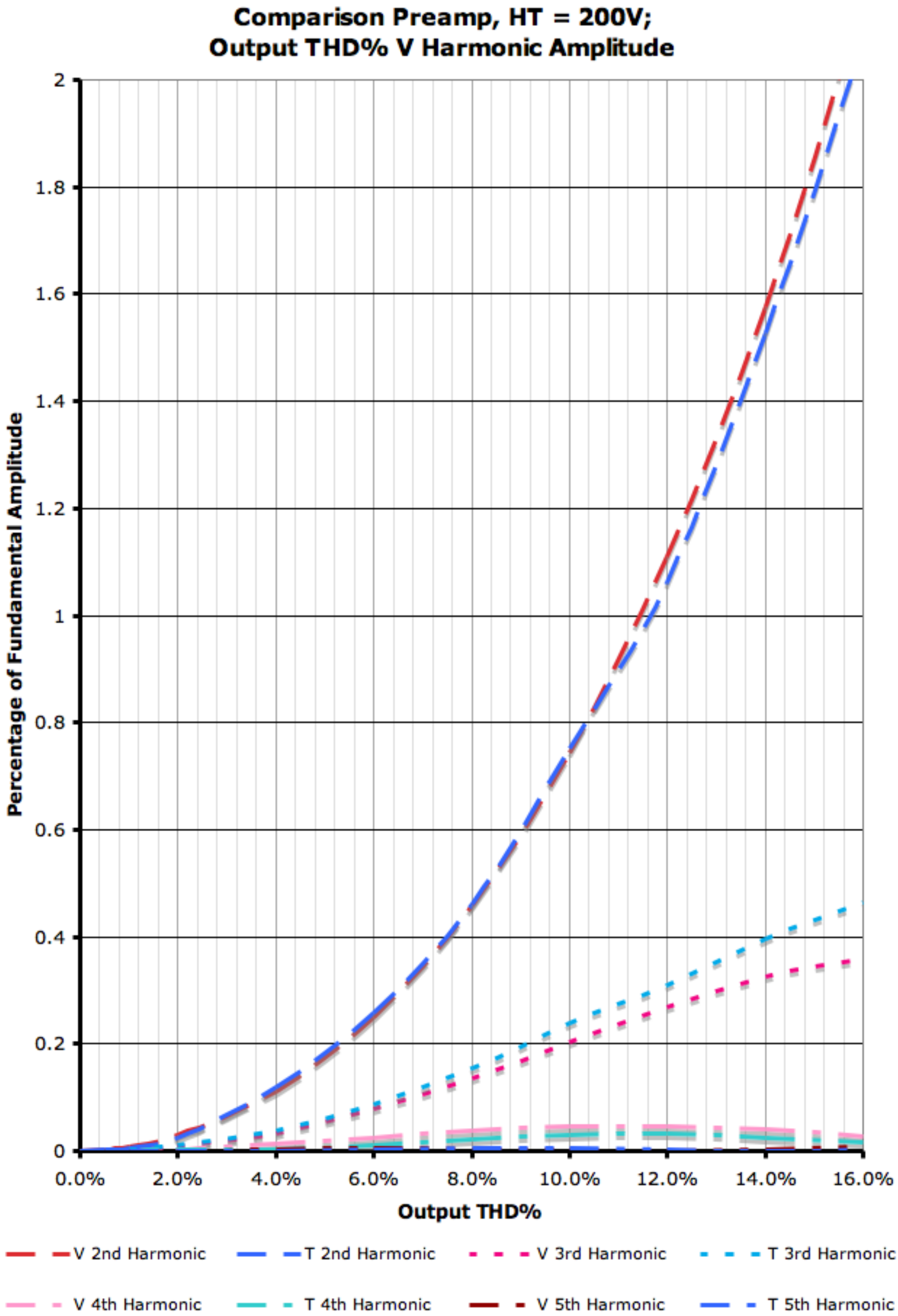

Valve Preamp, HT = 100V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
2nd Harmonic
3rd Harmonic
4th Harmonic
5th Harmonic
6th Harmonic
7th Harmonic
8th Harmonic
9th Harmonic
10th Harmonic

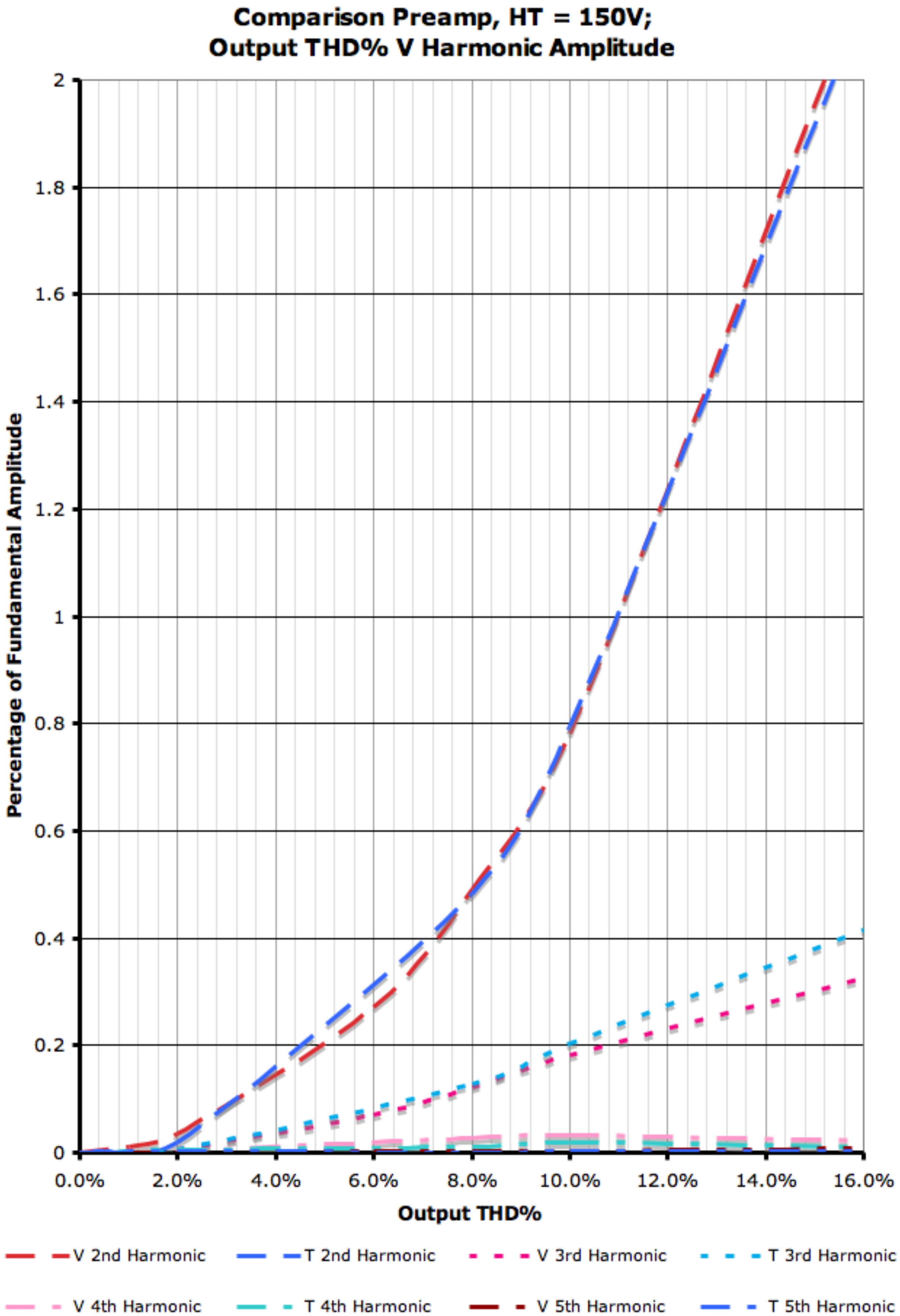

Valve Preamp, HT = 50V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
2nd Harmonic
3rd Harmonic
4th Harmonic
5th Harmonic
6th Harmonic
7th Harmonic
8th Harmonic
9th Harmonic
10th Harmonic

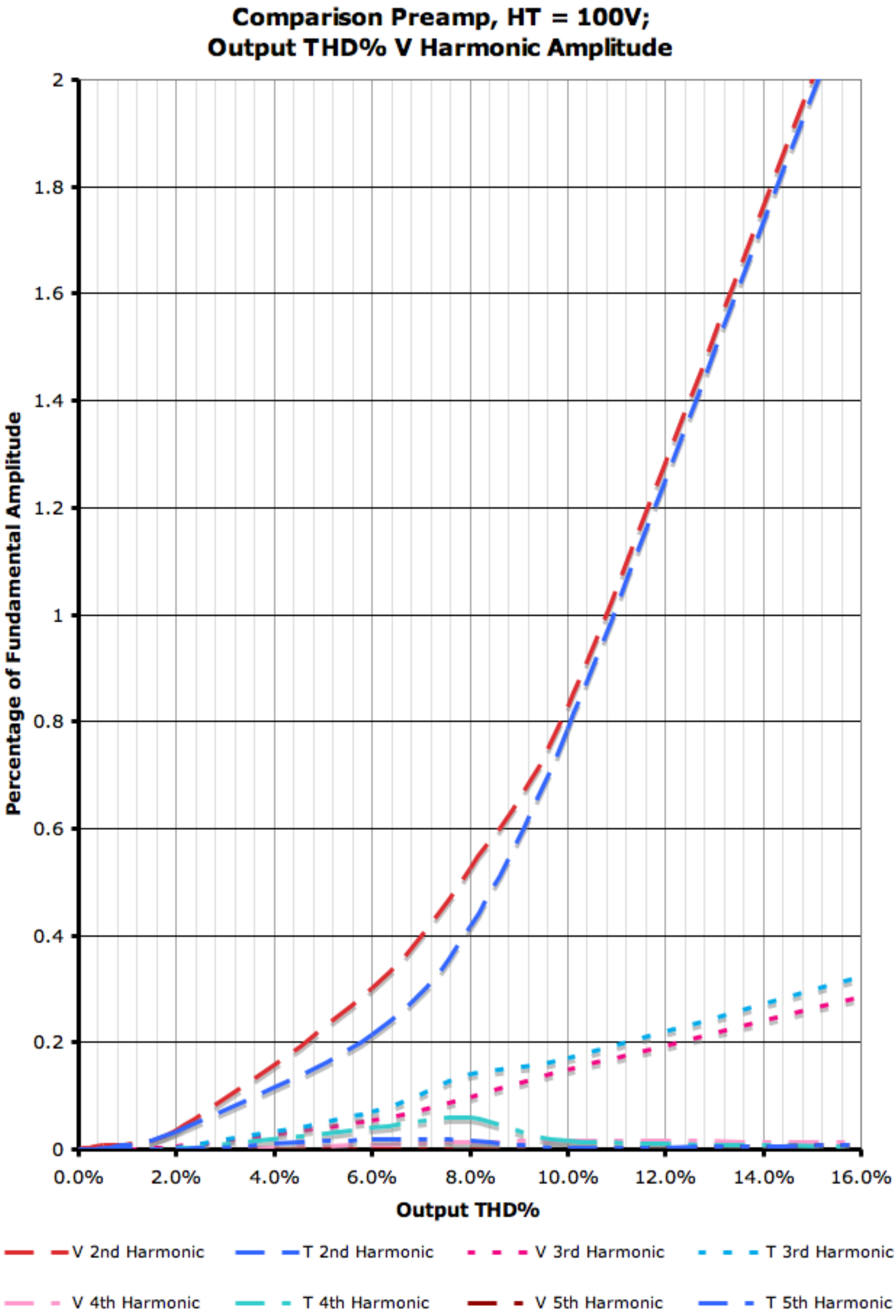
8.16 - Equivalent Hamm Plot Comparison Valve and M/F:



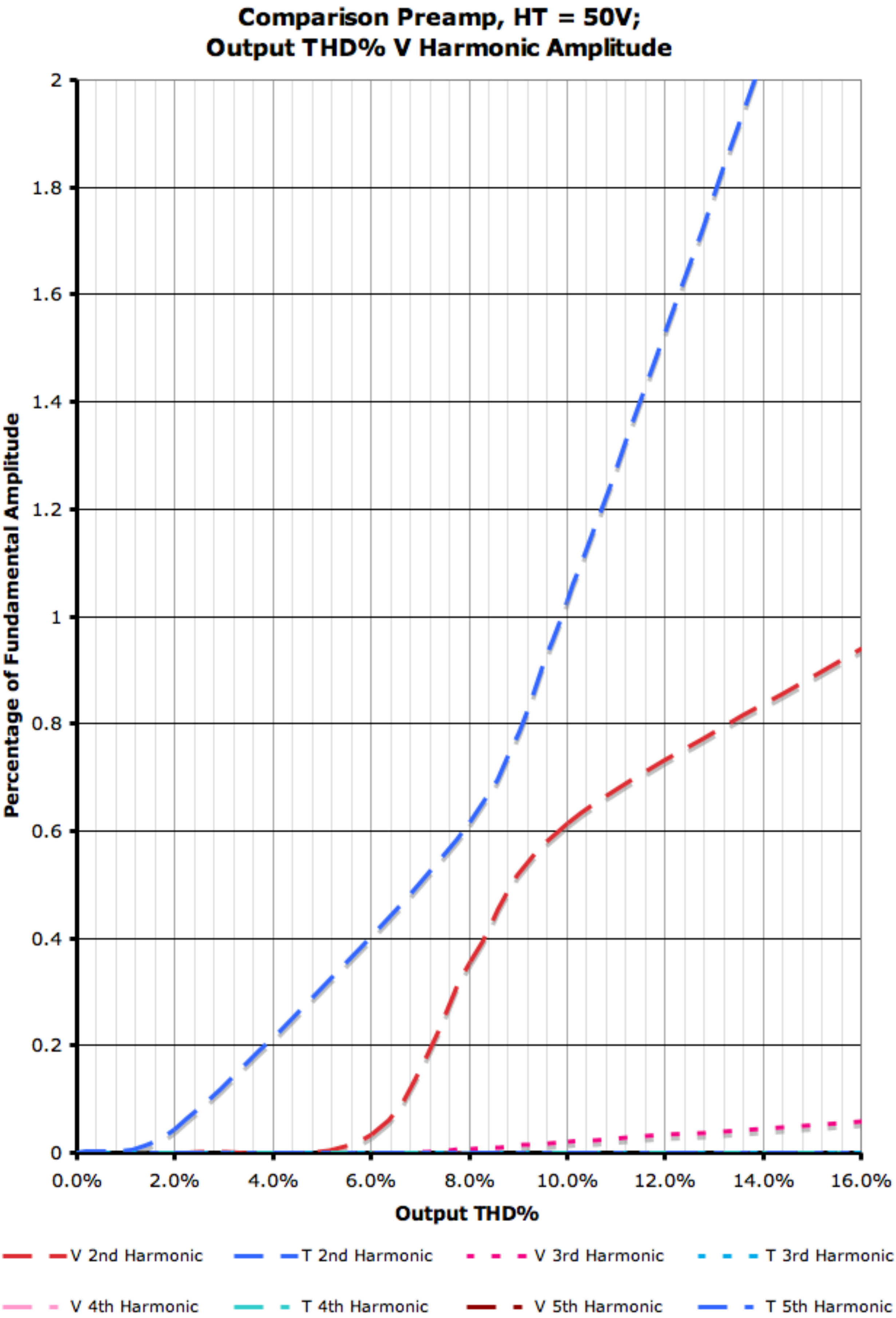

Comparison Preamp, HT = 250V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
V 2nd Harmonic
T 2nd Harmonic
V 3rd Harmonic
T 3rd Harmonic
V 4th Harmonic
T 4th Harmonic
V 5th Harmonic
T 5th Harmonic


Comparison Preamp, HT = 200V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
V 2nd Harmonic
T 2nd Harmonic
V 3rd Harmonic
T 3rd Harmonic
V 4th Harmonic
T 4th Harmonic
V 5th Harmonic
T 5th Harmonic


Comparison Preamp, HT = 150V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
V 2nd Harmonic
T 2nd Harmonic
V 3rd Harmonic
T 3rd Harmonic
V 4th Harmonic
T 4th Harmonic
V 5th Harmonic
T 5th Harmonic



Comparison Preamp, HT = 100V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
V 2nd Harmonic
T 2nd Harmonic
V 3rd Harmonic
T 3rd Harmonic
V 4th Harmonic
T 4th Harmonic
V 5th Harmonic
T 5th Harmonic


Comparison Preamp, HT = 50V;
Output THD% V Harmonic Amplitude
Percentage of Fundamental Amplitude
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0.0%
2.0%
4.0%
6.0%
8.0%
10.0%
12.0%
14.0%
16.0%
Output THD%
V 2nd Harmonic
T 2nd Harmonic
V 3rd Harmonic
T 3rd Harmonic
V 4th Harmonic
T 4th Harmonic
V 5th Harmonic
T 5th Harmonic

8.17 - HT Drop Subjective Instruction:

Distortion Characteristic Preferences

Instructions for the Undertaking of An Audio Evaluation

Thank you for participating in the audio evaluation stage of this research project. Please read through these instructions and take time to understand the research question;

“Can you identify a difference between the samples, if so, in relation to the reference, how much do you prefer the remaining samples distortion characteristics; More, less or the same?”

The audio evaluation should last on average approximately 15-20 minutes. There are eighteen experiments, each with a reference sample and a comparative sample. Before beginning the grading phase each subject is asked to familiarize themselves with the media used for testing.

Training Phase:

You will be presented with a grid of all the samples present in the ensuing evaluation. The overall aim of this training phase is to become familiar with the testing process. This should hopefully allow you as the listener to achieve two objectives:

- To become familiar with all the sound excerpts under test and their relative quality level ranges; and

- To learn how to use the testing user interface.

In this phase you will be able to listen to all the sound excerpts that have been selected for the tests in order to illustrate the whole range of possible qualities. Figure 1. Shows the training phase user interface. You may click on any buttons to play the chosen sample including the reference, however once clicked, the sample will play in full. Only once a sample has finished playing will another sample be able to be selected to enable the listener to appreciate the range of differing quality levels and differing distortion characteristics available to hear during the grading phase.

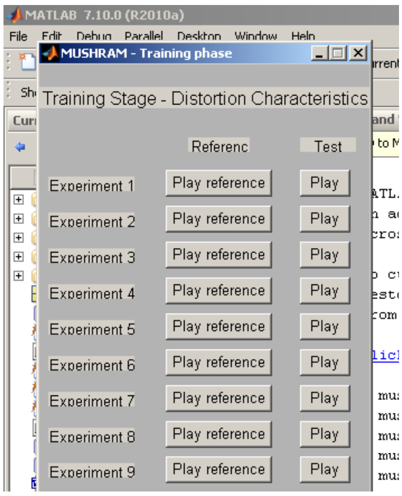


Figure 1. - The Training Phase General User Interface (GUI)

During the training phase you should be able to learn how you, as the individual, interpret the audible distortion characteristics and overall perceived sound quality, and begin to formulate an idea of personal preference. You should not discuss these preferences or interpretations with any other subjects at any time during the training phase.

It is recommended that you take a few minutes break before beginning the grading phase to prevent ear fatigue during the test.

Grading Phase:

The grading phase is the actual testing phase of the audio evaluation. You are invited to assign your grades using the quality scales provided. Said scale is a scale of preference in relation to the reference sample, positive grading being a more preferable audio quality, negative grading being a less preferable audio quality.

The audio evaluation should last on average approximately 15-20 minutes. There are eighteen experiments. In each experiment you are provided with a labelled reference sample and two unknown samples. You may listen to the samples in any order and any number of times. Unlike the training phase each sample will start anew when clicked. This should ease the process of directly comparing the samples with each other to allow grading. If at any point you wish for a few seconds silence, a "Stop All" button is provided.

You are asked to identify which of the sliders is the reference sample and score this sample as being the same (0 on the sliding scale). The sample you perceive as being different, if indeed you can perceive a difference, you are asked to preference grade using the interface.

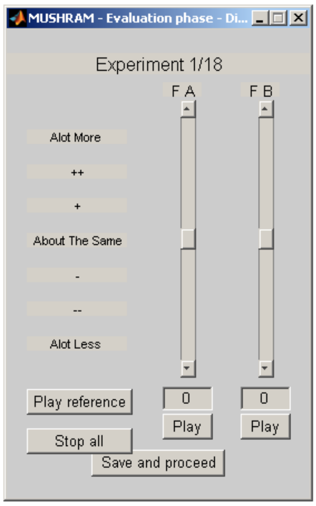


Figure 2. - The Grading Phase General User Interface (GUI)

Figure 2 shows the user interface for the grading stage of testing. Please use the slider for your identified sample to indicate your opinion of preference. The grading scale is continuous about a zero point, "About the same", from "A lot more" to "A lot less."

One sample in each experiment must be graded as 0 because the reference signal is included as one of the samples to be graded.

When you are satisfied with your grading of all the samples you should click on the "save and continue" button at the bottom of the screen to move on to the next experiment. Once all six experiments are complete please click the "save and exit" button located in the same position.