Application Note

Smart Amp Application Guide for TAS5825M

Paul Li, Jesse Ji

Consumer Audio Amplifiers and Haptics

ABSTRACT

Smart amp technology significantly enhances sound quality, maximizes peak power output, and improves system reliability through intelligent predictive algorithms. This paper provides a concise introduction and detailed implementation guideline for smart amp applications with TAS5825M such as notebooks, smart speakers, and TVs.

Smart amp fundamentals including the basic principles and modeling of typical speakers as well as the smart amp algorithms are discussed firstly. Then, this paper illustrates the necessary preparation work for implementation of the smart amp. Finally, detailed guidelines are included in this paper for both the speaker characterization and the smart amp tuning and verification, helping to facilitate the rapid implementation of smart amp with TAS5825M.

Table of Contents

Trademarks

Purepath™ is a trademark of Texas Instruments.
MATLAB® is a registered trademark of The MathWorks, Inc.
All trademarks are the property of their respective owners.

1 Introduction

With the deep understanding of the speaker characteristics and behaviors, TI's smart amp offers an advanced, intelligent and configurable processing flows to produce more powerful and richer sound than conventional systems without incurring speaker damages from either over-heat or over-excursion. To achieve a satisfying acoustic effect and robust speaker protection with TI's smart amp, both an understanding of audio and speaker fundamentals and the appropriate implementation of the speaker characterization and performance tuning are needed.

Thus, this paper aims to provide a comprehensive introduction of TI's smart amp technology and an easy-to-use guideline to facilitate the rapid implementation of smart amp with TAS5825M.

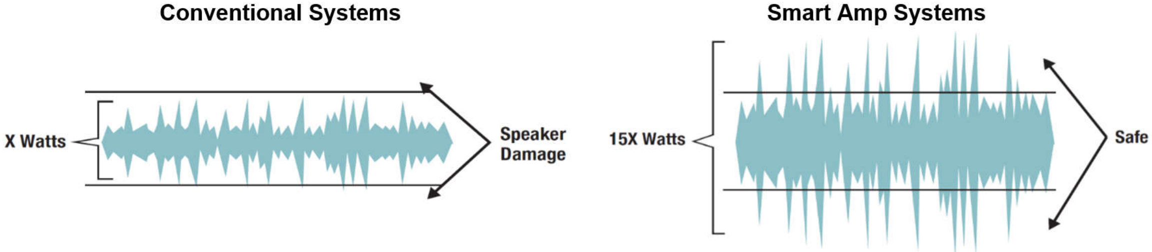


Figure 1-1. Comparison of Conventional and Smart Amp Systems

In conventional audio systems, as demonstrated in Figure 1-1, to prevent the speaker from being damaged during audio playback, the outputs of the amplifiers are typically limited within the ratings of the speaker by compressors and high-pass filters. However, the tuning results of conventional systems tend to be too conservative and lead to the poor dynamic range and bass performance in consequence.

On the contrary, based on the accurate modeling of the speaker's mechanical, electrical, and thermal properties, TI's smart amp can dynamically monitor the speaker's excursion and thermal behavior, predict the potential damage situations and take timely precautions to make sure the speaker is within the safe-operating-area (SOA) at all times. Therefore, the potential of both the speakers and amplifiers can be fully utilized to deliver stunning sound quality without causing speaker damages. In the meanwhile, transient peaks of the audio signals are no longer strictly limited or compressed, which can extend the dynamic range of the music and enhance the bass performance with far more depth and punch.

The typical implementation of TI's smart amp can be divided into two parts, namely, the speaker characterization and the smart amp tuning, and Figure 1-2 shows the general steps of smart amp application with TAS5825M.

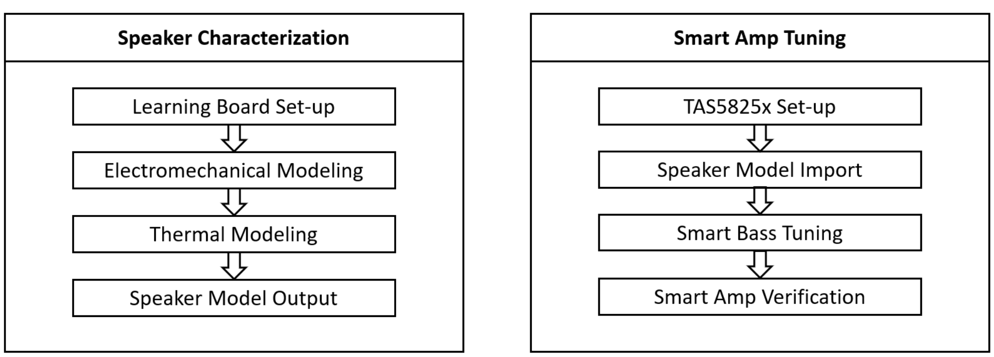


Figure 1-2. Application Steps of Smart Amp With TAS5825M

As illustrated above, the speaker characterization mainly consists of the set-up with the speaker learning board and Purepath™ Console 3 (PPC3) software, the characterization of speaker's electromechanical model and thermal model, and the output of the speaker models. The smart tuning process includes the set-up of TAS5825M and PPC3 software, the import of speaker models, the smart bass tuning and the verification of the smart amp.

2 Smart Amp Fundamentals

2.1 Speaker Basics and Models

Typical structure of the speaker can be presented in Figure 2-1. With alternating current at certain frequency applied to the voice coil, magnetic force is generated between the magnet and the voice coil, and drive the attached cone membrane (all the moving parts including the cone, dust cap, surround, and so forth) to move back and forth at the same frequency, leading to sound.

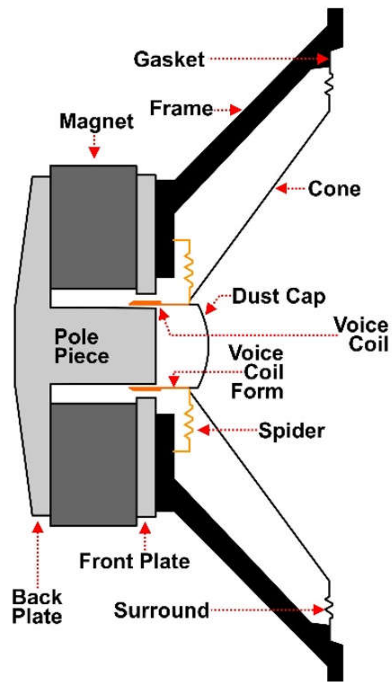


Figure 2-1. Typical Speaker Structure

To better understand and analyze the principles and behaviors of the speaker, mathematical models, including the electromechanical and thermal models of speaker have been developed. Figure 2-2 shows the linearized electromechanical model of typical speakers, and the description of the main parameters has been listed in Table 2-1.

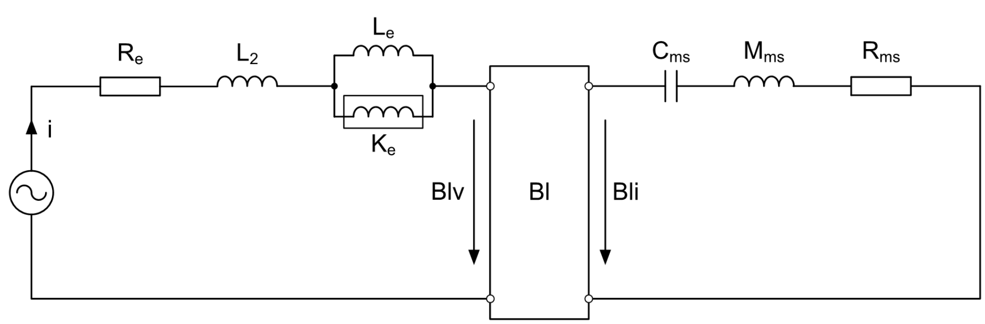


Figure 2-2. Typical Electromechanical Model of Speakers

Table 2-1. Parameters of the Electromechanical Model

Parameters	Unit	Description
R_e	Ω	DC resistance of the voice coil
S_d	cm^2	Area of the diaphragm
Bl	T·m	Force factor
R_{ms}	N·s/m	Mechanical damping factor
M_{ms}	g	Mechanical mass
C_{ms}	m/N	Mechanical compliance
L_e	mH	Leakage inductance of the voice coil
L_2	mH	Inductance of the voice coil
K_e	sH	Semi inductance of the voice coil
u	V	Input voltage
i	A	Input current
v	m/s	Velocity of the membrane
X	m	Membrane excursion

Based on the above electromechanical model, the transfer function of typical speakers can be derived. For simplicity, parasitic parameters with small values, such like L_e, L_2 and K_e can be omitted in further analysis. Therefore, the input electrical impedance of the speaker can be deduced as:

$$Z_{in}(s) = \frac{u(s)}{i(s)} = R_e + \frac{(Bl)^2}{sM_{ms} + R_{ms} + 1/sC_{ms}} \tag{1}$$

And the transfer function from input voltage to the excursion can be derived as:

$$H_{exc}(s) = \frac{X(s)}{u(s)} = \frac{Bl}{sR_e} \cdot \frac{1}{sM_{ms} + \left(R_{ms} + (Bl)^2/R_e\right) + 1/sC_{ms}} \tag{2}$$

Furthermore, the equivalent Thiele/Small (T/S) parameters of the electromechanical model of typical speakers can be derived, as listed in Table 2-2.

Table 2-2. T/S Parameters of the Electromechanical Model

Parameters	Unit	Description
F_s	Hz	Resonance frequency of the speaker
Q_{es}	–	Electrical quality factor at Fs
Q_{ts}	–	Mechanical quality factor at Fs
Q_{ms}	–	Total quality factor at Fs
V_{as}	liter	Equivalent compliance volume

$$F_s = \frac{1}{2\pi\sqrt{M_{ms}C_{ms}}} = \frac{\omega_s}{2\pi} \tag{3}$$

$$Q_{es} = \frac{R_e}{(Bl)^2}\sqrt{\frac{M_{ms}}{C_{ms}}} \tag{4}$$

$$Q_{ms} = \frac{1}{R_{ms}}\sqrt{\frac{M_{ms}}{C_{ms}}} \tag{5}$$

$$Q_{ts} = \frac{Q_{es}Q_{ms}}{Q_{es} + Q_{ms}} \tag{6}$$

$$V_{as} = 1000 \cdot \rho c^2 {S_d}^2 C_{ms} \tag{7}$$

In the table, ρ is the density of air (1.184kg/m^3 at 25 °C), and c is the speed of sound (346.1m/s at 25 °C). In this case, the transfer function of the electromechanical model can be transformed into the following equations.

Input electrical impedance:

$$Z_{in}(s) = R_e + \frac{(Bl)^2}{M_{ms}} \times \frac{s}{s^2 + s\omega_s/Q_{ms} + \omega_s{}^2} \tag{8}$$

Excursion transfer function:

$$H_{exc}(s) = \frac{Bl}{M_{ms}R_e} \times \frac{1}{s^2 + s\omega_s/Q_{ts} + \omega_s{}^2} \tag{9}$$

Similarly, the thermal behavior of speakers can also be described with the linearized mathematical model, that is, the thermal model, as shown in Figure 2-3. Table 2-3 lists the corresponding parameters of the thermal model of typical speakers.

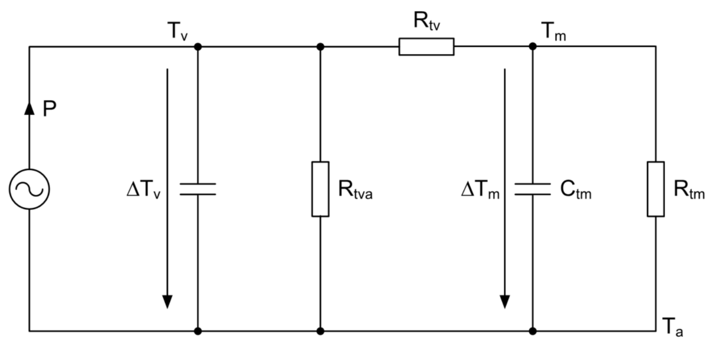


Figure 2-3. Typical Thermal Model of Speakers

Table 2-3. Parameters of the Thermal Model

Parameters	Unit	Description
R_{tv}	K/W	Thermal resistance from voice coil to magnet
C_{tv}	J/K	Thermal capacitance of voice coil
R_{tm}	K/W	Thermal resistance from magnet to ambient air
C_{tm}	J/K	Thermal capacitance of the magnet
R_{tva}	K/W	Thermal resistance from voice coil to air gap
P	W	Power dissipation on voice coil as heat
T_v	K	Voice coil temperature
T_m	K	Magnet temperature
T_a	K	Ambient temperature
ΔT_v	K	Temperature difference between voice coil and ambient
ΔT_m	K	Temperature difference between magnet and ambient

For better understanding, the relation between the dissipated power and the temperature difference in the thermal model is similar as the relation between current and voltage in electric circuits. Thus, for thermal resistance:

$$\Delta T(s) = R_{thermal} \times P_{dissipated}(s) \tag{10}$$

And for the thermal capacitance:

$$\Delta T(s) = \frac{1}{sC_{thermal}} \times P_{dissipated}(s) \tag{11}$$

Therefore, the transfer function from the dissipated power to the voice coil temperature needs to be:

$$H_{coil}(s) = \frac{\Delta T_v(s)}{P(s)} = \left(R_{tv} + \frac{1}{sC_{tm}} \parallel R_{tm}\right) \parallel \left(\frac{1}{sC_{tv}}\right) \parallel R_{tva} \tag{12}$$

Similarly, the transfer function from the dissipated power to the magnet temperature is:

$$H_{mag}(s) = \frac{\Delta T_m(s)}{P(s)} = H_{coil}(s) \times \left(\frac{1}{sC_{tm}} \parallel R_{tm}\right) \times \left(R_{tv} + \left(\frac{1}{sC_{tm}}\right) \parallel R_{tm}\right)^{-1} \tag{13}$$

2.2 Smart Amp Algorithm

Feed-forward smart amp protection algorithm has been integrated into TAS5825M for both the excursion and thermal protection of target speakers, which can be presented as Figure 2-4.

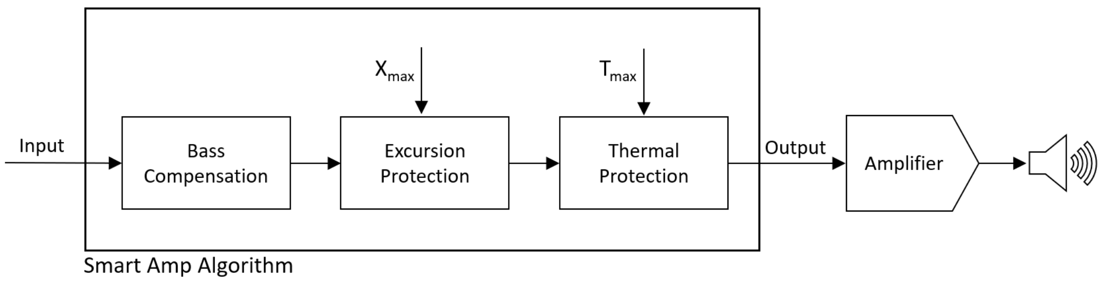


Figure 2-4. Smart Amp Algorithm

For excursion protection, a look ahead structure can be applied to make sure the excursion estimation and signal limitation have been completed before the signal is fed to the algorithm output. Figure 2-5 shows the block diagram of the excursion protection algorithm.

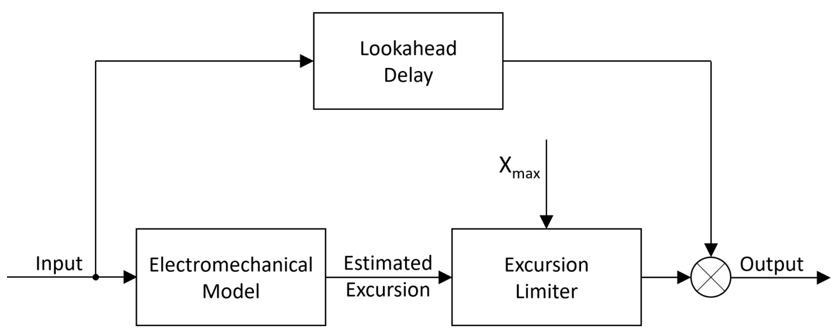


Figure 2-5. Smart Amp Excursion Protection Algorithm

As demonstrated in Figure 2-5, the excursion of the speaker's membrane is estimated firstly with the convolution operation between the audio signals and the derived excursion transfer functions:

$$X(t) = u(t) \times \mathcal{L}^{-1}[H_{exc}(s)] \tag{14}$$

Then the excursion is compared with the maximum excursion limit before deciding if protection kicks on. Once the estimated excursion exceeds the limit X_{max}, the input signal is attenuated to realize the speaker protection, otherwise the input signal passes through unchanged.

Figure 2-6 shows the block diagram of the smart amp thermal protection algorithm. A real-time processing structure is adopted here considering the slow response characteristic of the thermal system.

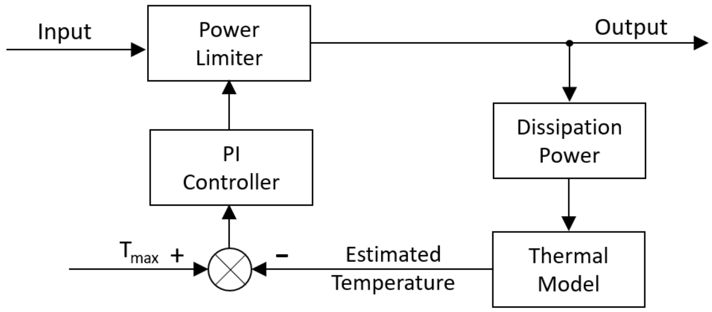


Figure 2-6. Smart Amp Thermal Protection Algorithm

Firstly, the temperature of the voice coil is estimated with the dissipated power and the thermal model of the speaker, given by:

$$T_v(t) = T_a(t) + P(t) \times \mathcal{L}^{-1}[H_{coil}(s)] \tag{15}$$

Then the estimated temperature is compared with the temperature limit of the voice coil, and generate the reference signal to the PI thermal controller. The output of the thermal controller is then sent into the power limit module to attenuate the audio signal when necessary, which keeps the voice coil temperature within the thermal limit.

3 Preparation Work

To correctly implement the speaker characterization and tuning with the smart amp devices, use the following for details to make the necessary preparations.

3.1 Hardware Preparation

The required hardware for the characterization of speakers includes:

- Smart amp learning board (PP-SALB-EVM-XMOS)
- Target speaker with speaker box
- DC power supply (support 24V/3A output)
- USB cable (micro-USB to type-A)
- Multimeter
- Microphone
- Adhesive putty
- Weight scale
- Connection wires

The required hardware for smart amp tuning and verification:

- Development board with TAS5825M
- Target speaker with speaker box
- DC power supply (support 24V/3A output)
- USB cable (micro-USB to type-A)
- Multimeter
- Laser equipment
- Audio precision equipment

3.2 Software Preparation

The following software is required to run smart amp with TAS5825M:

- PurePath™ Console 3:
  - Learning Board module for PP-SALB-EVM-XMOS
  - TAS5825M module
- MATLAB® Compiler Runtime:
  - Installed with PurePath™ Console 3

3.3 Speaker Information

Knowing the speaker parameters is important and also making sure of the accuracy when starting an audio design with smart amp, and the suggestion is to request the following details in the data sheet from the speaker vendor:

- Diaphragm area:
  - Area or diameter of the speaker diaphragm
- Excursion limit:
  - X_{max}: maximum linear excursion of the speaker diaphragm/coil
- Thermal limit:
  - T_{max}: maximum temperature of speaker coil
  - P_{max}: maximum peak power of the speaker
- Thiele/Small (T/S) parameters:
  - Having the T/S parameters of the target speakers for validation is recommended.

4 Speaker Characterization

4.1 Characterization Set-up

The hardware set-up for speaker characterization with learning board (PP-SALB-EVM-XMOS) can be described as in Figure 4-1.

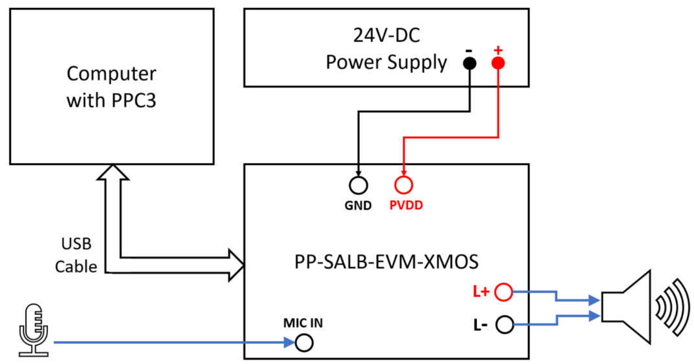


Figure 4-1. Hardware Set-up for Smart Amp Learning Board

4.2 Characterization Process

Figure 4-2 shows the speaker characterization process with the learning board.

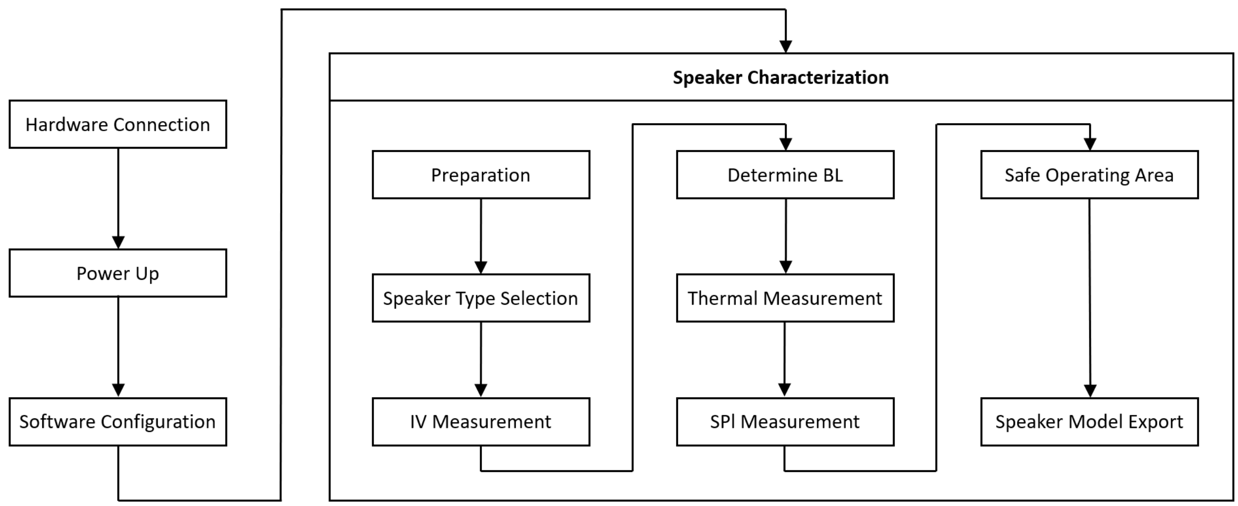


Figure 4-2. Speaker Characterization Process

4.3 Speaker Characterization Guide

4.3.1 Hardware Connection

Turn off the DC power supply, and make proper hardware connections between the learning board, the computer, and the DC power supply according to the hardware set-up shown in Figure 4-1. The speaker can be left disconnected in this step.

4.3.2 Power Up

Set the output voltage of the DC supply to 24V/3A, and turn on the power supply.

4.3.3 Software Configuration

- Open the sound settings of Windows, and make sure to choose the correct output and input devices in the sound settings:
  - Output: Speakers (2-TI USB Audio UAC2.0)

  - Input: Line (2-TI USB Audio UAC2.0)
- Check the volume levels of both output and input to be set at 100%.
- Open the additional devices properties of both output and input devices, and check the sampling rate of the devices in the *Advanced* item. The correct configuration needs to be as shown in Figure 4-3.
- Open the PPC3 software, and enter the *Learning Board* module.
- Click *New* to create a new file, and then click *Connect* button in the left bottom side to connect PPC3 with the learning board, shown as in Figure 4-4.

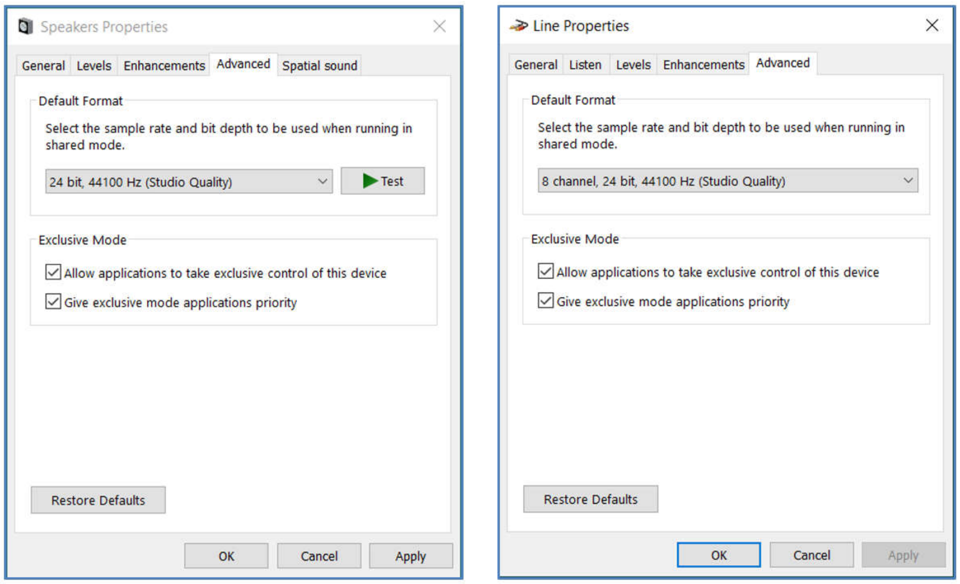


Figure 4-3. Sound Settings for Learning Board

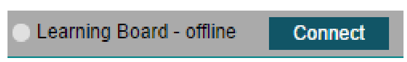


Figure 4-4. PPC3 Connect Button

4.3.4 Speaker Characterization

Click the *Characterization* module on the PPC3 page, as shown in Figure 4-5.

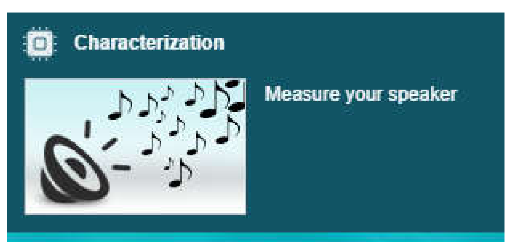


Figure 4-5. Characterization Module in PPC3

4.3.4.1 Preparation

Double check to make sure the DC power supply is fine, and click the *Supply is Connected* button in the *Hardware Setup* page. And then click the *Start Checks* button to start the hardware check. Once completing the hardware check correctly, PPC3 needs to show the check results as in Figure 4-6.

Next, in the *System Gain Calibration* page, click the *Start Gain Calibration* button to start the system gain calibration process. Use the DC mode of the multimeter to measure the supply DC voltage over PVDD and GND, and use the AC mode of the multimeter to measure the output AC voltage (RMS) across the L+ and L- pins of the learning board.

Type the measurement results into the corresponding boxes in the *System Gain Calibration* page, and click the *Next* button to enter the next step.

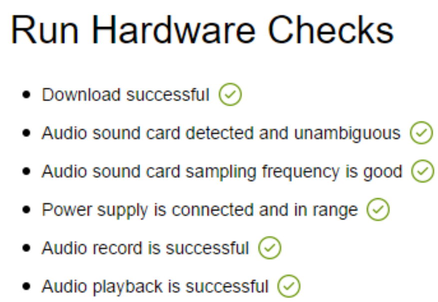


Figure 4-6. Hardware Check Results

4.3.4.2 Speaker Type Selection

On the *Choose your Speaker Type* page, choose the correct type according to the properties of your target speakers. There are three types of the speakers can be selected: closed box, ported box, passive radiator. For example, for a single speaker with a ported box, you can choose the *Single Driver Ported Box* as shown in Figure 4-7. Noted that sometimes passive radiator speaker cannot be modeled successfully, in this situation you can need to choose ported box to replace passive radiator speaker model.

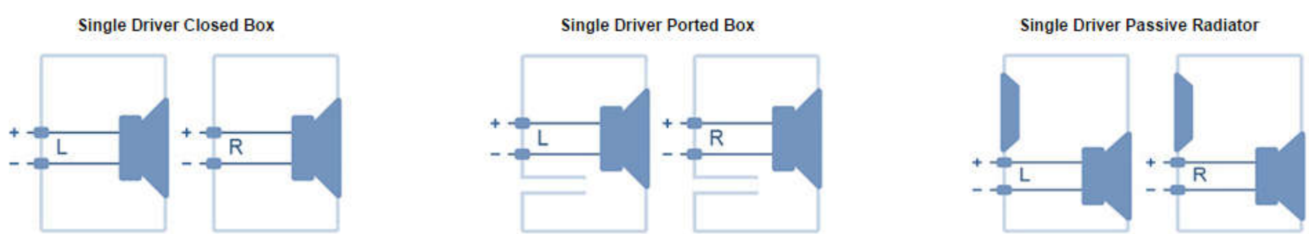


Figure 4-7. Speaker Type Selection

4.3.4.3 IV Measurement

Connect your target speaker to the L+ and L- pins of the learning board, and make sure your speaker has been held stabilized (possibly with some adhesive putty). Then click the *Start IV Measurement* button to start the IV measurement process. Be aware that a certain level of noise can be heard during the measurement process.

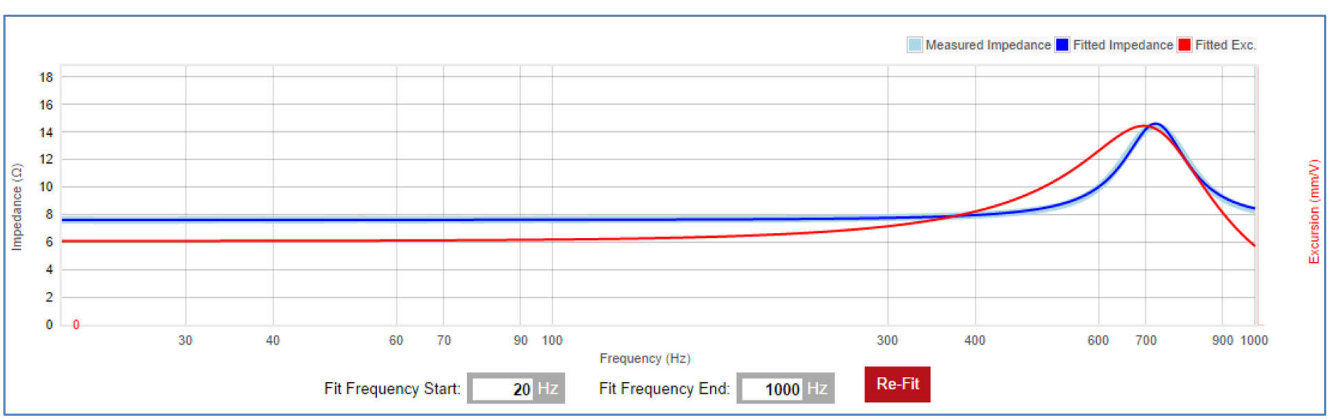


Figure 4-8. Example of Model Fitting Results

Wait until the *Model Fit* process is completed, and observe the fitting results on the *Review Speaker Model* page, and the light blue curve (measured impedance) needs to align well with the dark blue curve (fitted impedance), like shown in Figure 4-8. If so, you can click the *Accept* button to move to the next step. If not, you need to check your hardware and software configuration and click the *Re-run IV Measurement* button to re-run the IV measurement process.

4.3.4.4 Determine BL

On the *Speaker Details* page, type in the area or diameter of your speaker's diaphragm, as described in Section 2, and click the *Next* button.

On the *Force Factor (BL)* page, the recommendation is to use the *Added Mass Method*, rather than the *Enter BL Method*, to acquire the BL parameters. Click the *Added Mass Method* button to continue the characterization.

According to the instruction on *Added Mass Specification* page, get some adhesive putty as the *mass*, and use the weight scale to get the accurate weight of the mass, and type in the measured value. The suggestion is to make the mass to be approximately 20% of the speaker moving mass. If the moving mass is not given, use a putty ball that is roughly 1/5 the diameter of the speaker.

Add the mass to the target diaphragm of the speaker as described in the *Added Mass Specification* page, as shown in Figure 4-9, and click the *Mass Added* button to start the BL measurement process.

Figure 4-9. Examples of Added Mass Method

Wait until the *Added Mass IV Measurement* and the *Find BL by Added Mass* steps are completed, then remove the mass and click the *Next* button on the *Remove Added Mass* page.

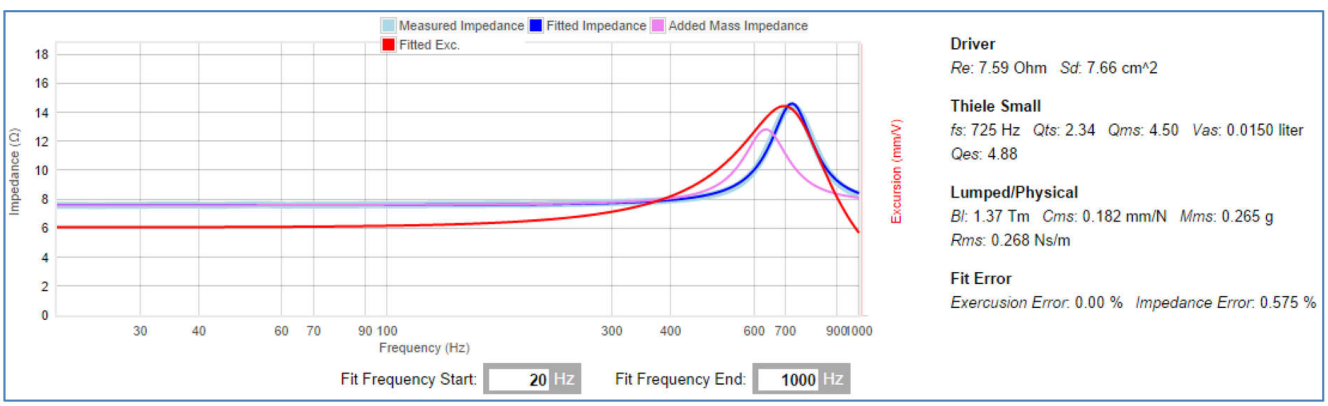


Figure 4-10. Electromechanical Model Fitting Results

Wait until the *Run Model Fit* is completed, please review the fitted models on the *Review Speaker Model* page, as shown in Figure 4-10.

First, make sure that the light blue curve (measured impedance) aligns well with the dark blue curve (fitted impedance).

Next, compare the *Re* and *fs* value of the modeling results with the Thiele/Small parameters of your speakers. There cannot be a huge gap between the modeled and the rated values.

Finally, check the *Impedance Error in Fit Error* part, and the suggestion is to keep the impedance error less than 1%. If the model fits well, click the *Accept* button to move on to the next step. If not, you're suggested to check your hardware and software configuration, check the added mass and the related data, and click the button to redo the BL measurement.

4.3.4.5 Thermal Measurement

On the *Thermal Parameters* page, read through the text and click the *Thermal Characterization* button and move to the next page.

Next, input the excitation frequency according to the suggestions on the *Thermal Characterization: Caution* page. For example, as shown in Figure 4-11, input *995Hz* as the excitation frequency.

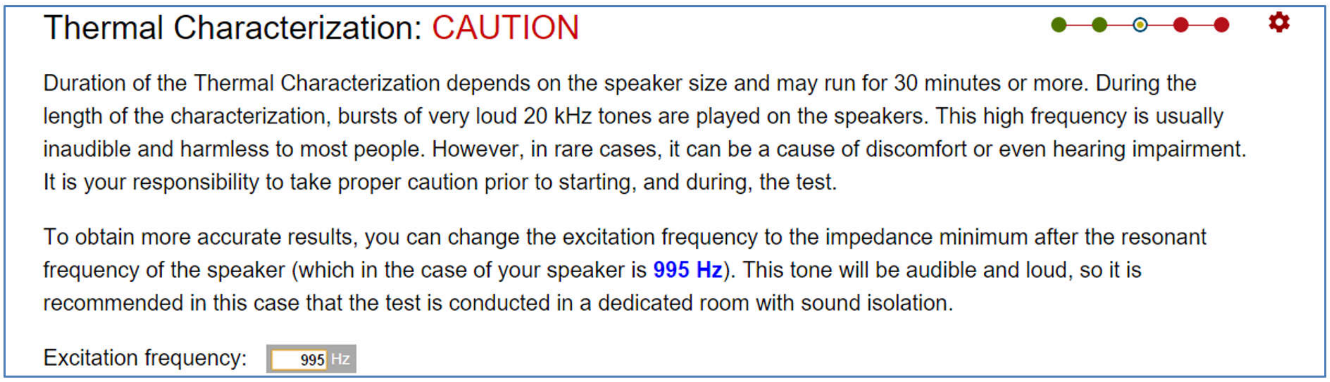


Figure 4-11. Input the Excitation Frequency

Then, click the *Next* button to start the thermal measurement process. The whole process can take about 10 to 20 minutes.

Once the thermal measurement process is completed, review the thermal model of the target speaker on the *Review Temperature Model* page.

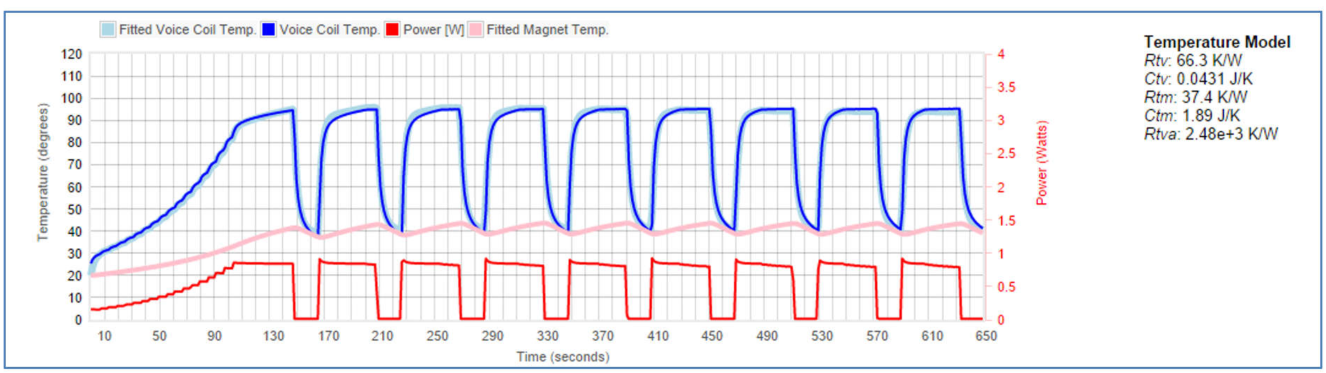


Figure 4-12. Speaker Thermal Modeling Results

As shown in Figure 4-12, the light blue curve (voice coil temp.) needs to align well with the dark blue curve (fitted voice coil temp.), and you can click the *Accept* button to finish the thermal modeling of the target speaker.

Otherwise, if the two curves of the coil temperature cannot match well, or the thermal measurement process reports failure in the PPC3 software, the suggestion is to check your hardware and software configuration, and wait until the speaker's voice coil to cool down (for about 20 minutes). Then, please click the *Run Temperature Characterization* button to re-run the temperature characterization.

4.3.4.6 SPL Measurement

To measure the SPL characteristic of the speaker, connect a microphone to the learning board and make the SPL measurement according to the instructions on the *Speaker Acoustic Response* page.

Figure 4-13 shows the typical results of the SPL characterization on *Review SPL Model* page. Check the SPL results with the information from the data sheet of the target speaker, and click the *Accept* button to continue the characterization.

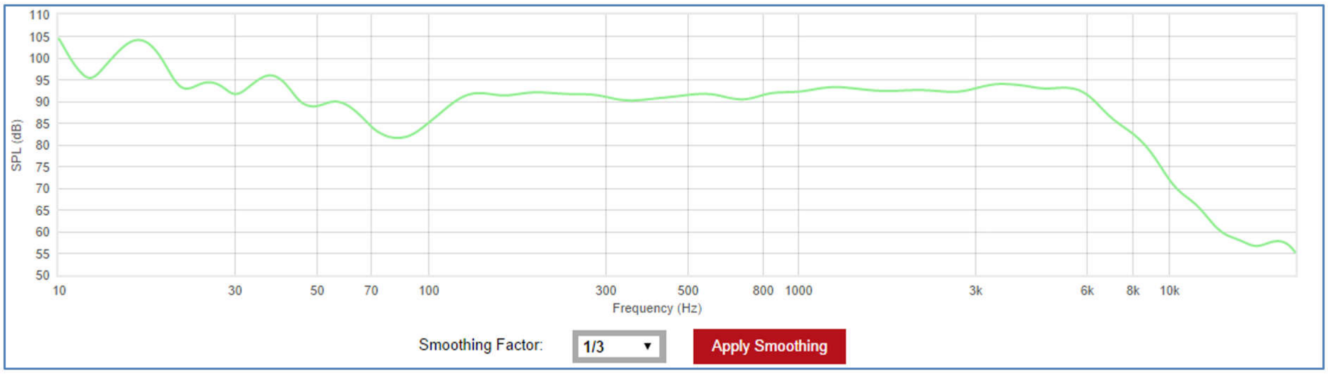


Figure 4-13. Typical SPL Modeling Results

4.3.4.7 Safe Operating Area

On the *Safe Operating Area* page, as shown in Figure 4-14, input the peak excursion limit, the thermal limit or power limit according to the data described in previous sections. The thermal limit for temperature data is set at $T_{max} – T_{ambient}$.

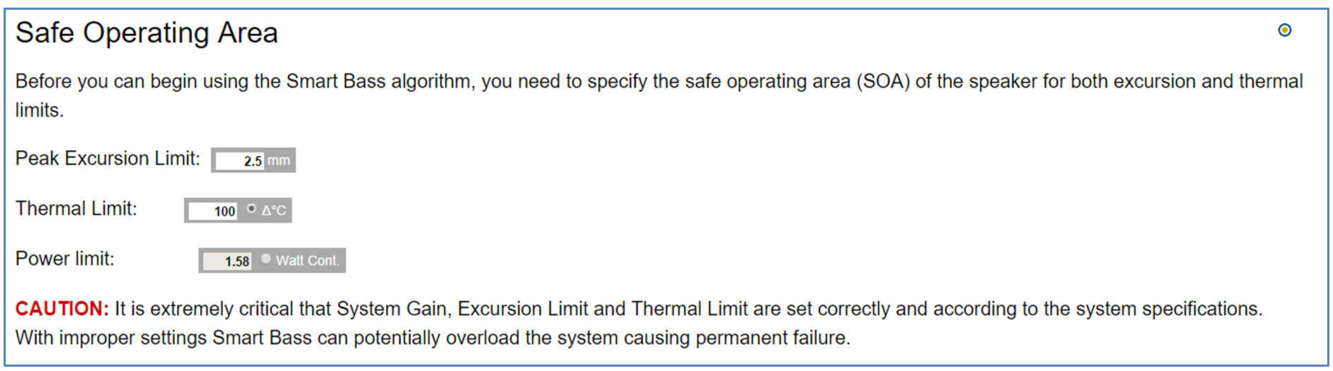


Figure 4-14. Safe Operating Area Page

4.3.4.8 Speaker Model Export

By clicking the *Accept* button on the *Safe Operating Area* page, the characterization of the speaker has been completed.

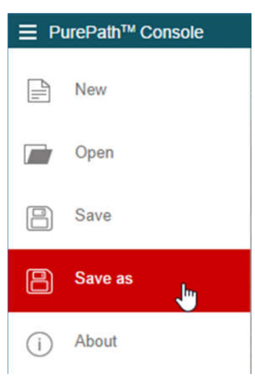


Figure 4-15. Export the Speaker Model With PPC3

To export the speaker model, click the button at the upper left corner of PPC3, and click the *Save As* to export the ppc3 file, as shown in Figure 4-15. And when you want to use the speaker model, this can be convenient to import the characterization data with the saved ppc3 file.

5 Smart Amp Tuning and Verification

5.1 Smart Amp Tuning Guide

TAS5825M has a powerful DSP audio processing core, which supports several different audio processing flows. The built-in smart amp processing flows can help to achieve significant improvements in peak power output, loudness, and sound quality relative to conventional amplifiers. The PPC3 software GUI can help developers to understand how speakers are performed in the system and then make adjustments to improve the audio performance. The algorithm, characterization, and tuning tools allow developers to overcome a wide variety of audio challenges. The smart amp tuning process is illustrated below:

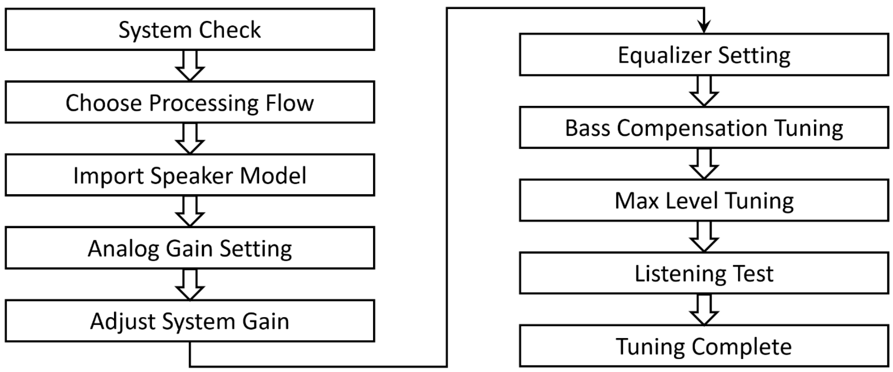


Figure 5-1. TAS5825M Smart Amp Tuning Process

5.1.1 System Check

Before tuning, click *System Check* to detect whether the whole system can work normally and calibrate the system gain as Figure 5-2 shows, which can influence the accuracy of smart amp algorithm.

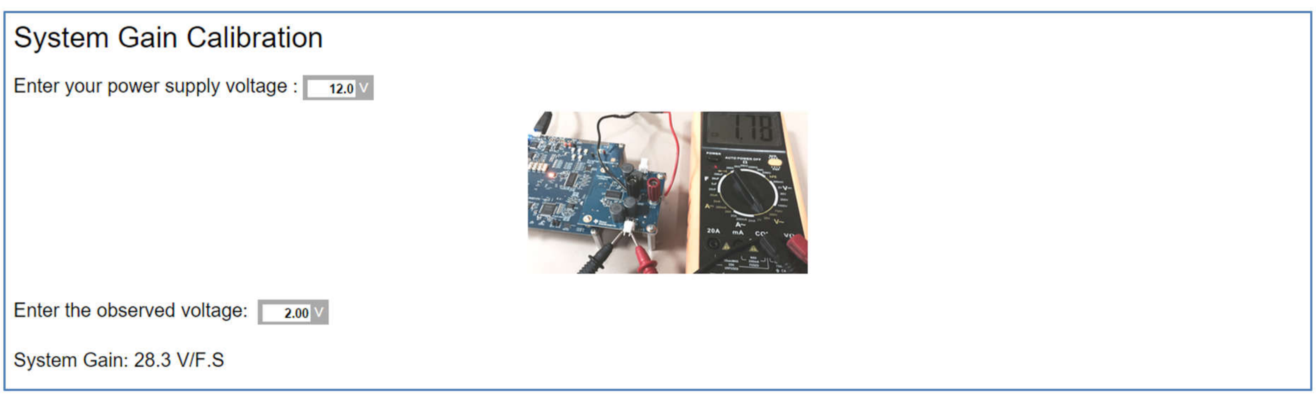
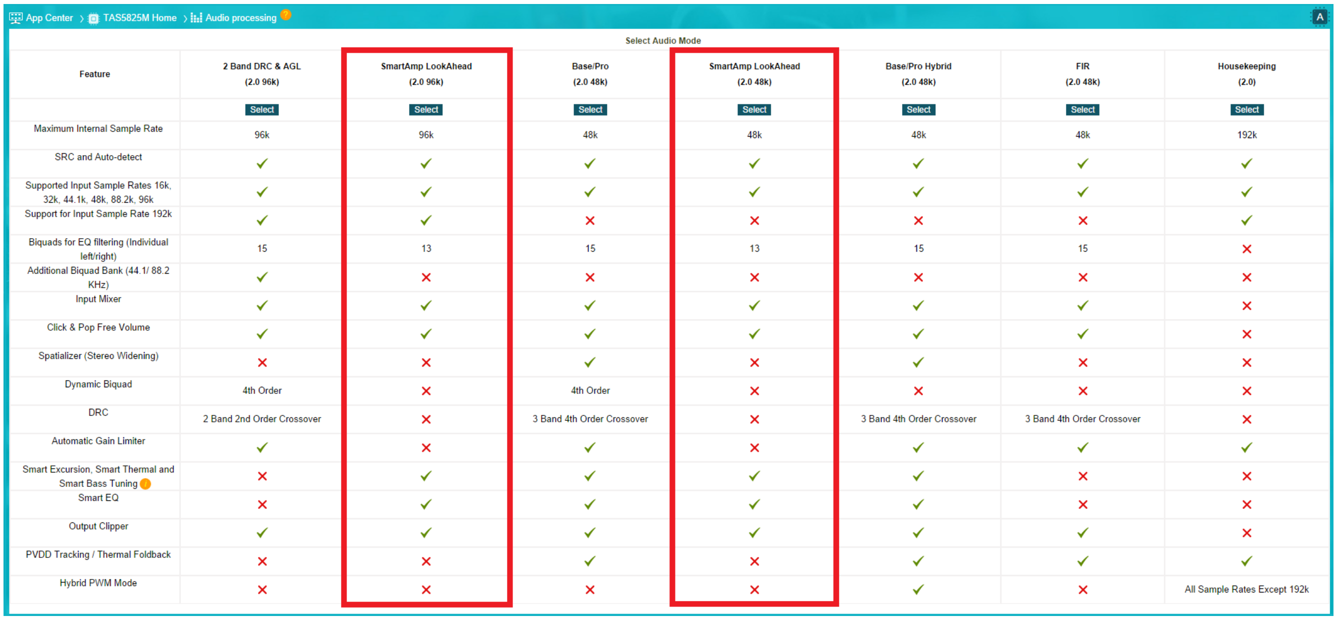


Figure 5-2. System Check

5.1.2 Choose Processing Flow

TAS5825M offers many kinds of DSP processing flows, and the recommendation is to use the smart amp look ahead processing flows for smart amp application, as shown in Figure 5-3. The algorithm uses a look ahead structure which delays 128 samples and gives the algorithm enough time to calculate. The most difference between two smart amp processing flows is the internal DSP sample rate. For example, for 48K smart amp look ahead processing flow, the look ahead delay time is:

$$LookAhead\_time = 128 \times \frac{1}{48k} \approx 2.67ms \tag{16}$$



Feature	2 Band DRC & AGL (2.0 96k)	SmartAmp LookAhead (2.0 96k)	Base/Pro (2.0 48k)	SmartAmp LookAhead (2.0 48k)	Base/Pro Hybrid (2.0 48k)	FIR (2.0 48k)	Housekeeping (2.0)
	Select	Select	Select	Select	Select	Select	Select
Maximum Internal Sample Rate	96k	96k	48k	48k	48k	48k	192k
SRC and Auto-detect	✓	✓	✓	✓	✓	✓	✓
Supported Input Sample Rates 16k, 32k, 44.1k, 48k, 88.2k, 96k	✓	✓	✓	✓	✓	✓	✓
Support for Input Sample Rate 192k	✓	✓	✗	✗	✗	✗	✓
Biquads for EQ filtering (Individual left/right)	15	13	15	13	15	15	✗
Additional Biquad Bank (44.1/ 88.2 KHz)	✓	✗	✗	✗	✗	✗	✗
Input Mixer	✓	✓	✓	✓	✓	✓	✗
Click & Pop Free Volume	✓	✓	✓	✓	✓	✓	✗
Spatializer (Stereo Widening)	✗	✗	✓	✗	✓	✗	✗
Dynamic Biquad	4th Order	✗	4th Order	✗	✗	✗	✗
DRC	2 Band 2nd Order Crossover	✗	3 Band 4th Order Crossover	✗	3 Band 4th Order Crossover	3 Band 4th Order Crossover	✗
Automatic Gain Limiter	✓	✗	✓	✗	✓	✓	✓
Smart Excursion, Smart Thermal and Smart Bass Tuning	✗	✓	✓	✓	✓	✗	✗
Smart EQ	✗	✓	✓	✓	✓	✗	✗
Output Clipper	✓	✓	✓	✓	✓	✓	✗
PVDD Tracking / Thermal Foldback	✗	✗	✓	✗	✓	✓	✓
Hybrid PWM Mode	✗	✗	✗	✗	✓	✗	All Sample Rates Except 192k

Figure 5-3. TAS5825M Processing Flow

And for 96K smart amp look ahead processing flow, the signal delay is 1.33ms, which leads lower delay comparing to 48K processing flow. Customer needs to choose correct processing flow based on system audio signal sample rate.

5.1.3 Import Speaker Model

After system checks and processing flow selection, enter *Tuning and Audio Processing* view.

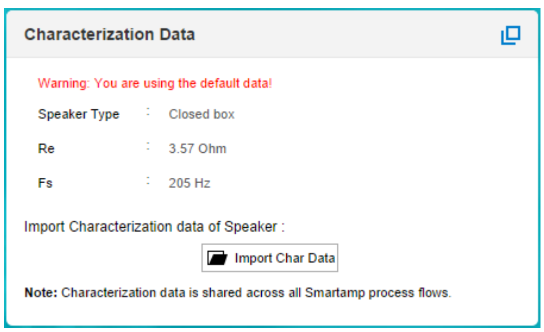


Figure 5-4. Characterization Data Block

As shown Figure 5-4 is the *Characterization Data* block, in which you can *Import Char Data* modeled by learning board, or click the button in the upper right corner to manually input your speaker key parameters. TI suggest to import the speaker model captured with TI learning board.

5.1.4 Analog Gain Setting

Analog gain of TAS5825M needs to be revised according to actual PVDD supply voltage, which can avoid analog PVDD clipping. For more information, please refer to the application note, *General Tuning Guide for TAS58xx Family*. Please use the following formulas to calculate designed for analog gain:

$$Analog\_gain \approx 20 \times \log_{10}\left(\frac{V_{speaker\_max}}{29.5V}\right)dB \tag{17}$$

$$V_{speaker\_max} \approx PVDD \times \frac{R_{speaker}}{R_{speaker} + 2 \times \left(R_{ds\_on} + Z_{DC}\right)} \tag{18}$$

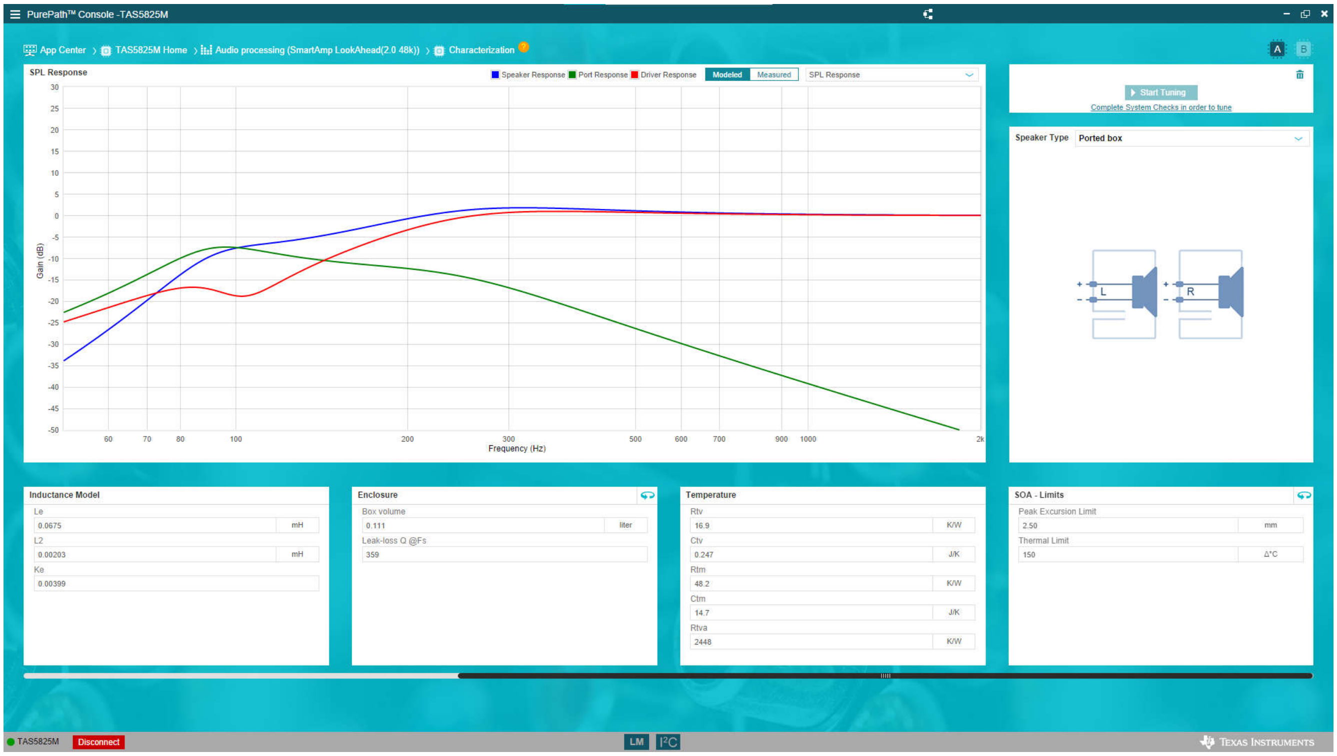


Figure 5-5. Characterization View

For example, if using 12V PVDD as power supply of TAS5825M:

$$V_{speaker\_max} \approx 12V \times \frac{4}{4 + 2 \times (0.18 + 0.023)} = 10.89V \tag{19}$$

$$Analog\_gain \approx 20 \times \log_{10}\left(\frac{10.89V}{29.5V}\right)dB \approx -8.7dB \tag{20}$$

For TAS5825M, the analog gain step is 0.5dB. So in this example, the analog gain needs to be set to -9 dB to avoid PVDD clipping, shown Figure 5-6.

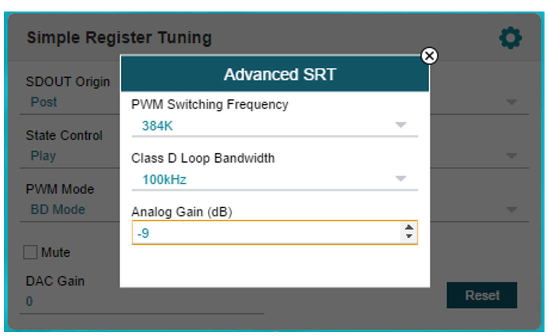


Figure 5-6. Analog Gain Setting

5.1.5 Adjust System Gain

After setting analog gain, please go back to *Characterization* view to adjust the system gain and supply voltage accordingly. As shown in Figure 5-7, the supply voltage needs to be set firstly, and then the system gain needs to be set according to actual analog gain and original system gain measured in *System Check* step.

$$System\_gain = Original\_system\_gain \times 10^{\frac{Analog\_gain}{20}} \tag{21}$$

Figure 5-7. SOA Settings in Characterization View

For example, if the original system gain configured in *System Check* step is 19.5V/Fs, then, with the analog gain set to –9dB, the system gain need to be adjusted to:

$$System\_gain = 19.5 \times 10^{\frac{-9}{20}} V/Fs \approx 6.92 V/Fs \tag{22}$$

5.1.6 Equalizer Setting

In TAS5825M, the smart amp processing flow provides 10 BQs for both L/R channel which can be manually configured for tuning.

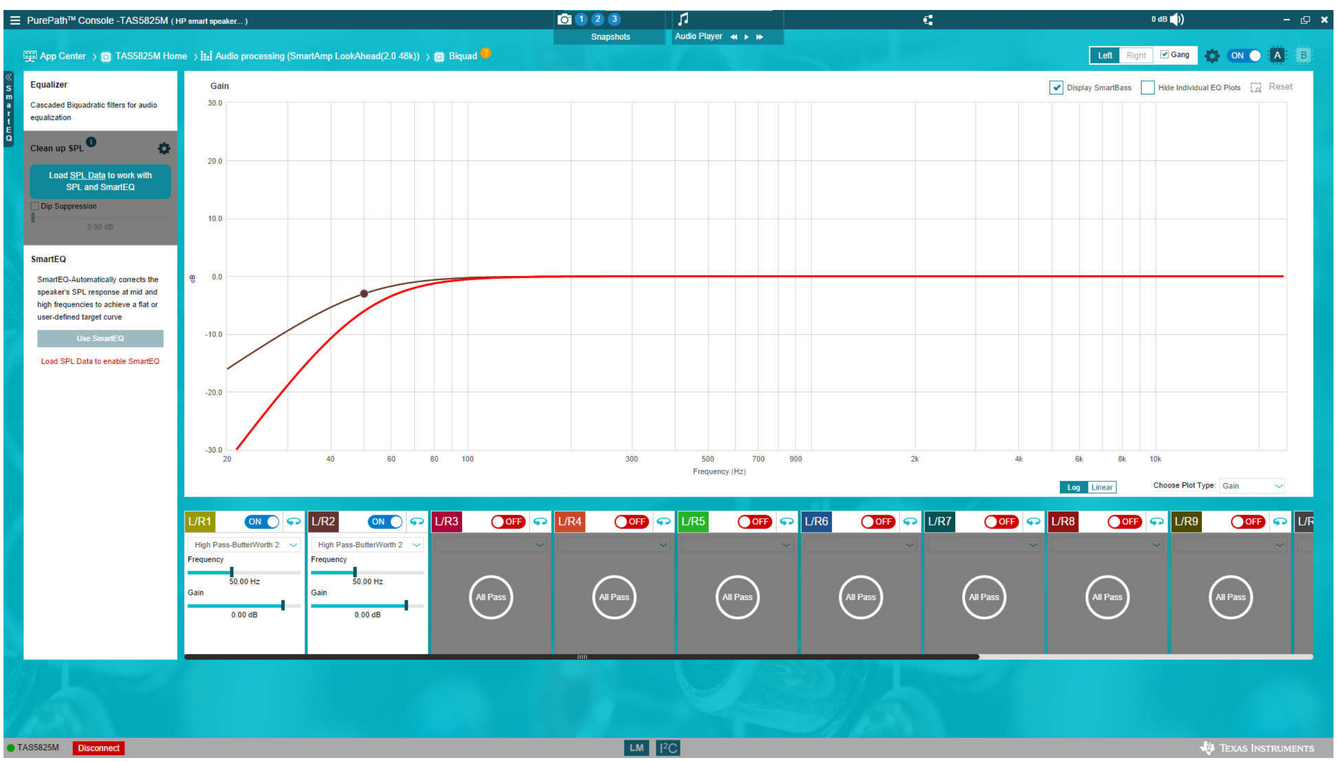


Figure 5-8. Equalizer Settings

Usually, at the beginning, the recommendation is to set two high pass filter to bypass very low frequency signals, because at very low frequency range, the speakers can consume too much power but cannot generate large audio outputs, which can decrease the system efficiency. Usually 50Hz to 100Hz frequency can be used to be the cutoff frequency for this high-pass filter.

The device also supports a Smart EQ function. By using the Smart EQ function, the PPC3 software can help to configure EQ parameters automatically according to speaker SPL data to deliver a flat response or match a target curve conveniently. To use this Smart EQ function, the speaker SPL data needs to be imported first.

5.1.7 Smart Bass Tuning

Smart bass is the key protection block to configure the smart amp algorithm, and contains four main function blocks: bass compensation, excursion protection, thermal protection, and anti-clipper. To achieve desired and satisfying audio effect, each function block can be turned on or off based on actual requirements.

Figure 5-9. Smart Bass Configuration

5.1.8 Bass Compensation

With the bass compensation function, the algorithm can automatically morph the audio response to improve the the base performance.

5.1.8.1 Corner Frequency

The corner frequency indicates the –3dB point the target magnitude response (indicated by the green curve in Figure 5-9). Selecting a proper corner frequency is important for the overall performance of the system. If this frequency is too high, the speaker's bass response can be limited. If the frequency is too low, the electrical power can be wasted trying to drive low frequencies that the speaker cannot response to well, and also make the excursion protection oversensitive.

5.1.8.2 Alignment Order and Type

The order and type settings determine the attenuation speed of bass signals for the target magnitude response, which can have significant influence on the audio response for signals below the corner frequency, and this setting can be kept in default values for most applications. If the speakers are very sensitive and cannot withstand large excursions, the suggestion is to select a higher order. And a lower order setting can be chosen to keep more low frequency signals and improve the audio bass performance.

When adjusting these settings, a series of listening tests need to be performed, and here are some practical suggestions:

1. Adjust *Corner Frequency* while watching the compensation curve (red curve) in the response plot window.
2. Compensation (red curve) between 10dB to 20dB usually can provide good performance.
3. Do not exceed the 20dB line (at least in the beginning of tuning).

The speaker magnitude response curves with and without the bass compensation have been demonstrated in Figure 5-10 and Figure 5-11.

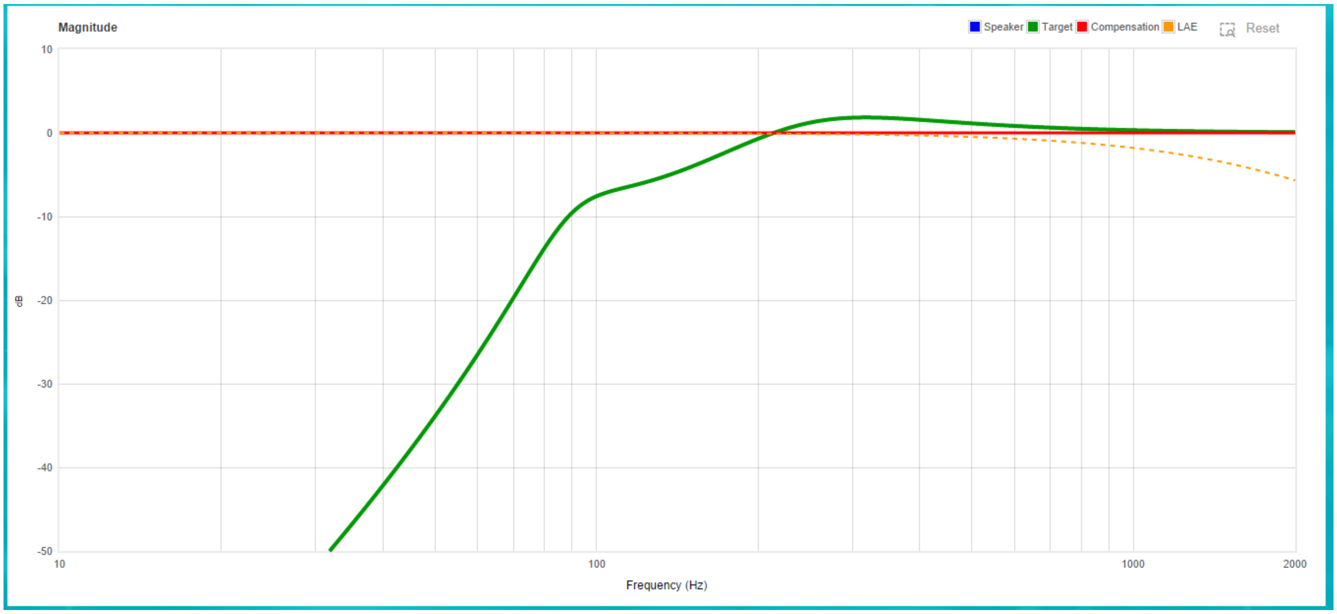


Figure 5-10. Speaker Response With Bass Compensation Disabled

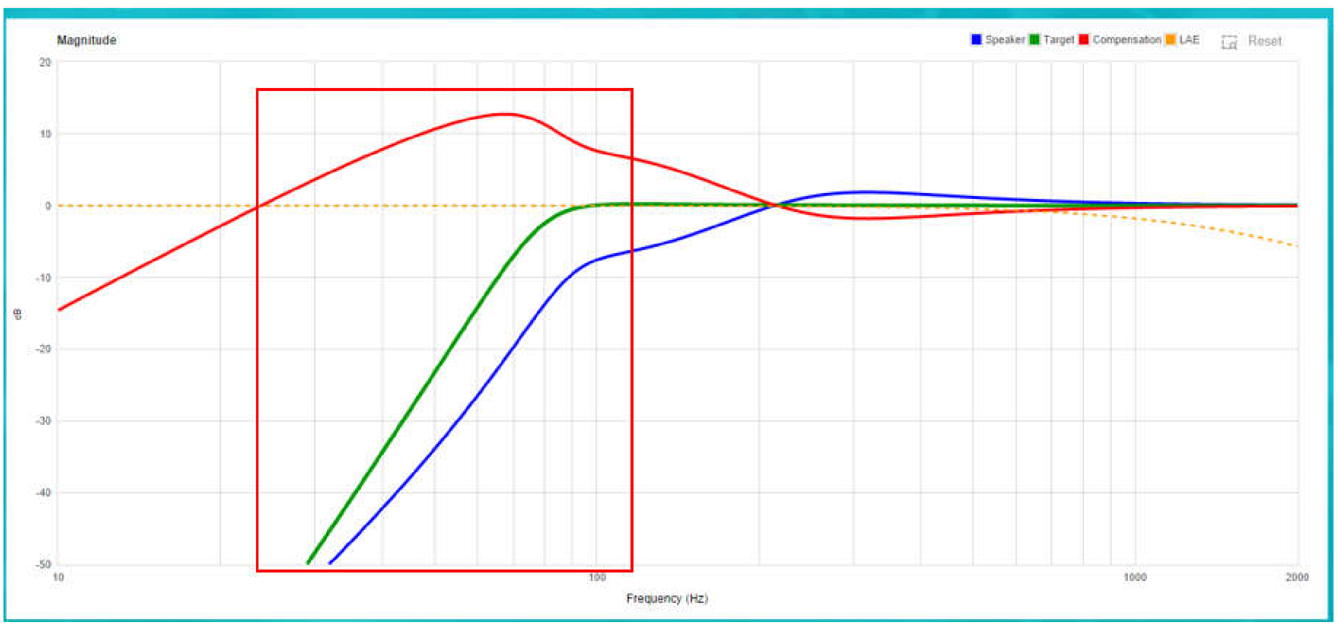


Figure 5-11. Speaker Response With Bass Compensation Enabled

5.1.9 Max Level Tuning

Max level tuning includes speaker excursion protection (Xmax) and speaker thermal protection (power limit), which can make sure speakers to work within SOA. This is easy to enable or disable these two protections independently by clicking the bypass buttons. Due to the potential error of protection algorithm and speaker models, this is needed to adjust the setting carefully according to actual listening test and algorithm protection situation.

5.1.9.1 Xmax

Xmax means the limit of the speaker excursion, and can be set according to the specifications provided by speaker vendors.

5.1.9.2 LAE Frequency

LAE Frequency refers to the look ahead excursion frequency, and this is the active range for the excursion protection algorithm. As shown in Figure 5-13, the peak of the excursion of the speaker usually happens in low-frequency range, while the speaker excursion in high frequency range is very small. So the speaker excursion damage mostly occurs with large amplitude low frequency signals, and therefore the excursion protection algorithm do not take high frequency signal into considerations.

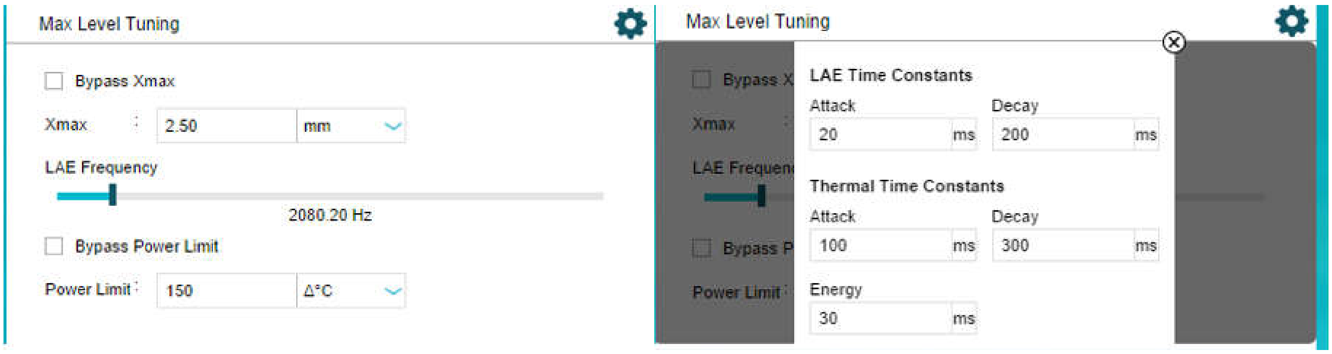


Figure 5-12. Max Level Tuning Settings

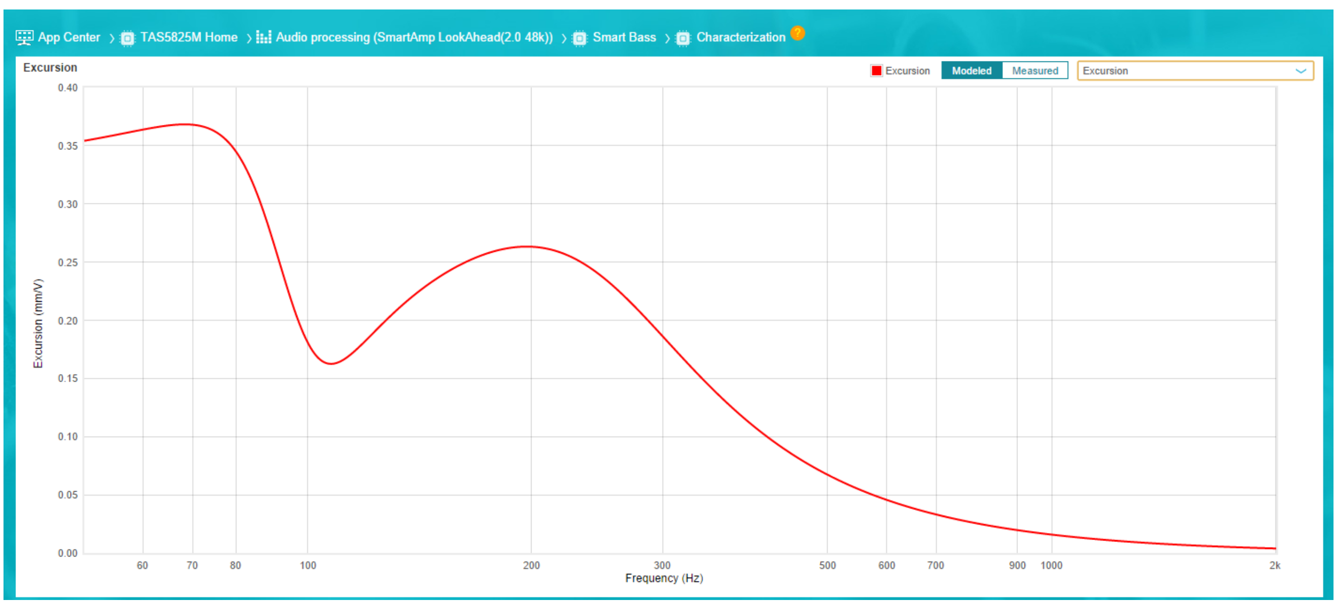


Figure 5-13. Speaker Excursion vs Frequency Plot

As shown by the orange dash line in Figure 5-14, LAE frequency is the cutoff frequency of the low pass filter which is applied to reduce high frequency signals, and then the signals is sent into the excursion protection algorithm for calculation and protections.

If the value of LAE frequency is too high, the algorithm active range can includes too much high-frequency signals, which can cause the protection behavior to be too active. On the contrary, if the value of LAE frequency is too low, the protection algorithm can miss too much low frequency signals, and can cause the speaker excursion to exceed the limits. Usually, the suggestion is to adjust this value as 1 to 3 times of the model fitting frequency range of the speaker (for example, in Figure 5-14, we set the LAE frequency to around 2000Hz, similar as the model fitting range).

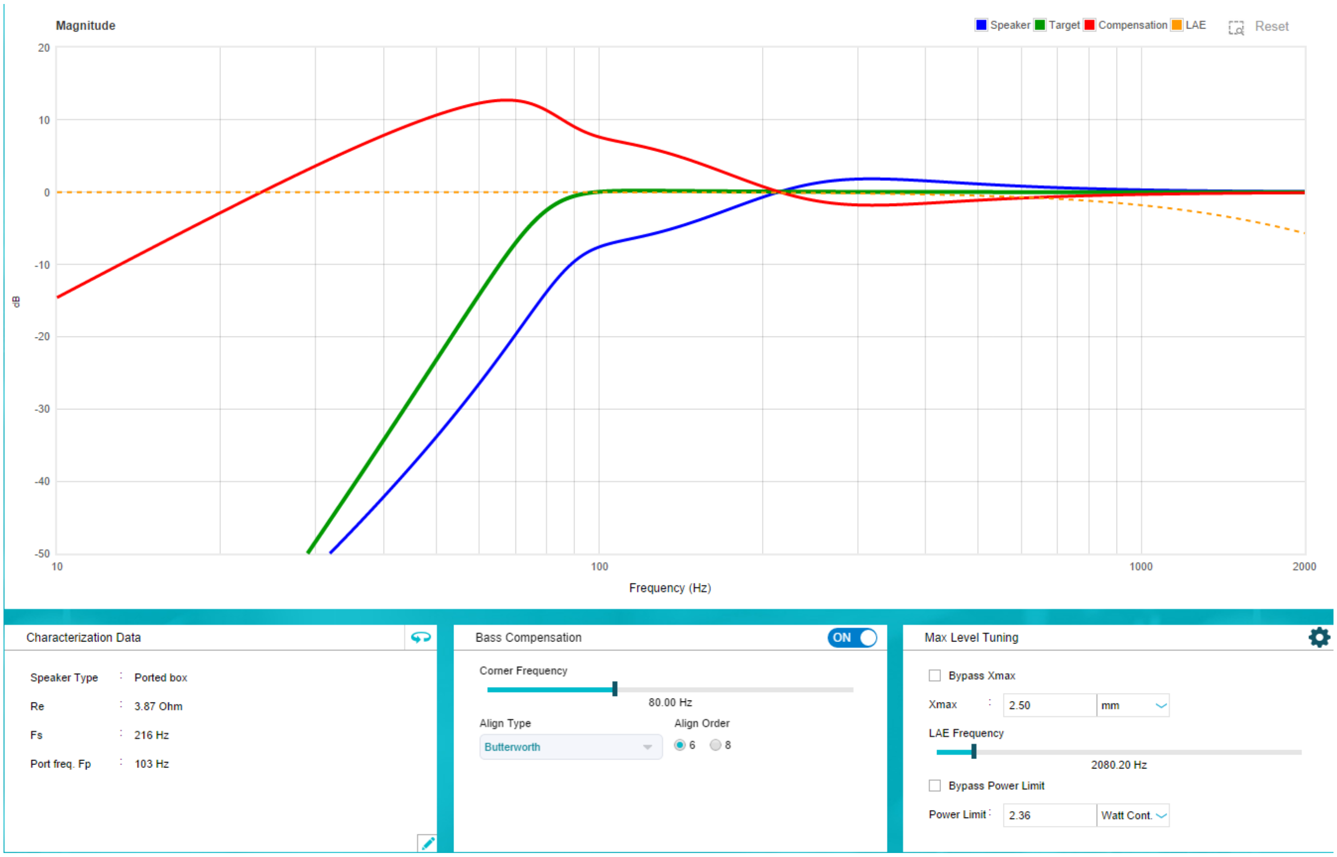


Figure 5-14. LAE Frequency Settings

5.1.9.3 Power Limit

Power limit is the threshold for the algorithm to trigger speaker thermal protections. This can be input in different methods, namely, power threshold or temperature threshold, which are equivalent to each other based on the speaker models.

5.1.9.4 Attack, Decay, Energy

Same as the DRC block in TAS5825M's typical processing flows (see the application note, *General Tuning Guide for TAS58xx Family*), the attack and decay time refer to the time for protection algorithm to take effect and release. Energy time is the time for the algorithm to detect the input signal energy.

Usually, this is suggested to set the attack and energy time near the look ahead time of the algorithm (for example, for 48K processing flow, the look ahead time is about 2.67ms), and the decay time can be set slightly larger than the attack time. A shorter attack time can make the algorithm to respond to audio signals more quickly, while this can also cause loudness loss when music with large dynamics is played. If the energy time is very short, the algorithm can be more sensitive and cause more compression, which can also affect the music loudness.

The following graphs illustrate the examples of speaker excursion (measured by laser equipment) with different attack and energy time settings, with the same sweep sine wave signal. When adjusting these settings, this is advisable to perform a series of listening tests. To get the best configuration to achieve great listening and protection result, it's also recommended to choose several speakers for listening tests and protection tests.

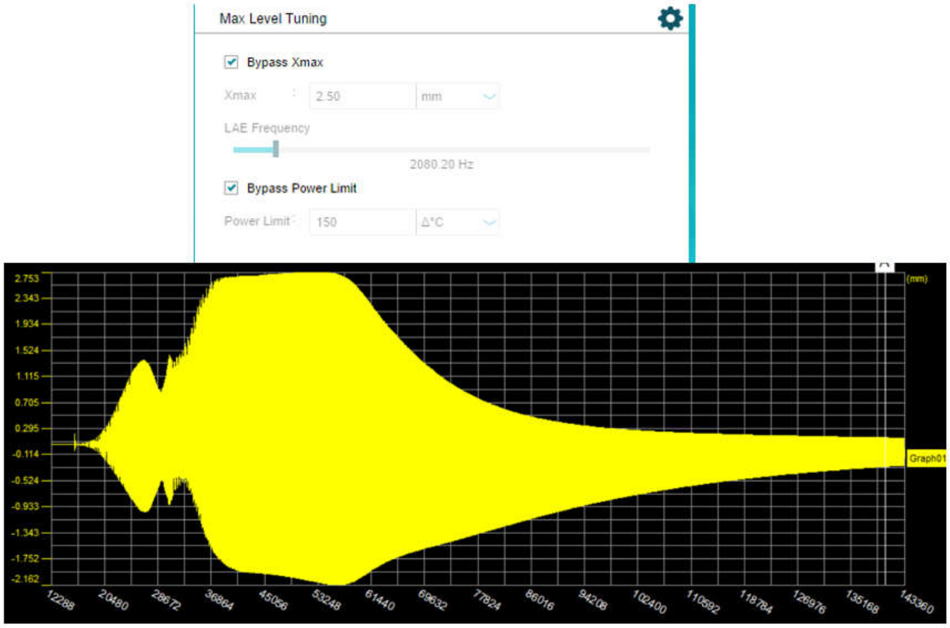


Figure 5-15. Speaker Excursion Without Protection (X_{max} = 2.75mm)

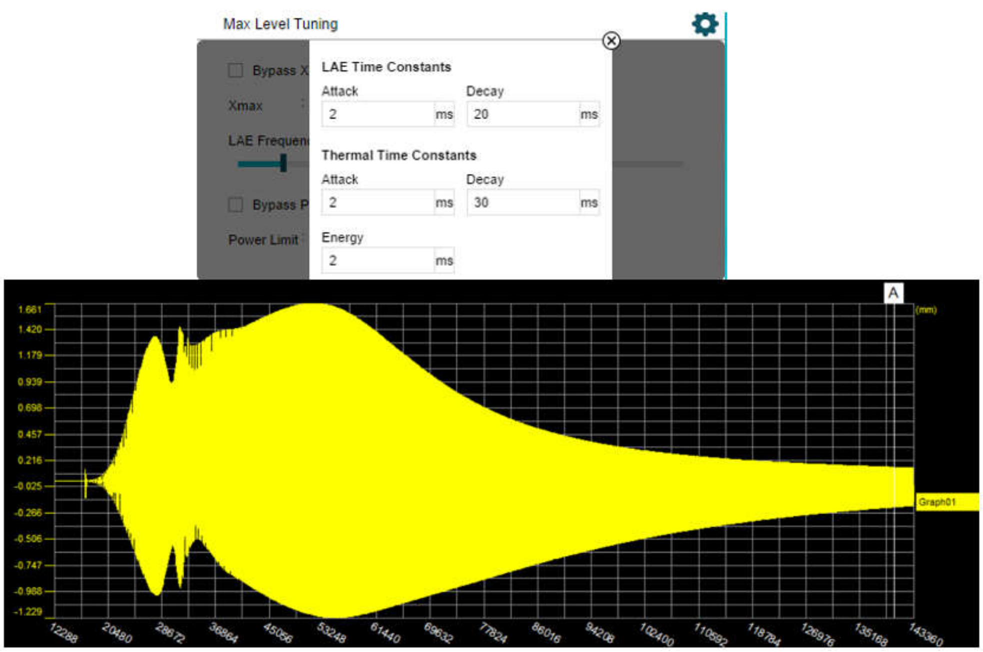


Figure 5-16. Speaker Excursion with Protection (X_{max}=1.66mm)

5.1.10 Anti Clipper

Anti clipper is a full band AGL block and this is the final signal limiting block in the whole processing flow to provide more flexibilities for tuning and protections.

5.2 Smart Amp Verification

5.2.1 SPL Response Verification

In this section, pink noise SPL responses have been measured to demonstrate the performance improvements when applying smart amp algorithm. As shown in Figure 5-17, TAS5825M can deliver more bass, louder mid and high frequency effect when smart amp algorithm is turned on, because the excursion and thermal protection can allow impressive audio improvement without speaker damage concerns.

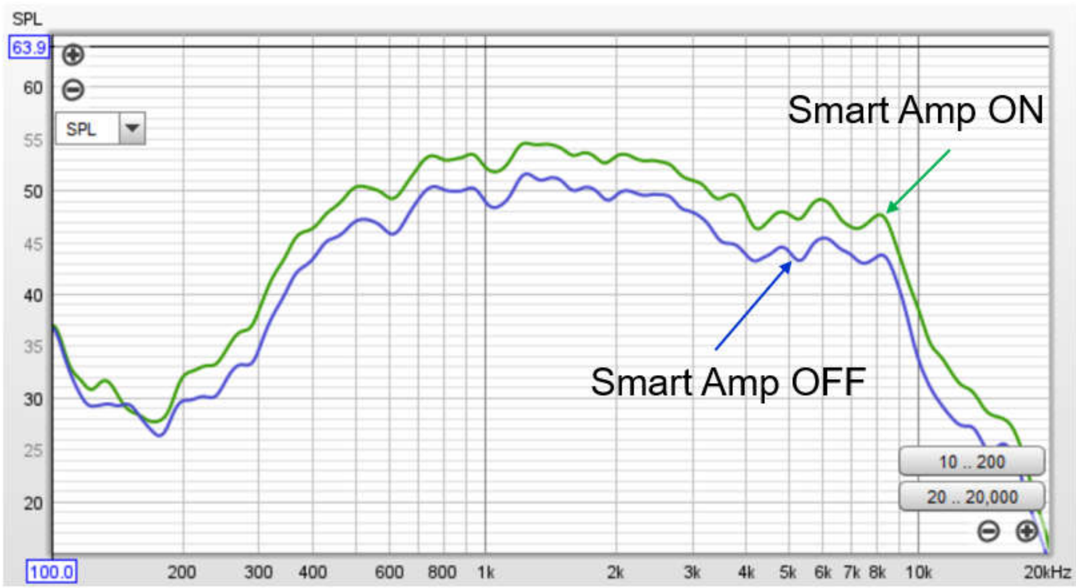


Figure 5-17. SPL Response Comparison (TI Smart Amp Tuning vs Normal Tuning)

5.2.2 Thermal Protection Verification

In an audio system with speaker I/V sense function, the speaker resistance Re can be measured in real time, whlie the ambient temperature can also be calculated. TAS5825M does not support I/V sense function, so the speaker Re cannot be automatically captured using the real-time IV data. But TAS5825M smart amp algorithm can use the audio data and speaker models to predict the temperature of the speaker voice coil, and reserve some margin to achieve thermal protection.

Table 5-1 and Figure 5-18 show the protection verifications of TAS5825M's thermal protection functions with single frequency audio signals. With different power limit settings (temperature increase margin), the algorithm can automatically limit the output power of TAS5825M to avoid thermal damage to speaker voice coils. In practice, TAS5825M takes the speaker Re in 25°C to calculate the output active power into speaker voice coils, and does not consider the increase of Re along with the rise of voice coil temperature, so the actual output power can be lower than expectations of the protection algorithm, which can cause the protection behavior to be more aggressive when temperature rises. As a result, like shown in Table 5-1, the actual temperature increase of voice coil is lower than the thermal limit settings, and the gap increases grows when temperature grows, which results in a more conservative protection mechanism.

Table 5-1. Thermal Protection Verification Results

Power Limit (W)	Thermal Limit (K)	Output RMS (V)	Re (Ω)	Measured Temp Change (K)	Error (K)
3	122.96	4.608	7.64	113.91	-9.05
2	81.97	3.767	7.12	81.66	-0.31
1	40.99	2.66	6.73	43.09	2.1

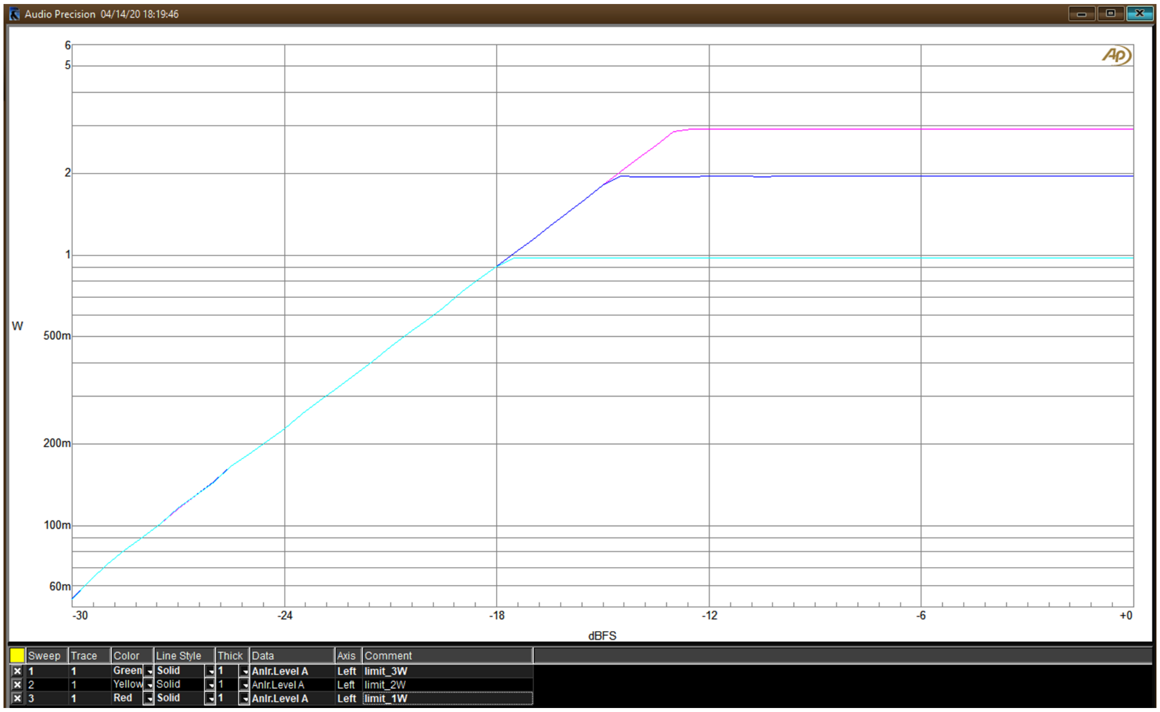


Figure 5-18. Thermal Protection Verification Results

6 Summary

This paper provides a comprehensive overview of implementing smart amp technology using the TAS5825M audio amplifier. Smart amp technology improves sound quality, maximizes audio power output, and protects speaker by using predictive algorithms to monitor and control speaker behavior.

The paper begins by explaining the fundamentals of speaker modeling and the basic principles of the smart amp algorithm. The paper outlines how TI's smart amp dynamically adjusts audio output based on speaker excursion and temperature to prevent damage while delivering better bass and dynamic range than conventional systems. Then, this paper provides detailed guidance on preparing hardware and software to apply smart amp with TAS5825M. Finally, detailed guidelines have been provided in this paper for both the speaker characterization and the smart amp tuning and verification, helping to facilitate the rapid implementation of smart amp with TAS5825M. Verification experiments have also been provided to validate the benefits and audio performance improvements with the smart amp technology.

7 References

- Texas Instruments, *General Tuning Guide for TAS58xx Family*, application note.
- Texas Instruments, *Smart Amp Tuning Guide*, application note.
- Texas Instruments, *Smart Amp Quick Start Guide*, application note.
- Texas Instruments, *Getting Started with Smart Amp Development*, application note.
- Texas Instruments, *SmartPA Speaker Protection Algorithm*, application note.

Mailing Address: Texas Instruments, Post Office Box 655303, Dallas, Texas 75265