



PXA988 PXA1088 Project

CAT-Audio

Audio Calibration

V0.5

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ASR

Document Conventions

	Note: Provides related information or information of special importance.
 Caution	Caution: Indicates potential damage to hardware or software, or loss of data.
 Warning	Warning: Indicates a risk of personal injury.

Document Status

Doc Status: User Guide	Technical Publication: V0.05
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About This Document

1.1 Purpose

Describe the CAT-Audio tool of the CAT-Studio.

1.2 Product/Sub-Product Overview

CT means Cellular Tool. CT Hall is a tool set for PXA910 or PXA920 based cellular communication system.

1.3 Abbreviations and Acronyms

Table 1: Abbreviations and Acronyms

Acronym	Description
ACI	AP-CP Interface
ACM	Audio Components Manager
AEC	Acoustic echo canceler
AGC	Automatic Gain Control
AMR	Adaptive Multi Rate
APPS/Apps	Application subsystem
AVC	Automatic Volume Control
BT	Bluetooth
COMM	Communication subsystem
CTM	Cellular Text Telephone Modem
EC	Echo Canceler
EQ	Equalizer
ES	Ear-Seal
ETSI	European Telecommunications Standard Institute
GSM	Global System Mobile
GSM-AMR	GSM Adaptive Multi-Rate
GSM-EFR	GSM Enhanced Full Rate
GSM-FR	GSM Full Rate
GSM-HR	GSM Half Rate
HF	Hands-Free
HLPF	High/Low Pass Filter
HPF	High Pass Filter

HSL	High Speed Logger
HW	Hardware
LPF	Low Pass Filter
MSA	Macro Signal Architecture
NB-AMR	Narrow-Band AMR
NREC	Noise Reduction, Echo Cancellation
NS	Noise Suppressed
PCM	Pulse Code Modulation
RES	Residual echo canceling
Rx path	Path to output device (speaker).
SP	Speaker
ST	Side tone
SW	Software
Tx path	Path from input device (microphone)
UMTS	Universal Mobile Telecommunications System
UMTS-AMR	UMTS Adaptive Multi-Rate
WB-AMR	Wide-Band AMR

1.4 References

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1.5 Document Structure

This document contains the following chapters:

Table 2: Document Structure

Section	Description
Chapter1	About this document
Chapter2	Overview
Chapter3	Audio Voice Call Calibration
Chapter4	CAT-Audio Enhancement Tab
Chapter5	HIFI Calibration
Chapter6	Audio Calibration Procedures
Chapter7	Calibration Files
Chapter8	NvmMerge Tool
Chapter9	NvmXml Tool

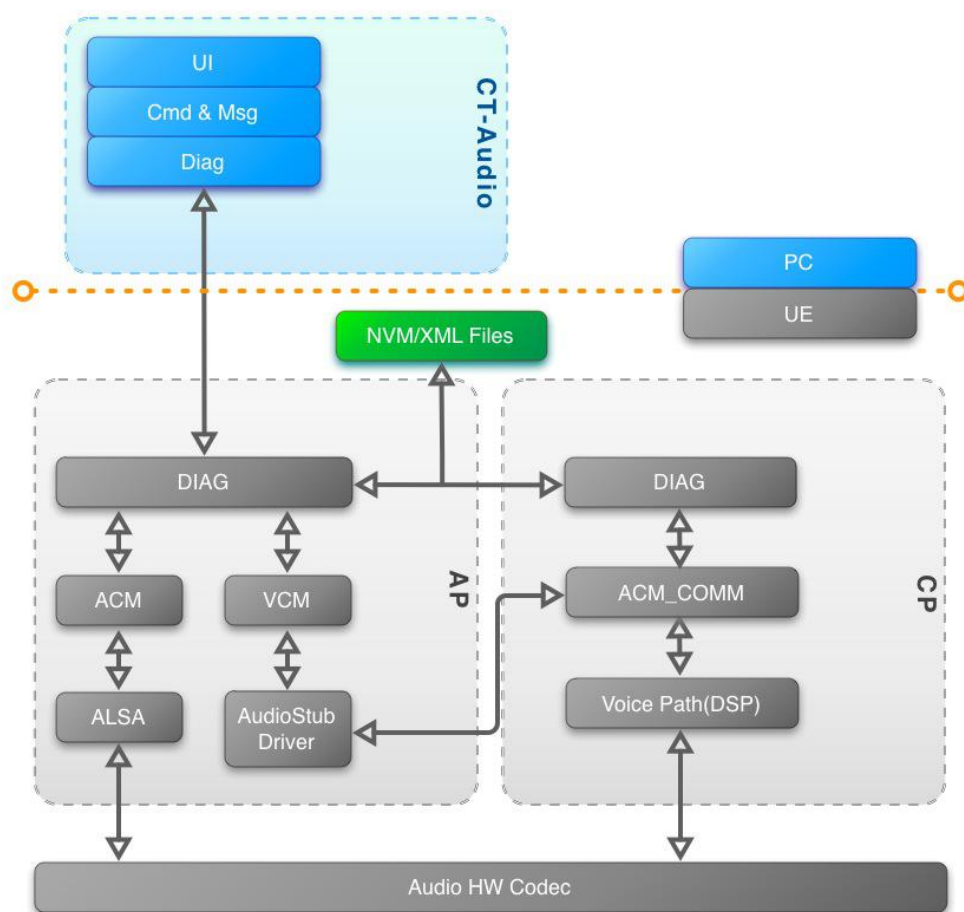
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Overview

2.1 Function

The CAT-Audio tool is designed as a module that runs from the CAT-Studio to provide immediate feedback about audio conditions in the system. The user interface includes pressing buttons and setting parameters and files. The CAT-Audio can work in off-line or on-line mode. There is rich material describing the usage of CAT-Studio.

The goal of this document is to demonstrate the audio calibration mechanism of CAT-Audio.



2.2 Related components

Flasher Explorer, NVM Editor.

2.3 CAT-Audio Version

Version 1.0.0.0 is for new audio architecture. It support PXA988, PXA986 and PXA1088.

2.4 Build Audio Calibration System

2.4.1 Introduction

The CT Audio Calibration Tab is specifically designed for calibrating cellular handsets built with the PXA910 or PXA920 cellular processor family. The software works with the Rohde & Schwarz Audio Analyzer (Model UPL16 or UPV) or the Head Acoustics ACQUA and a CMU-200 base station and the Advanced Cellular Analysis Tool system (CAT-Studio). The CT Audio software package enables calibration of audio parameters (DSP and codec) to ensure that mobile cellular handsets (developed on the PXA90x Cellular Processor Family platform) meet the minimum requirements of acoustic test cases as defined in 3GPP TS 51.010-1, Section 30 (up to Release 99).

This chapter explains the installation of the CT Audio software package, CT Audio Installation procedure, manual calibration, and automatic calibration.

2.4.2 Applicable Documents

The following lists supplemental information sources for CAT-Studio.

1. AUDIO ANALYZER R&S UPL16 Operating Manual
AUDIO ANALYZER R&S UPV Operating Manual
This document is available at <http://www.rohde-schwarz.com>
2. Supplement to AUDIO ANALYZER R&S UPL16. 1078.2008.16.
3. CAT-Studio User's Guide and Reference Manual
4. 3GPP TS 51.010-1, section 30, up to release 99.

This documentation is available at <http://www.3gpp.org/ftp/Specs>

2.4.3 Acoustics Hardware Setup

The acoustics setup is shown in Figure 1. Ensure that the following hardware conditions are met before starting the calibrations.

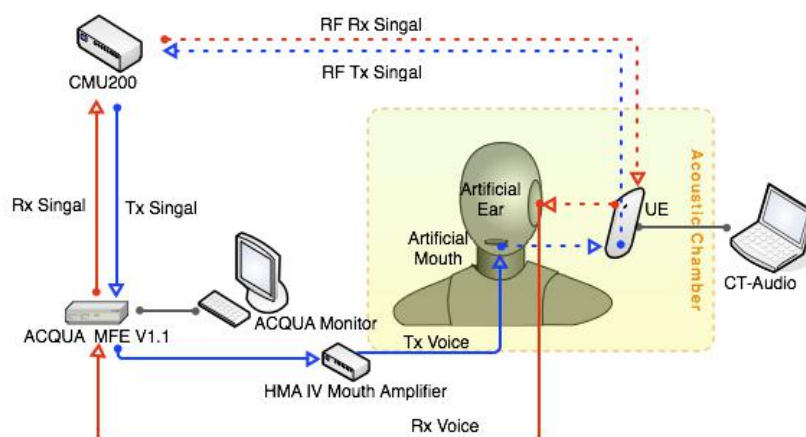


Figure 1 Acoustic Setup

The specific model numbers of the entire setup is given in Table 3.

Figure 2 Acoustic VOIP Setup



Note 1 : ASR used the ITUT Recommendation P.57 [108] Type 3.4 artificial ear for Release 96 MS or later. The positioning is defined in ITU-T Recommendation P.64..

Perform the following steps:

1. Turn on power.
2. Wait until initialization ends.
3. Choose the desired network:

GSM Configuration

1. Press the Network Tab soft key.
2. Select Circuit Switched by the cursor.
3. Select Bit Stream = Speechcod. / Handset Low.
4. Select Traffic Mode = desired Codec.
5. Return to the connection Tab screen.

WB Configuration:

1. Press the BS Signal Tab soft key.
2. Select Circuit Switched and Voice Settings by the cursor.

3. Select Voice source = Speechcodec Low.
4. Select Adaptive Multi Rate = 12.2KBPS selection recommended.
5. Return to the connection Tab screen.

4.3.2

UE Setup

Perform the following steps:

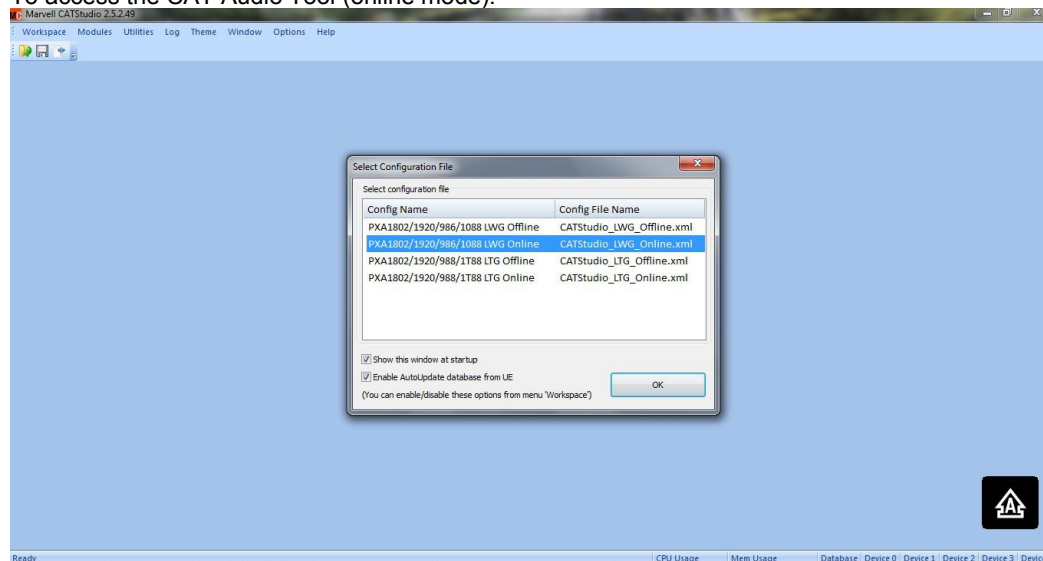
1. Make sure the UE is powered on.
2. Make sure the UE is connected by a USB cable to the CAT-Studio PC.
3. Make sure the UE is synchronised with the base and generate a voice call.
4. Place the UE in the proper place on the artificial head.
5. In case of Speakerphone, switch to Speakerphone mode.

Audio Voice Call Calibration

This chapter detail CAT-Audio basic operation and audio physical path.

3.1 CAT Audio Tool Activation

To access the CAT Audio Tool (online mode):

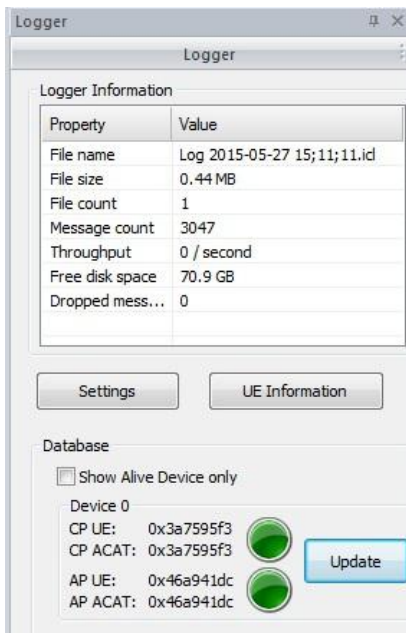


1. Open CAT-Studio and select "Online mode".

Figure 1: Open CAT-Studio online mode.

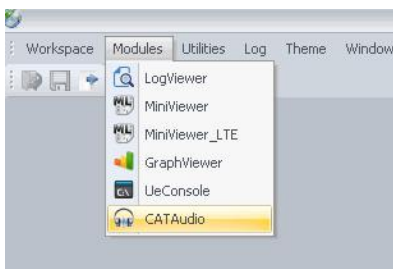
2. Connect CAT-Studio with board, and check whether AP and CP Database is matched and can receive DIAG messages correctly.

Figure 2: Check CP and AP database.



3. Open the CAT-Audio from CAT-Studio menu as next figure.

Figure 3: Open CAT-Audio



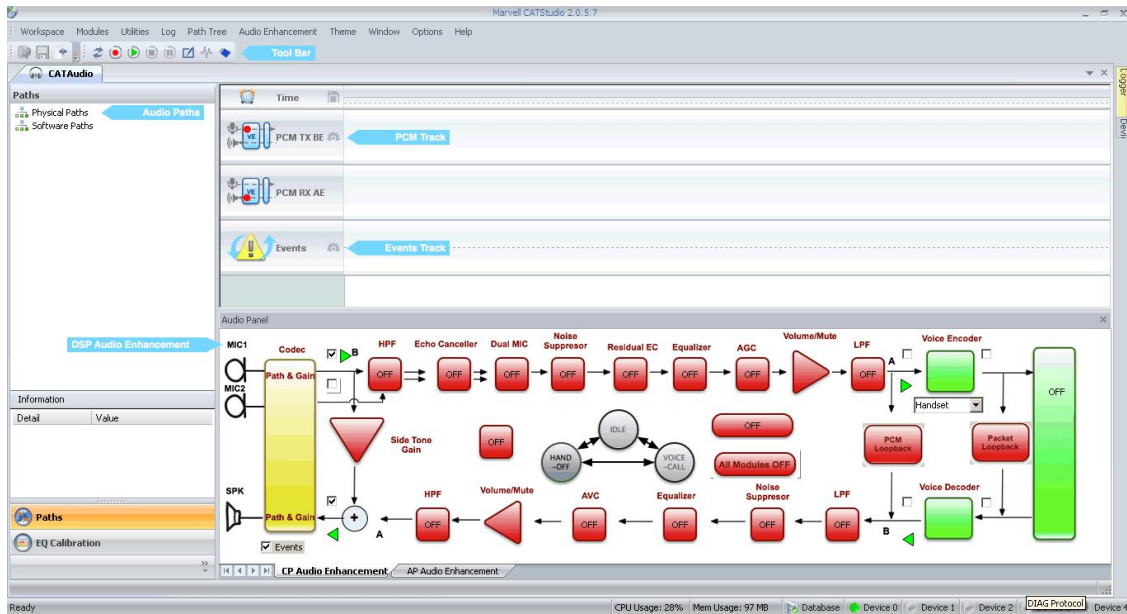


Figure 4: CAT-Audio interface

3.2 Audio Physical Path Calibration

To calibrate the audio paths, perform the following steps:

1. Setup voice call use UE's GUI or AT command.
2. Click button "Refresh ALL" to refresh paths. It will highlight and set current enabled path to top position of tree view.

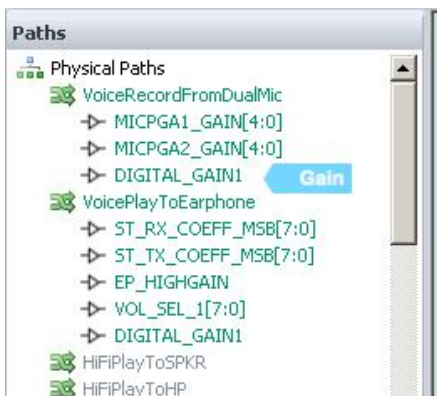
Figure 5: "Refresh ALL" button



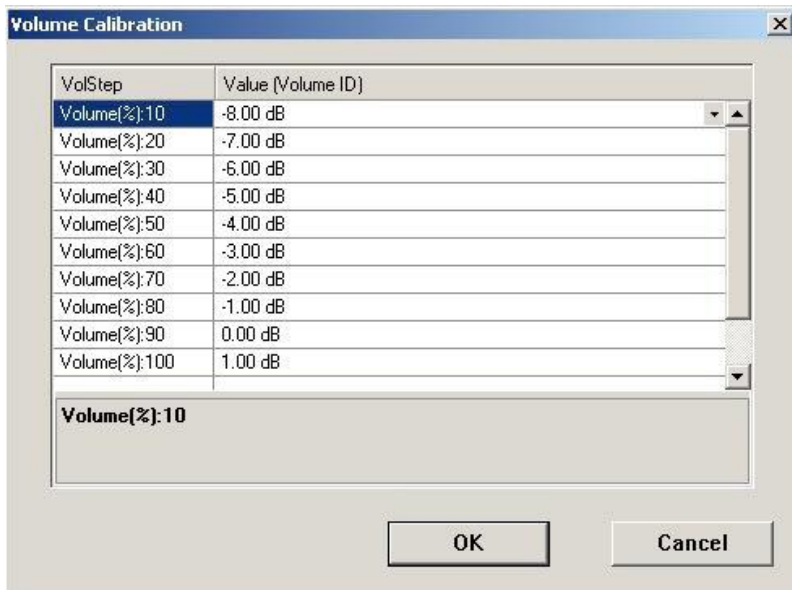
Figure 6: Highlighted Audio Paths



3. Double-click the codec to expand configuration path tree. The audio path sub-tree is displayed
4. Select the appropriate audio path and double-click to open the gains sub-tree associated with the selected audio path. For example, if you want calibration headset, you should make a call with headset and the path **VoicePlayToEarphone** will be enabled, and adjust the volume.



5. Select a gain and right-click to open a popup menu.
6. Select Cal Gain from the popup menu, the Calibration Gain dialog box is displayed
7. Set the Gain Value for every volume steps.



8. Click the **[OK]** button to update the calibration date and ensure that the change has been updated.
9. Run test case by ACQUA or UPV.
10. Wait for the test to end. If all pass, test is complete.

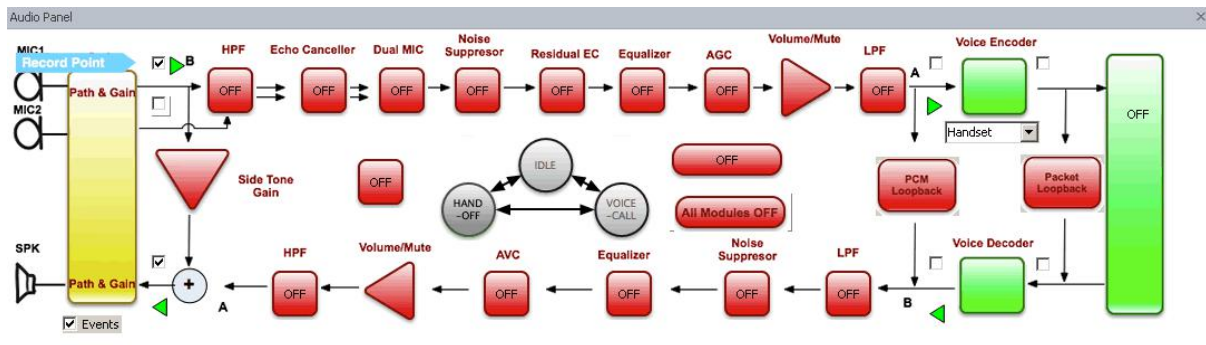
3.2.1 Refresh Components

When you make a call, some path will be enabled. But the path of component maybe can't update status right now. So you need click [Refresh All] button to refresh path status. It will refresh both physical and software path.

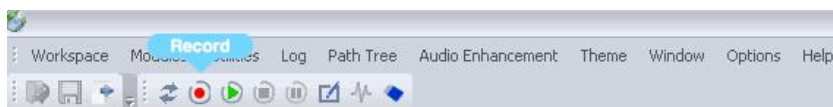
3.3 Voice Recoding

1. Connect with target.
2. Click tracks view button to select tracks view.

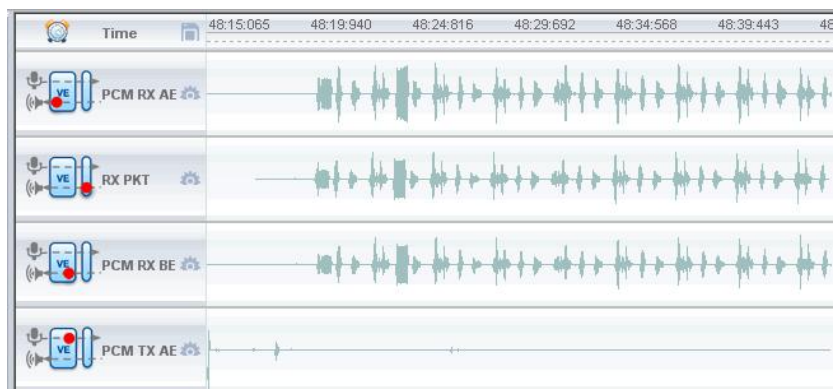




4 Click Record Button



5 Make a call, and you can check wave graph of record point.



6 Click the stop button to stop recording.

7 Click Save button on Time Axis to export data to wave files.



3.4 Voice Playback

1 Right click the track viewer and drag the mouse to select a part of speech data.



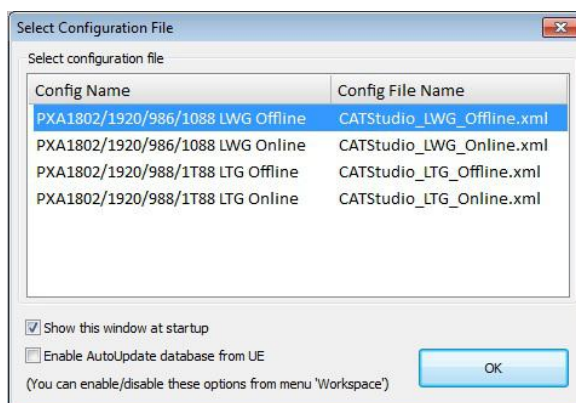
2 Click play button to play selected PCM. And you can save this part of PCM by click “Save Button”.



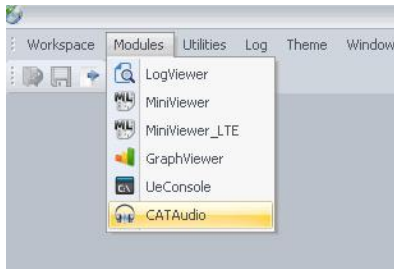
3.5 Load Log

User could load zip log to check Voice PCM data and Messages.

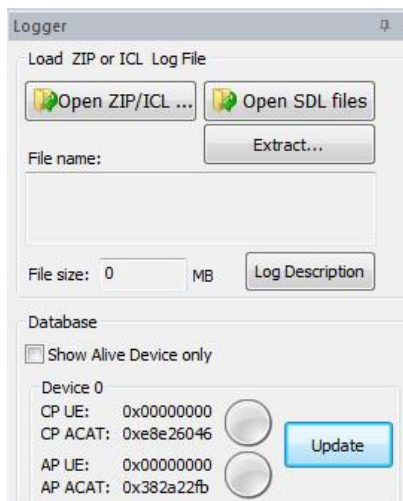
1 Select **Offline mode** when you open CAT-Studio.



2 Open CATAudio by clicking CATAudio menu.



3 Click Button **Open** button to import a log file.



4 Click Button **Tracks View** and **Messages View** to select related view.



Tracks:

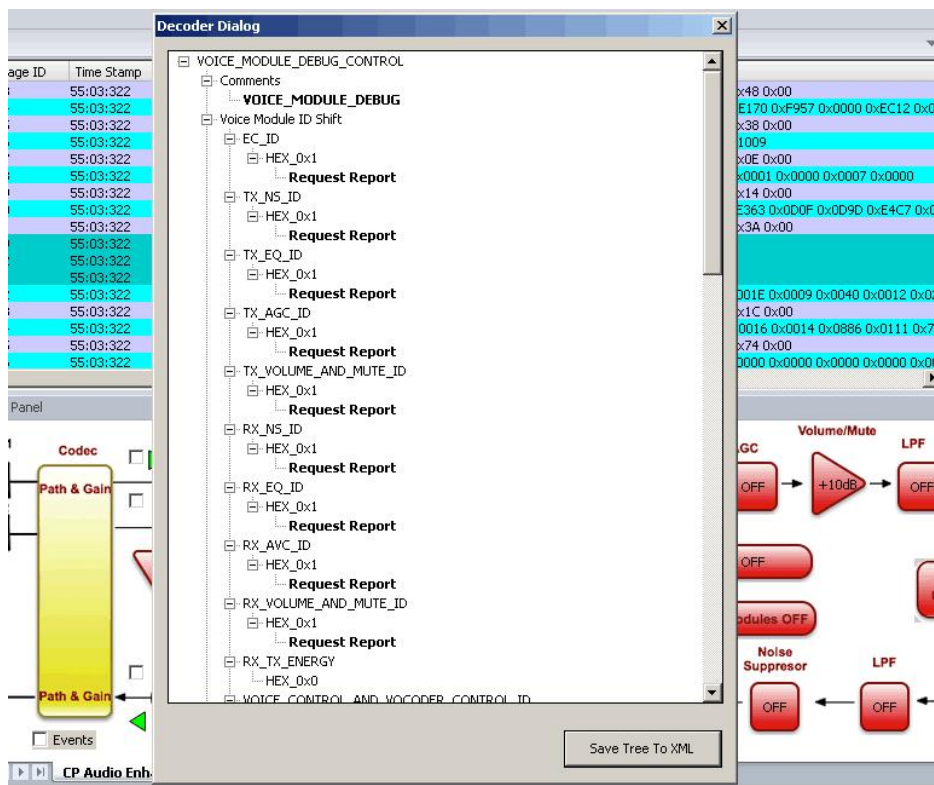
1. Put mouse on the Time Axis. **Wheel up** to Zoom in graph. **Wheel down** to Zoom out graph.
3. Drag the slider bar to move forward or backward.

Messages:

- 1 This viewer shows the command and message of DSP.

Message ID	Time Stamp	Control Name	OpCode	Data	Remarks
257536	48:53:539	AUX MODE CONTROL	0x0069	0x0000	DSP Command
257537	48:53:539	AVC CONTROL	0x0067	0x0000 0x0000 0x0400 0xDAB4 0xF558 0x0000 0xE806 0x0005 0x0052 0x0014	DSP Command
257538	48:53:539	EQ CONTROL	0x0066	0x0000 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000	DSP Command
257539	48:53:539	NS CONTROL	0x0064	0x0000 0x0000 0x1CA8 0x0016 0x0014 0x0886 0x0111 0x7FFF 0x001E 0x0000	DSP Command
257542	48:53:539	VOLUME CONTROL	0x0062	0x0000 0x002F	DSP Command
257543	48:53:539	FILTERS CONTROL	0x0074	0x0000 0x0001 0x0000 0x1000 0xE363 0x00DF 0x0D9D 0xE4C7 0x0D9D	DSP Command
257544	48:53:539	FILTERS CONTROL	0x0074	0x0000 0x0001 0x0001 0x1000 0x1B01 0x0BE9 0x0CF6 0x19EC 0x0CF6	DSP Command
257545	48:53:539	FILTERS CONTROL	0x0074	0x0000 0x0000 0x0000 0x1000 0xE363 0x00DF 0x0D9D 0xE4C7 0x0D9D	DSP Command
257546	48:53:539	FILTERS CONTROL	0x0074	0x0000 0x0000 0x0001 0x1000 0x1B01 0x0BE9 0x0CF6 0x19EC 0x0CF6	DSP Command
257549	48:53:539	VOICE CONTROL	0x0061	0x3C0C 0xCCD8	DSP Command
257590	48:53:539	START VOICE PATH	0x0040	0x3C0C 0xCCD8 0x0000 0x0000 0x0007 0x0000 0x0001	DSP Command
257591	48:53:539	MUTE CONTROL	0x0063	0x0001 0x0000 0x0000	DSP Command
257592	48:53:539	MUTE CONTROL	0x0063	0x0000 0x0000 0x0000	DSP Command
263252	48:56:101	VOICODER CONTROL	0x0060	0x0018 0x0000 0x0007	DSP Command
270003	48:59:913	DTMF CONTROL	0x0068	0x0000 0x0000 0x0002 0x0000 0x0000 0x0000 0x0250 0xD0A8 0x0000 0x0000	DSP Command
286473	49:08:332	VOICODER CONTROL	0x0060	0x0018 0x0000 0x0007	DSP Command

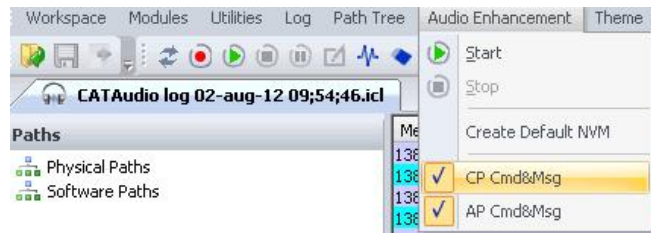
2 Select one command or message of DSP, and right click mouse button. It will pop up a window to decode the command of message.



The screenshot shows the 'Decoder Dialog' window. On the left, a tree view is expanded to 'VOICE_MODULE_DEBUG_CONTROL'. Under 'Comments', 'VOICE_MODULE_DEBUG' is selected. Below it, 'Voice Module ID Shift' is expanded, showing a list of controls like EC_ID, TX_NS_ID, TX_EQ_ID, TX_AGC_ID, TX_VOLUME_AND_MUTE_ID, RX_NS_ID, RX_EQ_ID, RX_AVC_ID, RX_VOLUME_AND_MUTE_ID, RX_TX_ENERGY, and VOICE_CONTROL_AND_VOICODER_CONTROL_ID. Each control has a 'Request Report' button. On the right, a panel displays various audio processing blocks: 'Volume/Mute' (with a +10dB gain block), 'LPF' (Low Pass Filter), 'Noise Suppressor', and 'AGC' (Automatic Gain Control). The 'Volume/Mute' block is currently set to 'OFF'. The 'Noise Suppressor' block is also set to 'OFF'. The 'AGC' block is set to 'OFF'. The 'LPF' block is set to 'OFF'. The 'Path & Gain' block is set to 'OFF'. The 'Events' block is set to 'OFF'. The 'CP Audio Enh' block is set to 'OFF'. The 'Save Tree To XML' button is visible at the bottom right of the dialog.

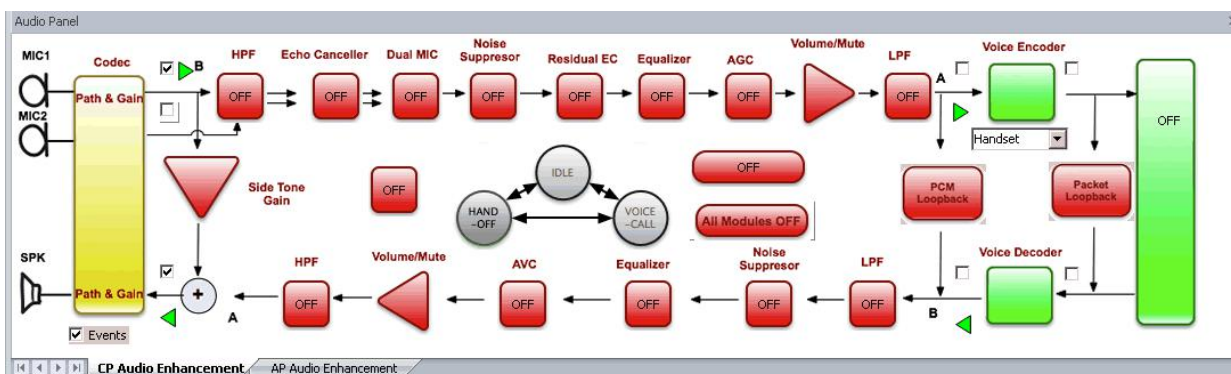
3 You can save the result to XML file.

4 You can enable or disable **CP Cmd&Msg** or **AP Cmd&Msg** by menu.



CAT-Audio Enhancement Tab

The Audio enhancement panel provides control of all voice enhancement (VE) modules. It enables turning on and off each module and setting the module's parameters. It also enables recording PCM audio samples from four different points in the voice enhancement path: before VE, after VE, in the TX and the Rx path. It also provides the ability to playback wave files from those points, and enables replacing VE samples or adding VE samples. The CT Audio Enhancement Tool also enables recording audio packets from the Tx-path immediately after the voice encoder, and, in the Rx-path, just prior to the voice decoder. It displays important information during a voice call, such as the



vocoder type used, vocoder rate, DTX status, codec devices and paths & gains information. The following is a description of the Audio Enhancement tab:

4.1 Voice Module Option Descriptions

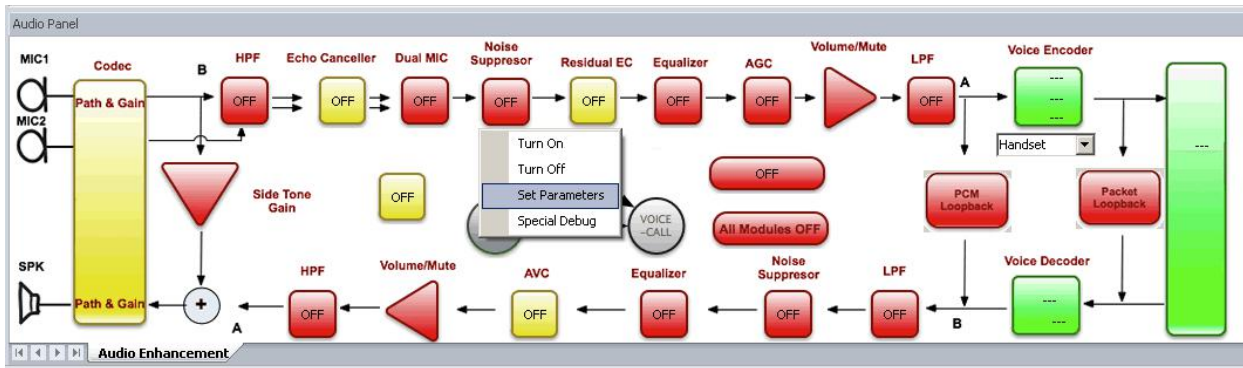


The Voice Module parameters are set in the CT Audio Enhancement tab. All module operations should be activated while a voice call or VOIP call is in progress on the User Equipment, and the **Record** button has been pressed.

4.2 Module Options

The GUI indicates that the module is On by displaying a Green color, and that the module is Off by displaying a Red color.

- ☑ Left click on this module on a Red button (indicating an OFF state) causes DSP commands to be sent that turns ON the pressed module (and it turns it Green).
- ☑ Left click on this module on a Green button (indicating an ON state) causes DSP commands to be sent that turns Off the pressed module (and it turns it Red).
- ☑ Right click on this module opens a list of available options: Turn ON, Turn OFF, Set Parameters.



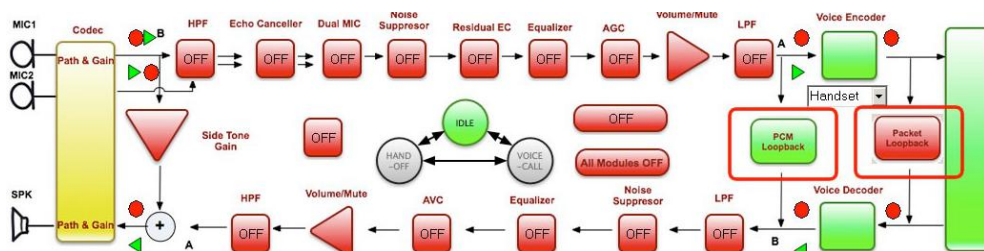
- ☑ Selecting the Set Parameters option from the right click menu, it will pop up a dialog that displays all parameters of the pressed module and their current values.
- ☑ Changing parameters and pressing the Send button sends the selection to the DSP. Pressing the Cancel button exits, leaving the parameters as they were before the list window was opened.
- ☑ In the List Parameters window, there is a Format Change button that allows the user to set the parameter values in a format specified in the Audio-Path specifications.

NOTE:

- Turning OFF the Echo Canceller (EC), when the Residual EC in the TX and the RX path is ON, also turns OFF the ECs.
- Turning ON or OFF of the Residual EC in the TX-path also turns ON or OFF the residual EC in the RX-path.
- In the Set-Parameters window of the EQ module, you can also see a graph of the frequency response of the equalizer taps that were selected.
- Not all modules have a Set Parameters option.
- The All Modules OFF button in the CAT-Audio Enhancement panel toggles ON and OFF for all modules at once.
- The state machine described in figure 2, shows the current state of the voice path in the DSP during a voice call.

4.3 PCM Loopback

The PCM Loopback button performs a loop on the PCM samples from the TX path into the RX path. This option can be activated when the UE is in idle state or in a voice call. During a voice call the received RX samples are overwritten by the TX path PCM samples.



Activate (Green) and Deactivate (Red) the loop back by left clicking on the button and toggling to the desired state.

4.4 Packets Loopback

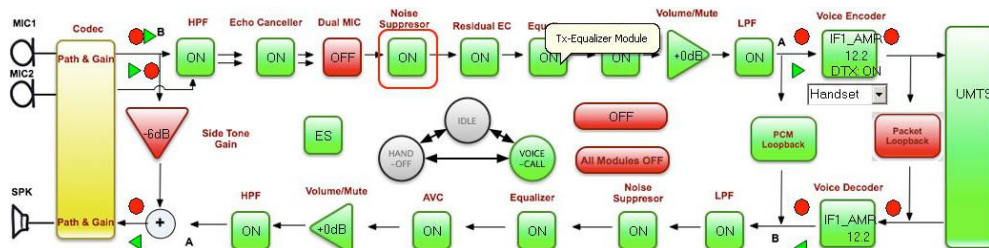
The Packets Loopback button performs a loop on the voice packets from the TX path to the RX path. This option can be activated when the UE is in an idle state or in a voice call. During a voice call, the received RX voice packets are overwritten by the TX path voice packets.

Activate (Green) and Deactivate (Red) by left clicking on this button and toggling to the desired state.

Note: The loop back options cannot be activated at the same time. When one is active and the user activates the second, the second will be activated and the first will be deactivated.

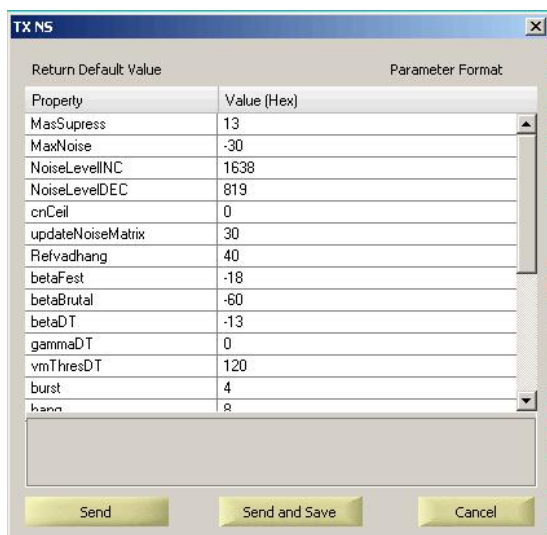
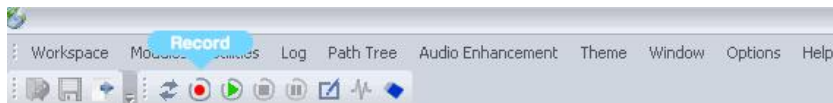
4.5 How to modify DSP parameters

Next sample will show you how to modify the parameters of DSP modules by change NS parameter.



1 Open CAT-Audio module, then select CAT-Audio Enhancement module

2 Press **Record** Button to enable Audio debug mode. Right click NS module and select [Set Parameters].



3 Make a call and tune MaxSupress from 13 to 20, and click send.

4 It will take effect immediately, but it will not save to NVM.

5 If you want save to NVM, please use button “Send and Save” on **Call Process**.

6 Check the saving result in NVM.

Notice:

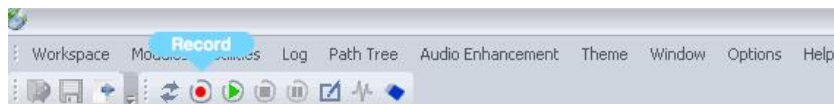
1 “Send and Save” function only could be used in call process. If you want save parameters of Speaker mode, please switch to speaker mode.

2 You should enable related modules in NVM previously. If DSP module is off, CAT-Audio will not save it.

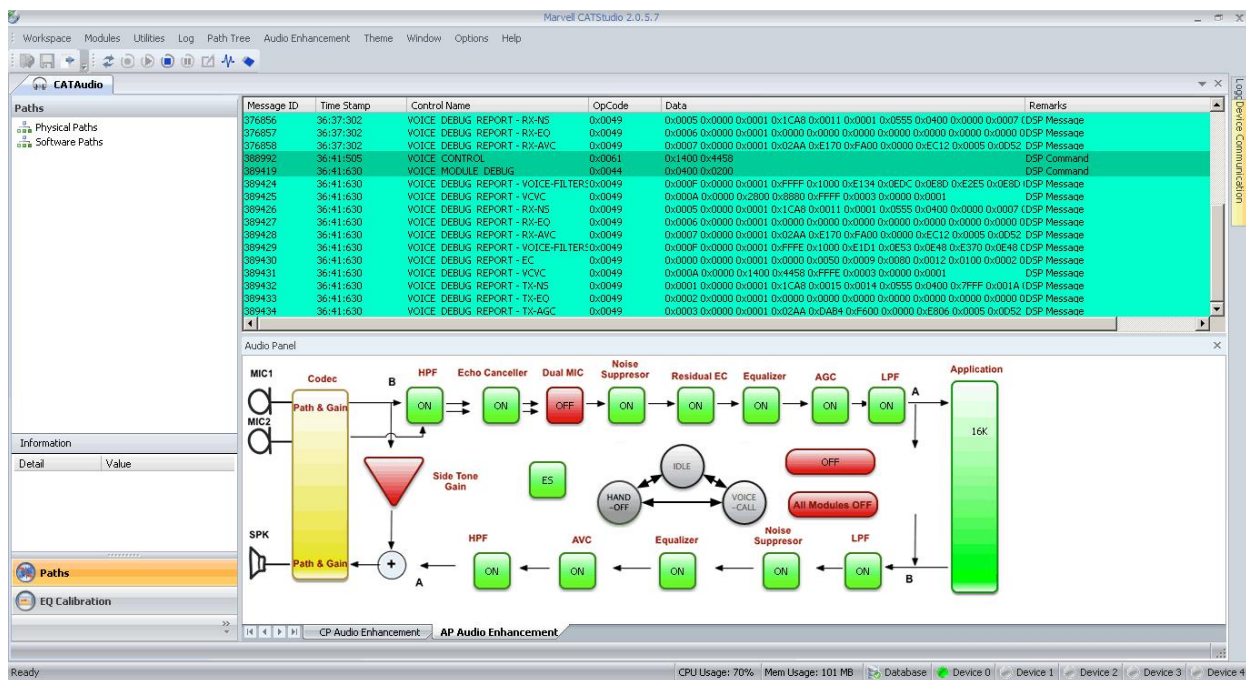
3 It only supports NS, EC, AVC, AGC, HPF, LPF modules. If you want tune EQ, you can refer to chapter Manually Updating Rx\Tx Equalizer Taps.

4.6 VOIP Tuning

1 Select the AP Audio Enhancement Panel.



2 Press **Record** Button to enable Audio debug mode.



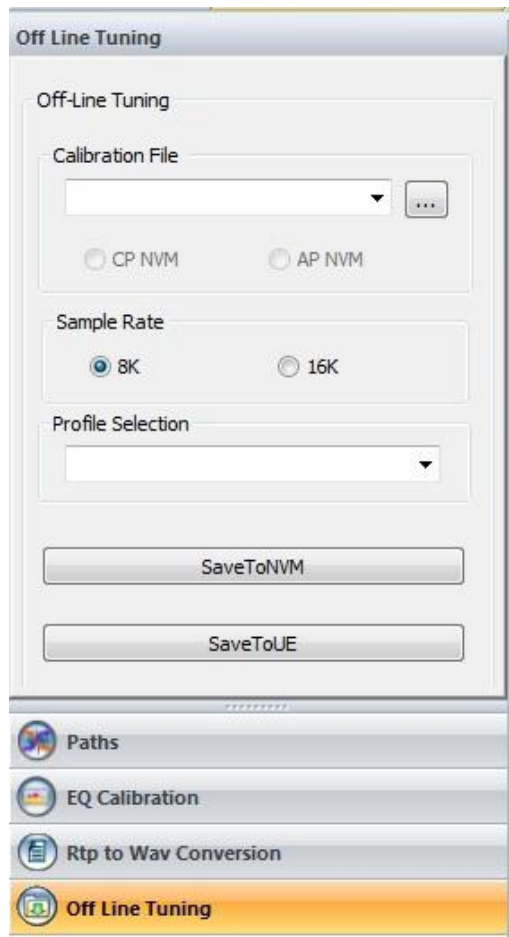
Message ID	Time Stamp	Control Name	OpCode	Data	Remarks
376856	36:37:302	VOICE DEBUG REPORT - RX-NS	0x0049	0x0005 0x0000 0x0001 0x1CA8 0x0011 0x0001 0x0555 0x0400 0x0000 0x0007	DSP Message
376857	36:37:302	VOICE DEBUG REPORT - RX-EO	0x0049	0x0006 0x0000 0x0001 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000	DSP Message
376858	36:37:302	VOICE DEBUG REPORT - RX-AVC	0x0049	0x0007 0x0000 0x0001 0x02AA 0xE170 0xFA00 0x0000 0xEC12 0x0005 0x0D52	DSP Message
388992	36:41:505	VOICE CONTROL	0x0061	0x1400 0x4458	DSP Command
388919	36:41:630	VOICE MODULE DEBUG	0x0044	0x0400 0x0200	DSP Command
389424	36:41:630	VOICE DEBUG REPORT - VOICE-FILTER	0x0049	0x000F 0x0000 0x0001 0xFFFF 0x1000 0xE134 0x0EDC 0x0E80 0xE2E5 0x0E80	DSP Message
389425	36:41:630	VOICE DEBUG REPORT - VCVC	0x0049	0x000A 0x0000 0x2800 0x8880 0xFFFF 0x0003 0x0000 0x0001	DSP Message
389426	36:41:630	VOICE DEBUG REPORT - RX-NS	0x0049	0x0005 0x0000 0x0001 0x1CA8 0x0011 0x0001 0x0555 0x0400 0x0000 0x0007	DSP Message
389427	36:41:630	VOICE DEBUG REPORT - RX-EO	0x0049	0x0006 0x0000 0x0001 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000	DSP Message
389428	36:41:630	VOICE DEBUG REPORT - RX-AVC	0x0049	0x0007 0x0000 0x0001 0x02AA 0xE170 0xFA00 0x0000 0xEC12 0x0005 0x0D52	DSP Message
389429	36:41:630	VOICE DEBUG REPORT - VOICE-FILTER	0x0049	0x000F 0x0000 0x0001 0xFFFF 0x1000 0xE1D1 0x0E53 0x0E49 0xE370 0x0E48	DSP Message
389430	36:41:630	VOICE DEBUG REPORT - EC	0x0049	0x0000 0x0000 0x0001 0x0000 0x0050 0x0009 0x0080 0x0012 0x0100 0x0002	DSP Message
389431	36:41:630	VOICE DEBUG REPORT - VCVC	0x0049	0x000A 0x0000 0x1400 0x4458 0xFFFF 0x0003 0x0000 0x0001	DSP Message
389432	36:41:630	VOICE DEBUG REPORT - TX-NS	0x0049	0x0001 0x0000 0x0001 0x1CA8 0x0015 0x0014 0x0555 0x0400 0x7FFF 0x001A	DSP Message
389433	36:41:630	VOICE DEBUG REPORT - TX-EO	0x0049	0x0002 0x0000 0x0001 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000 0x0000	DSP Message
389434	36:41:630	VOICE DEBUG REPORT - TX-AGC	0x0049	0x0003 0x0000 0x0001 0x02AA 0xDA84 0xF500 0x0000 0xE806 0x0005 0x0D52	DSP Message

3 Make a VOIP call by APP (such as Skype).

4. Tune parameters as VOICE CALL.

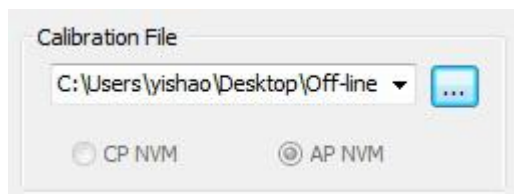
4.7 Offline Tuning

User also could tune voice call and VOIP enhancement in offline mode. But this function only work on single NVM solution.



1 Open Offline Panel

2 Select calibration file from PC, CAT-Audio will select CP NVM or AP NVM automatically.



3 Select related profile you want tune.



4 Tune parameter on VC or VOIP enhancement GUI and click send button.

Property	Value (Hex)
MasSupress	13
MaxNoise	-30
NoiseLevelINC	1638
NoiseLevelDEC	819
cnCeil	0
updateNoiseMatrix	30
Refvadhang	40
betaFest	-18
betaBrutal	-60
betaDT	-13
gammaDT	0
vmThresDT	120
burst	4
hann	8

Buttons: Send, Send and Save, Cancel

5 Clicks "SaveToNVM" button to save parameters to PC side. Or click "SaveToUE" button to save parameters to UE side in online mode.

Buttons: SaveToNVM, SaveToUE

4.8 Manually Updating Rx\Tx Equalizer Taps

This tool provides two main options:

- ☒ Loading the wrong frequency data file and using the the Matlab utility (With Matlab/MCR selection) fix the data.
- ☒ Loading 15 or 31 external calculated taps and writing them to the audio_eq.nvm & audio_MSMain.nvm files (Without Matlab/MCR selection).

Notice: Do not load FR files that have already passed.

Input Data Preparation

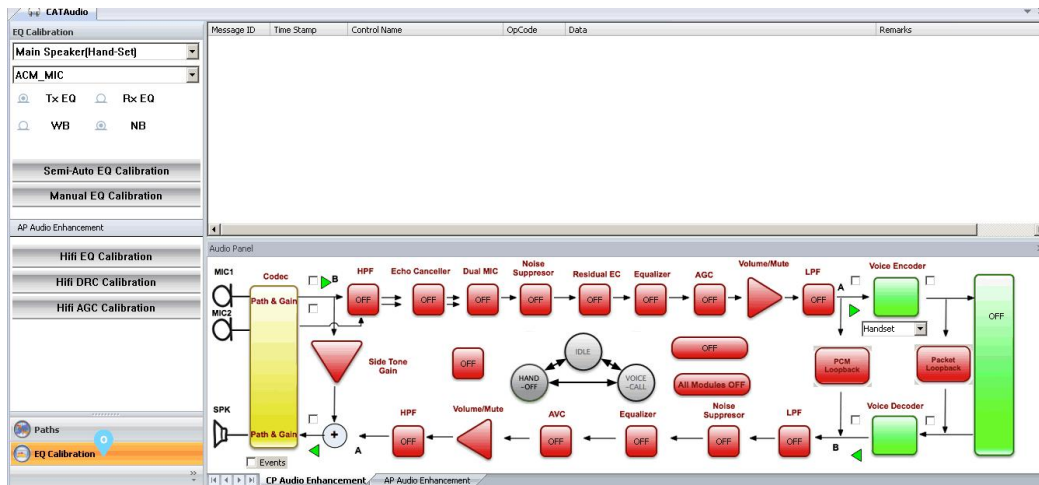
1. Run the SLR or RLR Nom and save the wrong FR data file.

Input Data Preparation with Matlab

1. Using this option (recommended) enables you to let the Manually Rx\Tx Equalizer utility fix the wrong FR.
2. Copy the FR data file from UPL (*.exp) or from UPV (*.tra) or ACQUA(*.txt) to the manual equalizer working directory, located in: \CAT-Studio\CT_Audio\EqualizerManCal 'EXP_RX_HS.exp for example.

Manually Rx\Tx Equalizer Procedure

1. Click [EQ calibration](#) to show EQ calibration and AP Audio Enhancement page.



2. In CAT-Audio Calibration Tap, select Test Setup..



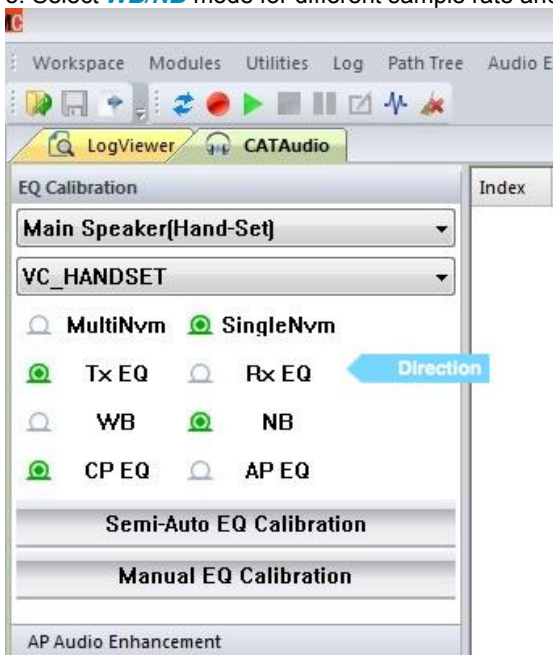
3. After the “Test Setup” selection, select Multi NVM or Single NVM solution for different platform. New platform such as PXA1908, PXA1928 and PXA1936 use single NVM solution.



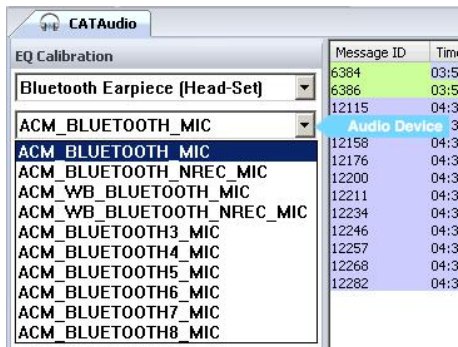
4. Select CP or AP EQ, CP EQ is for voice enhancement and AP EQ is for VOIP enhancement.



5. Select **WB/NB** mode for different sample rate and direction for Tx EQ or RX EQ..



4. After you selected “Test Setup” and “Tx-EQ or Rx-EQ”, CAT-Audio will select related audio device for test for multi NVM solution. If there are several audio devices in this test setting, you need select target audio device manually, such as Bluetooth EQ Calibration.



6. Single NVM solution will show audio profile, you need select related audio profile for EQ Calibration.

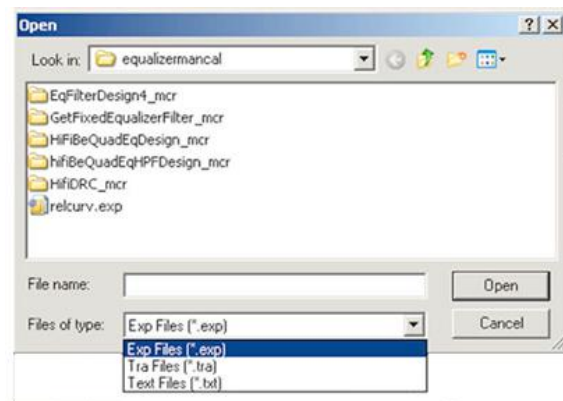


7. Click the Manual Equalizer Calibration or the Semi-Auto Equalizer Calibration button to start calibration.

Semi-Auto Equalizer Calibration with Matlab

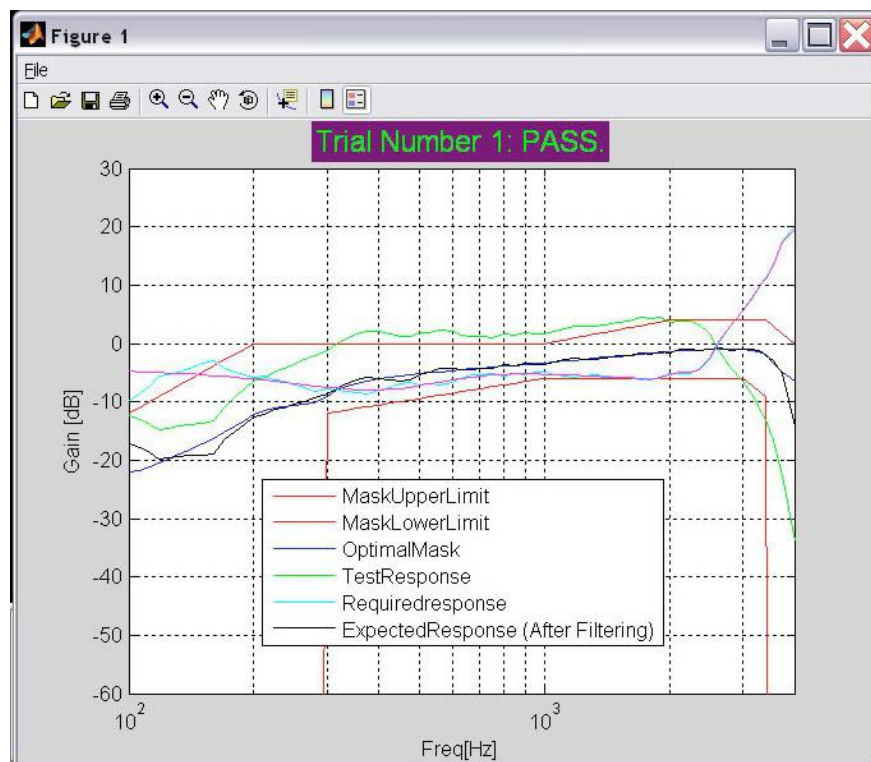
If YES is selected, the Matlab utility fixes the FR. The wrong FR data file is required. The browser is displayed and loads the wrong FR input file: CAT-Studio\CT_Audio\EqualizerManCal is the current and the working directory.

Load the input wrong FR file.

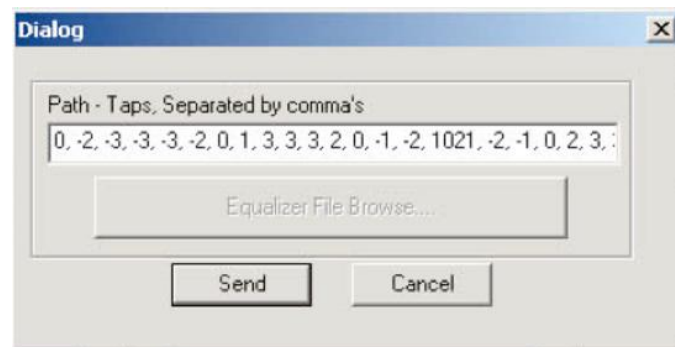


After few seconds, MatLab displays the plot of the updated FR and a message about whether it succeeded in fixing the data or not.

The Matlab plot:

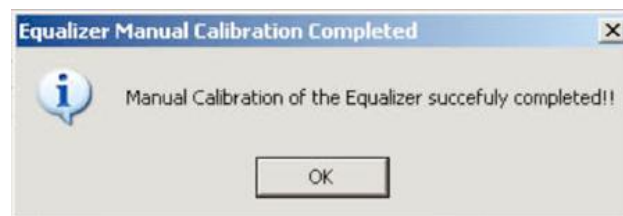


4. Close the plot.
5. Click the Send button to burn those taps to the audio_eq.nvm file (in the APPS flash explorer).

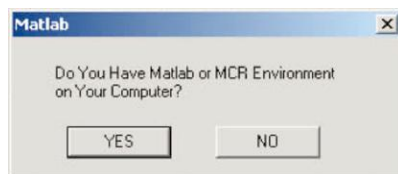


This procedure updates the relevant audio device pointer in audio_device.nvm file.

6. Ensure that the following confirmation message appears after the Send button has been clicked and click OK to terminate the process.



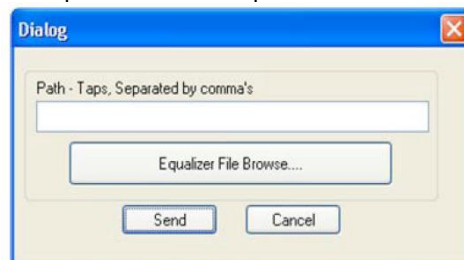
Semi-Auto Equalizer Calibration without Matlab

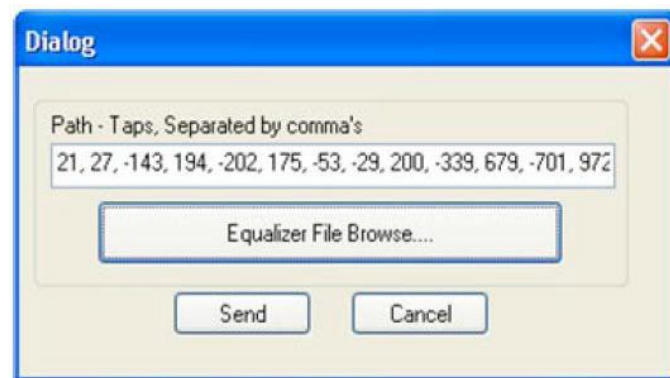
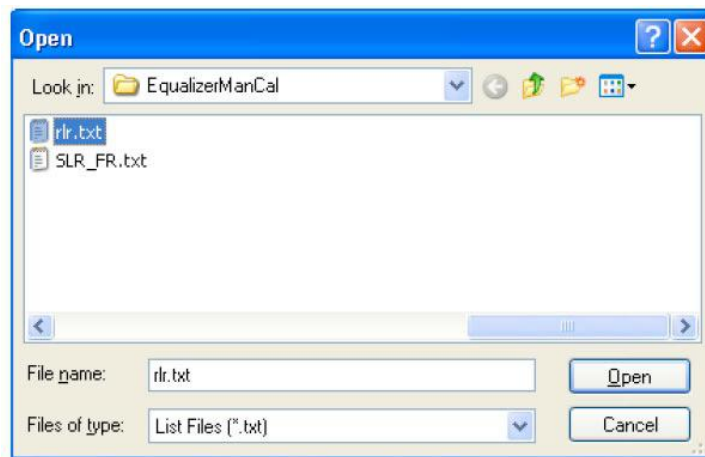


If NO is chosen, the Equalizer File Browse dialog box is displayed.

This section of Manual Equalizer assumes that you have your own utility for fixing equalizer response and you can load and burn your calculated equalizer taps to the Flash explorer.

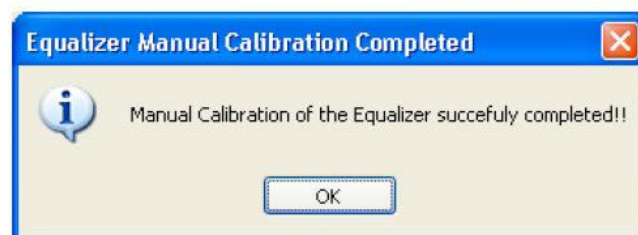
1. Click the Equalizer File Browse...





2 Ensure that the file is aligned and load the file with your calculated equalizer taps.

Ensure that the following confirmation message appears after the Send button has been clicked and click the OK



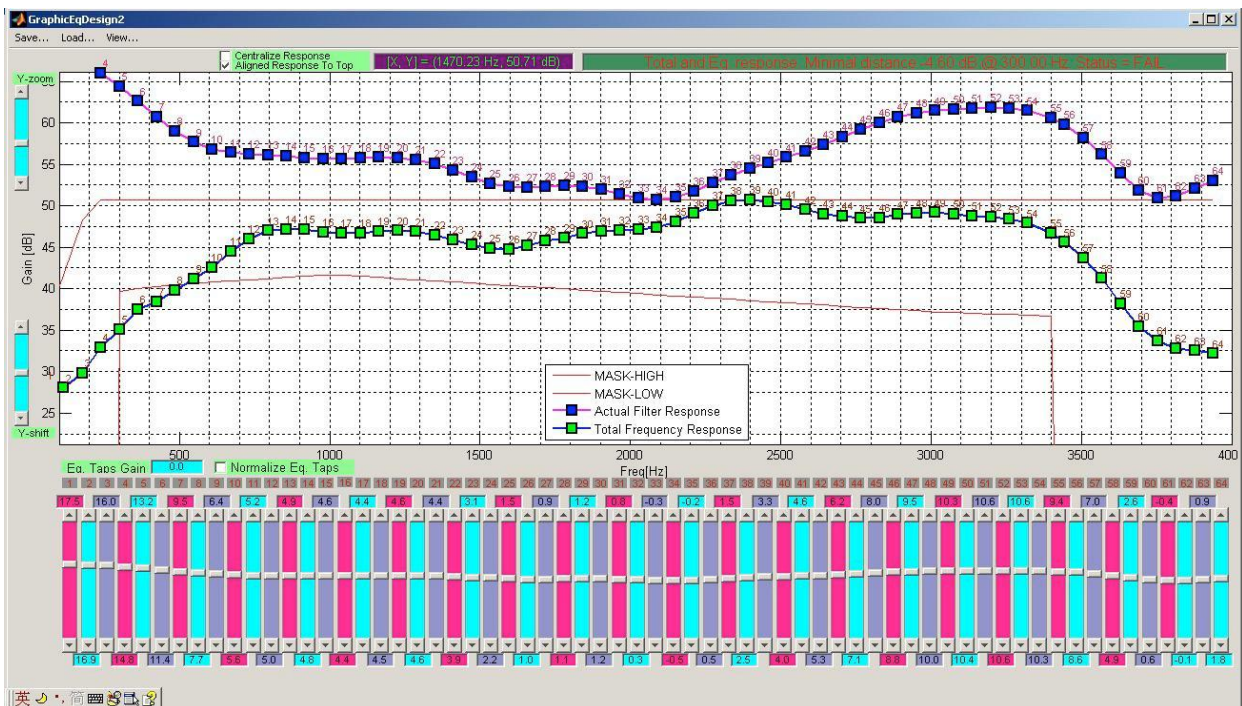
button to terminate the process

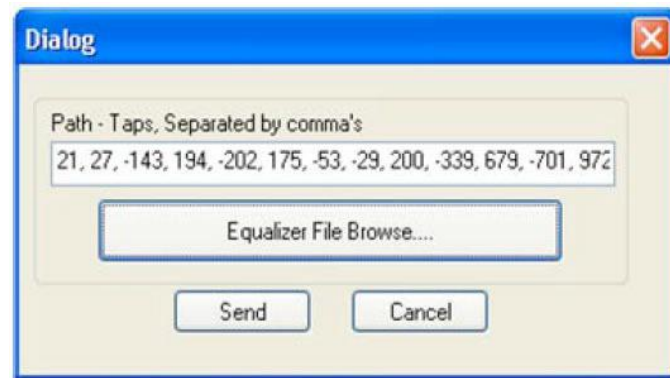
Manual Equalizer Calibration with Matlab

The procedure of manual equalizer is similar to Semi-Auto Equalizer Calibration. But you need click button "Manual Equalizer Calibration".

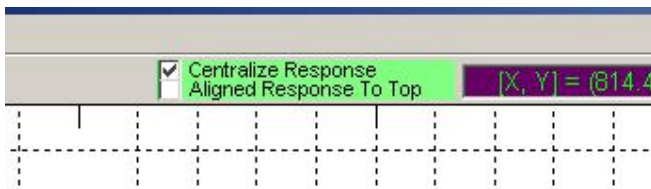


And it will pop up manual equalizer calibration interface.

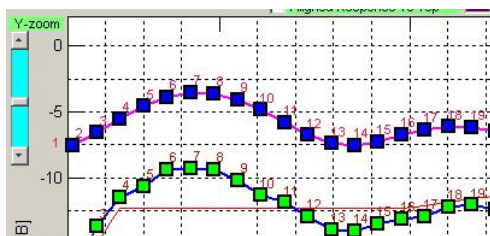




Check **Centralize Response** to let total frequency response in center.



Y-zoom slider bar let user could zoom in or zoom out Y axis.



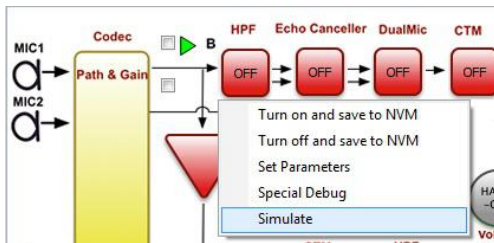
Y-shift slider bar let user could move graph in Y axis.

User can save Sliders Position to file for next time. And load Sliders Position from file.

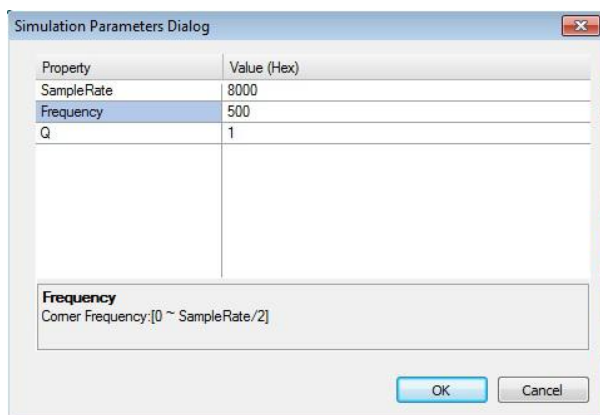


4.9 Voice Call HPF/LPF simulation

1 Right click on HPF or LPF module.



2 CAT-Audio will pop up a window. You can input parameter such as SampleRate, Frequency and Q.

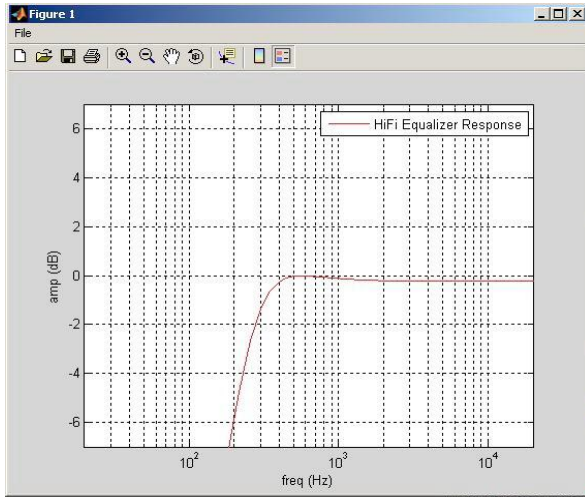


SampleRate: 8000 for NB voice call and 16000 for WB voice call.

Frequency: Corner Frequency.

Q: That adjustment in definition was made so that a boost of N dB followed by a cut of N dB for identical Q and Frequency/sample Frequency results in precisely flat unity gain filter.

3 Click OK, it will pop up a dialog show the frequency response.



4 When you close this dialog, the parameter will be set to next dialog, you can send then to UE by click “Send” or “Send and Save” button.

TX-High-Path Filter

Return Default Value Parameter Format

Property	Value (Hex)
A0_Coefficient	0x1000
A1_Coefficient	0xE72F
A2_Coefficient	0x0ADC
B0_Coefficient	0x0CEB
B1_Coefficient	0xE629
B2_Coefficient	0x0CEB

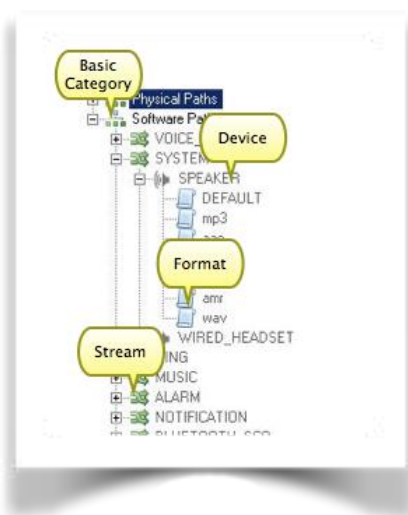
Send Send and Save Cancel

HIFI Calibration

AP use new ACM architecture for codec and software volume tuning.
Before software path tuning, user need tune fixed codec gain in Physical Paths.

5.1 Software Path Calibration

- 1) Define one software path by three category.
- 2) User could tuning every volume step of one software path.
- 3) Support default value tuning. User could tune a set of default value for stream and device. All of format with



same stream and device could share it. User can tune a set of value for special format as well.

Software Paths provides software gain tuning by three categories: stream, device and format.

Stream: Stream represent the function of current audio path. And the number of volume steps is based on stream. So it is set as first category.

Device: Device is hardware conception. Different hardware means different hardware gain. So it is set as second category.

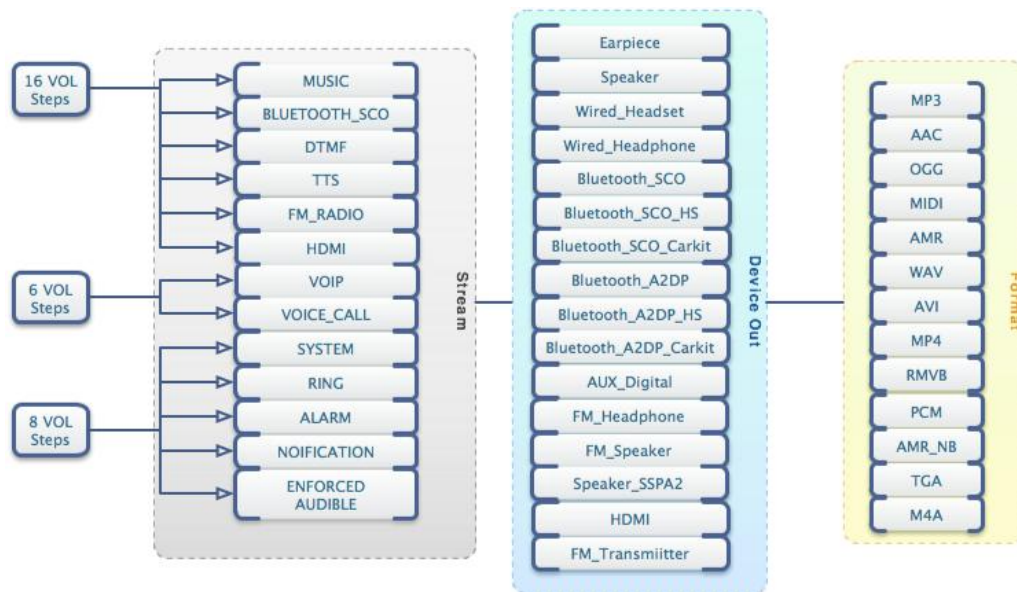
Format: Format is the file format saves on flash. It is last category.

Because only one software gain in one software path, we don't add extra gain item on the tree.

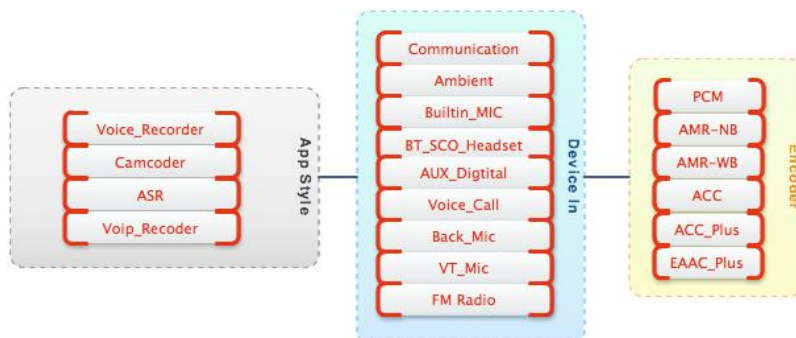
It composed with two parts: Playback and Records.

5.2 Software path directions

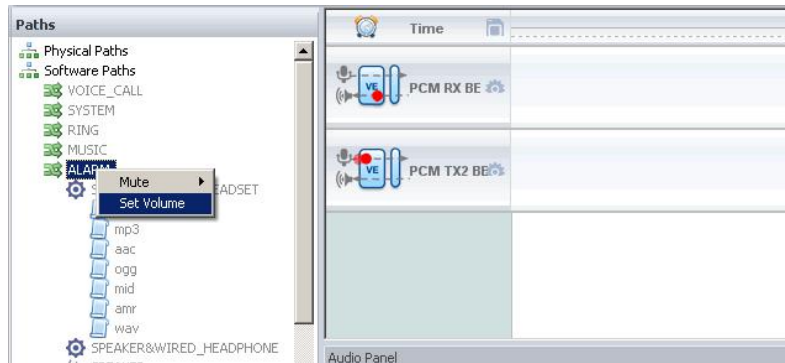
Playback direction



Record Direction

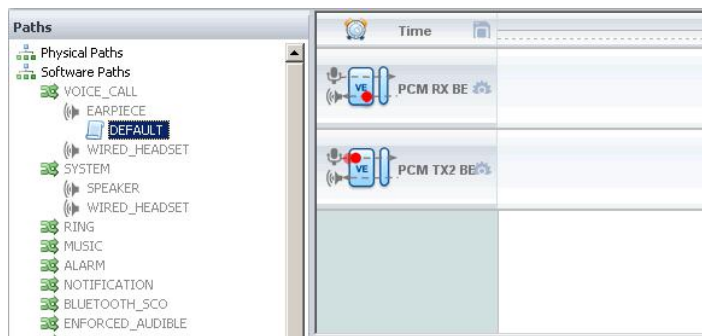


5.3 Software path operations



- 1) Select Steam and right click the item will pop up a menu. You can change the volume or mute on/off the stream.
- 2) Select the stream you can check the its state in state window.

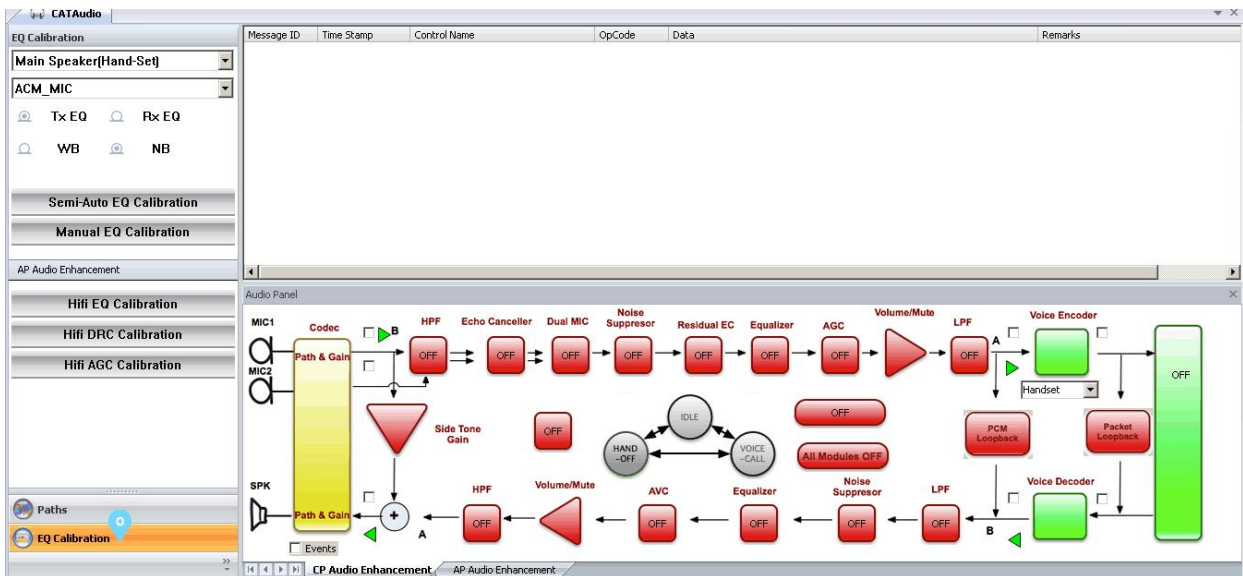
5.4 Software path tuning



- 1) Select Format/Encode Format, and right click item, you can tuning the software path value.
- 2) Click "OK" to save file to UE.

5.5 AP Audio Enhancement Page

Click [EQ calibration](#) to show AP Audio Enhancement page. It includes [Hifi EQ Calibration](#), [Hifi DRC Calibration](#) and [Hifi AGC Calibration](#).



5.6 HIFI EQ calibration

The screenshot shows the 'HiFi Eq Design Dialog' box. It has a 'Browse...' button and a 'Get Taps From PC File...' button. The 'HiFi Eq Design' section contains the following fields: Frequency (20-20kHz) set to 400, Gain [dB] set to 2, Q-Factor set to 0.5, Sample Rate set to 44100, and Filter Type set to PEAKING FILTER. There is a 'Run Taps Calculation' button. Below this, the 'BeQuad Taps' section has input fields for B0, B1, B2, A0, A1, and A2, all set to 0. A 'Band' dropdown menu is also present. A 'Control' dropdown menu is located below the 'Band' menu. A 'Clear All Fields' button is at the bottom right of the 'BeQuad Taps' section. A note states: 'Note: A0 is a Scale Factor for all Taps'. At the bottom, there is a 'Device' dropdown menu and three buttons: 'Write Taps to Flash', 'Read Taps From Flash', and 'Close'.

HiFi EQ Design Dialog

Custom can use this dialog to design HIFI BiQuad Equalizer.

Simulation

- Set Frequency, Gain and EQ.

HiFi Eq Design Dialog

Browse...
Get Taps From PC File...

HiFi Eq Design

Frequency (20-20k:Hz) 400 Gain [dB] 2
Q-Filter 0.5

Sample Rate 44100 Filter Type PEAKING FILTER

Run Taps Calculation

BeQuad Taps

B0 16589
B1 -31134
B2 14596
A0 14
A1 -31134
A2 14801

Note: A0 is a Scale Factor for all Taps

Band Band1

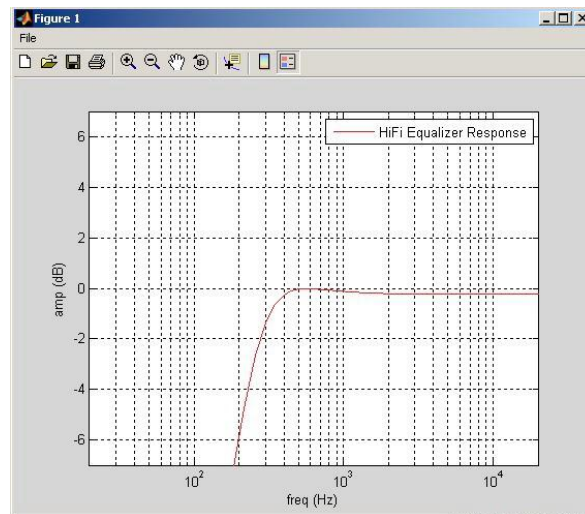
Control ON

Clear All Fields

Device AUDIO_DEVICE_OUT_SPEAKER

Write Taps to Flash Read Taps From Flash Close

- Select correct Sample Rate, we support 8000, 16000, 44.1k and 48k sample rate.
- Select Filter Type.
 - We support **peaking**, **high pass**, **low pass**, **high pass shelving**, **low pass shelving**, **band pass** and **notch** filter.
- Click “Run Taps Calculation” button, and then CAT-Audio will show a simulation figure as next one.



Write Taps to Flash.

- Close this figure to generate BiQuad parameters in the HIFI EQ Design Dialog
- Select “Band”, “Device” and “Control”
- Click button “Write Taps to Flash” to burn parameters to flash
- Select “Band” and “Device”
- Click button “Read Taps from Flash” to read parameters to dialog.

Audio Device

AUDIO_DEVICE_OUT_EARPIECE

AUDIO_DEVICE_OUT_SPEAKER

AUDIO_DEVICE_OUT_WIRED_HEADSET

AUDIO_DEVICE_OUT_WIRED_HEADPHONE

AUDIO_DEVICE_OUT_IN_BUILTIN_MIC

8 bands Multi EQ:

It support 8 bands multi EQ, you can select different band for tuning.

Filter Types

Peaking filter: Used for EQ

A peaking filter provides a boost or cut in the vicinity of center frequency.

Frequency: Center Frequency.

Gain: the gain of peaking, boost when gain > 0dB, cut when gain < 0dB.

Q: The bandwidth in octaves (between midpoint [dBgain/2] gain frequencies)

High pass shelving filer: Used for EQ

A *high pass shelving filter* adjusts the gain of higher frequencies while having no effect below its corner frequency.

Frequency: Corner Frequency.

Gain: the gain of shelving

Q: Shelf Slope

Low pass shelving filer: Used for EQ

A *low pass shelving filter* adjust the gain of lower frequencies while having no effect above its corner frequency.

Frequency: Corner Frequency.

Gain: the gain of shelving

Q: Shelf Slope

High pass filer: Used for IIR

Frequency: Corner Frequency.

Gain: NA.

Q: That adjustment in definition was made so that a boost of N dB followed by a cut of N dB for identical Q and Frequency/sample Frequency results in precisely flat unity gain filter.

Low pass filer: Used for IIR

Frequency: Corner Frequency.

Gain: NA.

Q: That adjustment in definition was made so that a boost of N dB followed by a cut of N dB for identical Q and Frequency/sample Frequency results in precisely flat unity gain filter.

Bands pass filer: Used for IIR

Frequency: Center Frequency.

Gain: NA

Q: The bandwidth in octaves.

Notch filer: Used for IIR

Frequency: Center Frequency.

Gain: N/A

Q: The bandwidth in octaves.

Biquad Taps Import

Taps "[b0,b1,b2,a0,a1,a2]" can be edit by user-self. And they are fixed-point 16bits signed data. a0 is used to save the scale factor of the other 5 taps. Formula to transfer float-point taps "[fb0, fb1,fb2,1,fa1,fa2]" to fixed-point taps "[b0,b1,b2,a0,a1,a2]" is illustrated as following. Float-point is designed by user-self.

$a0 = \text{floor}(\log_2(32768/\text{fbx}))$; fbx is the maximum absolute value of [fb0, fb1,fb2,1,fa1,fa2]

$b0 = \text{round}(\text{fb0} * 2^{a0})$;

$b1 = \text{round}(\text{fb1} * 2^{a0})$;

$b2 = \text{round}(\text{fb2} * 2^{a0})$;

$a1 = \text{round}(fa1 * 2^{a0});$

$a2 = \text{round}(fa2 * 2^{a0});$

After generating the fixed-point taps, user can fill in them to boxes and store them by clicking "Write Taps to Flash".

5.7 HIFI DRC Calculation

- Click button "HIFI DRC " to show "HIFI DRC Design Dialog"

HIFI DRC Design Dialog

Region0
☒ Compression ☐ Expansion Ratio: 1

Region1
☒ Compression ☐ Expansion Ratio: 1 Offset: 0
 Threshold1: -60 -100 0dB

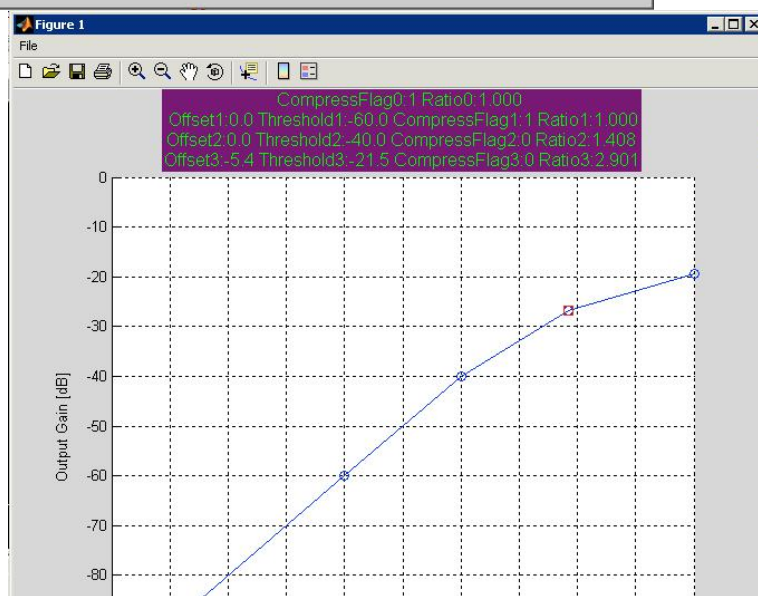
Region2
☒ Compression ☐ Expansion Ratio: 1 Offset: 0
 Threshold2: -40 -100 0dB

Region3
☒ Compression ☐ Expansion Ratio: 1 Offset: 0
 Threshold3: -30 -100 0dB

Attack Time(ms): 2 Release Time(ms): 400

Device
 AUDIO_DEVICE_OUT_SPEAKER

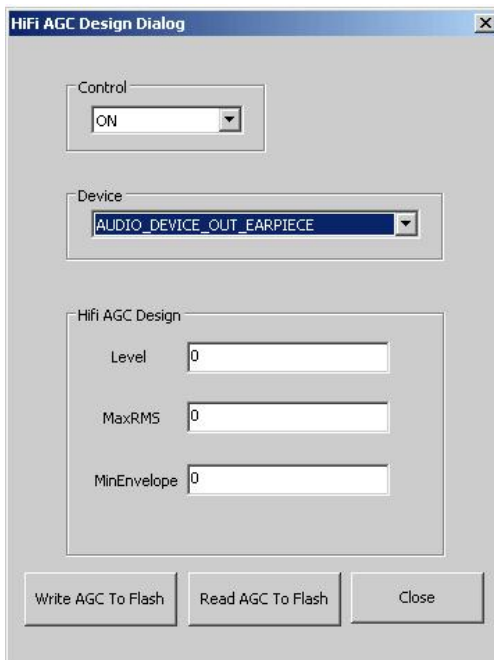
Run DRC Calculation Write DRC to Flash Read DRC From Flash



- Input “Attack time” and “Release time”
 - Select correct direction in combo box “Device”
 - Click button “Write DRC to Flash” to burn parameters to flash
- Currently, we only define 4 DRC sections.

5.8 HIFI AGC Calculation

- Select “Control” and “Device”.
- Input HIFI AGC parameters
- Click “Write AGC To Flash” to save AGC parameters.



The screenshot shows a software dialog box titled "HiFi AGC Design Dialog". It contains the following elements:

- Control:** A dropdown menu with "ON" selected.
- Device:** A dropdown menu with "AUDIO_DEVICE_OUT_EARPIECE" selected.
- HiFi AGC Design:** A section containing three input fields:
 - Level:** A text box with the value "0".
 - MaxRMS:** A text box with the value "0".
 - MinEnvelope:** A text box with the value "0".
- Buttons:** Three buttons at the bottom: "Write AGC To Flash", "Read AGC To Flash", and "Close".

Audio Calibration Procedures

The following section describes the flow and the specific procedures required to perform the calibration tests. The order of the calibration test cases is listed in [Table](#).

Users who have different paths need to follow instructions accurately to replace Tx and Rx path according to their configuration. For example, a Head set path should be:

- ☒ Tx Path – VoiceRecordFromMic1
- ☒ Rx Path – VoicePlayToEarphone
- ☒ Side Tone – LEVANTE_IN_L_TO_OUT for platform (920s) . In 978 or later platform, there is a side tone gain in Rx Path.

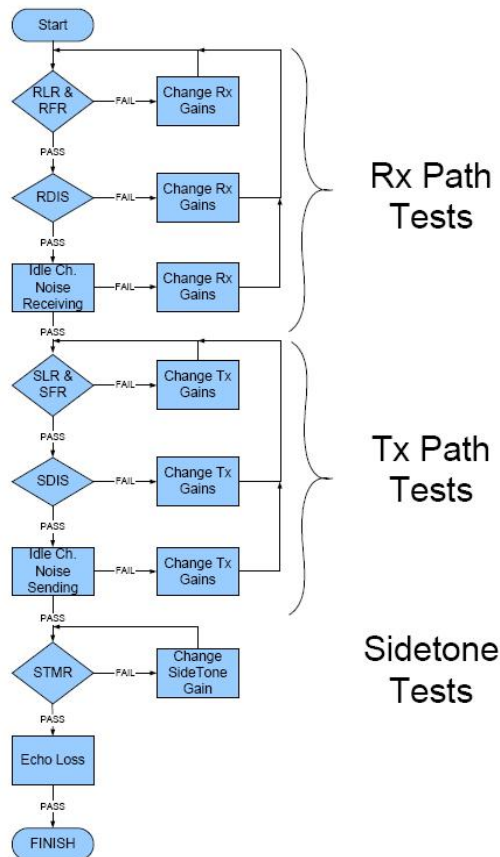
Table4: Test cases

Test Case	Acoustic Tests of GSM Mobile Phone	TS 51.010 GSM Rel. 99
1	Sending frequency response	30.1
2	Sending loudness rating	30.2
3	Sending distortion	30.7.1
4	Sending Idle channel noise	30.10.1
5	Side tone masking rating	30.5
6	Receiving frequency response	30.3
7	Receiving loudness rating	30.4
8	Receiving distortion	30.7.2
9	Receiving Idle channel noise	30.10.2
10	Echo loss	30.6.1
11	Stability margin	30.6.2
12	Ambient noise rejection	30.11

6.1 Audio Calibration Workflow

Tests are basically divided into three main groups that require gains change in order to pass them. The flowchart demonstrates this principal meaning, a gain change that occurred causes the an already passed test to be run again to make sure it still passes with current gain configuration.

Audio Calibration Workflow Diagram



6.1.1 Gain Calibration Methodology

This section describes the methodology of the gain calibration procedures. The methodology consists of the following basic guidelines:

- Device separation
- Independent audio path gain ID
- Maximum digital dynamic range

6.1.2 Independent Audio Path Gain ID

Every possible audio path is assigned a unique ID number. Each audio path contains an analog gain and a digital gain. Each path is assigned to a specific device.

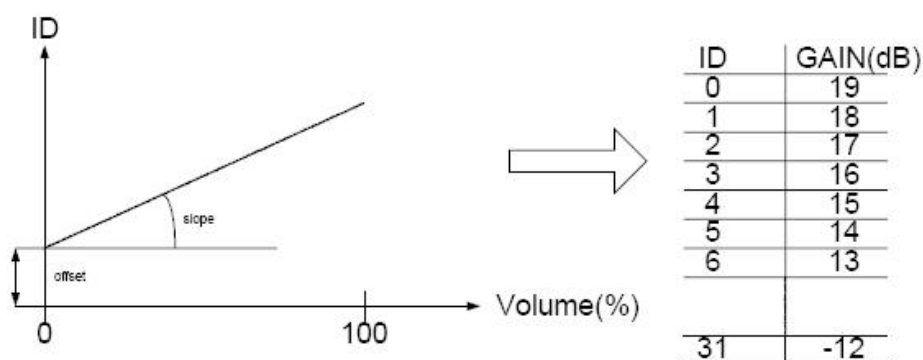
6.1.3 Maximum Digital Dynamic Range

The aim of the calibration procedure is to allow the maximum dynamic range that can be achieved on the Rx Path by digital alteration, without changing the analog gain. The purpose is to minimize the influence of analog gain changes on the voice enhancement modules.

6.1.4 Slope and Offset

The dynamic range of each analog and digital gain block is defined as a line over the Cartesian domain. This line is an index to a table in the UE database that holds the true gain of the block. As such, it holds two unique identifiers, the slope and offset. So for each gain block changed in the calibrated Rx path two equations need to be solved.

The Formula $\text{Slope} \times \text{Volume} + \text{Offset} = \text{ID}$ gives the correct index to the gain path ID in the corresponding gain



conversion table.

The index must be an integer, and the Audio Calibration calculates the nearest integer to the formula result. On the Tx Path, these parameters are not required because the user has no control over the Tx Path volume, which eliminates the need to allow operation over a wide range of signal levels. Tx path lanes always use slope=0 since no volume exists.

6.1.5 Volume Steps

In NEVO project, CAT-Audio use "Volume Steps" conception for audio gain tuning.

It is just a table for different volume percentage. You should input every value for different volume percentage.

Table 5: Volume Steps

Volume Percentage	Volume Value
10%	-36 dB
20%	-32 dB
30%	-28 dB
40%	-24 dB
50%	-20db

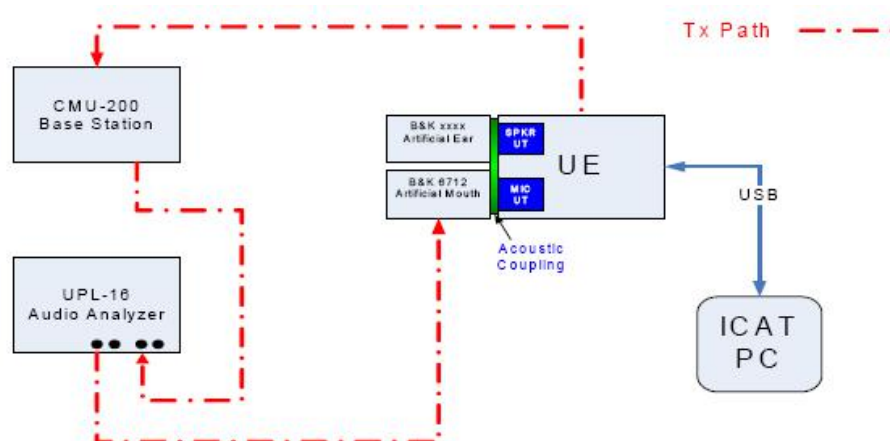
60%	-16 dB
70%	-12 dB
80%	-8dB
90%	-4 dB
100%	0dB

6.2 Calibration Procedures

6.2.1 Sending Loudness Rating & Frequency Response

2.1.1 Definition

Sending Loudness Rating (SLR) is a means to express the sending frequency response based on objective single tone measurements in a way that relates to how a speech signal would be perceived by a listener.



2.1.2 Setup Schematic

Sending Loudness Rating – Setup

2.1.3 Procedure

To run the Sending Loudness Rating test case, perform the following steps:

1. On the UPL or UPV instrument, run SLR test case.
2. Wait for the test to end.
3. View the results on the UPL-16 or UPV monitor. If the test case fails, continue to step 4.
4. In the Audio Configuration Path window, select the LEVANTE_AnaMIC1_TO_PCM_L and set *Digital Gain*

parameter to ID=0 (equals 0dB).

5. Repeat step 5 while changing the Digital Gain ID but no more than 8dB gain. If the test still does not pass, repeat step 4 changing the analog gain ID.

6. If Sending Frequency Response Test didn't pass, move to step 8.

7. Save SLR test results to a USB disk.

8. Update equalizer Tx taps (refer to [Section 3.1.5](#)). (Assuming Matlab(R) or MCR installed).

9. On the UPL-16 or UPV instrument, test SLR.

10. Wait for the test to end.

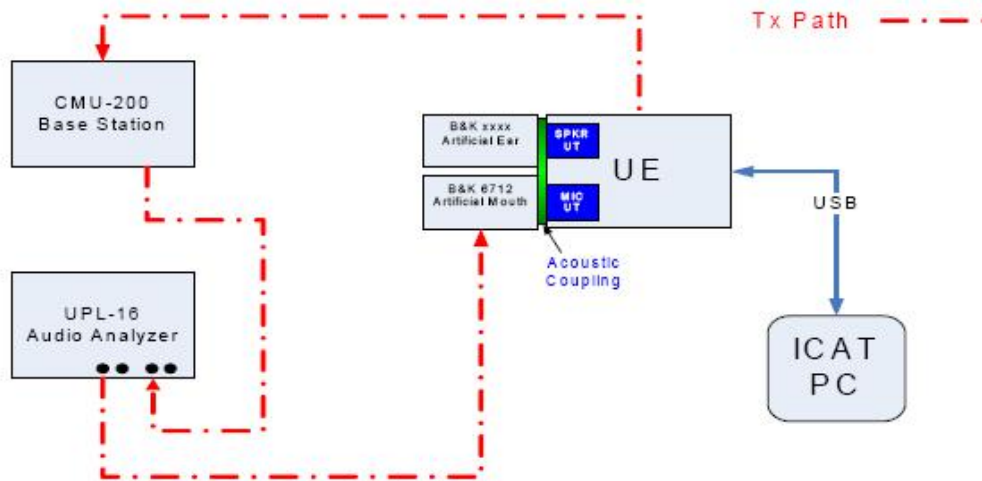


6.2.2 Sending Distortion

2.2.1 Definition

The transmit signal to total distortion ratio is a means to measure the linearity of the transmitter equipment (excluding the speech codec).

2.2.2 Setup Schematic



Sending Distortion - Setup

2.2.3 Procedure

1 Run Sending Distortion on UPV or UPL-16

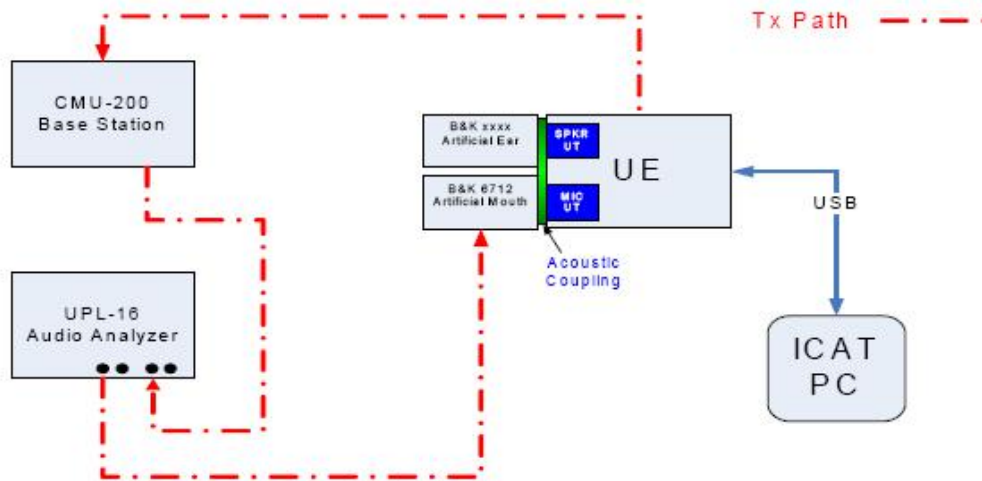
Note: To improve performance, decrease Tx analog gain, but keep the total gain (analog and digital) as calibrated during Sending Loudness Rating test case step.

6.2.3 Sending Idle Channel Noise

2.3.1 Definition

The idle channel noise in sending direction is equivalent to the noise level produced at the output of the reference speech decoder of the SS (System Simulator, that is, the CMU-200), when the mouth reference point is in a quiet environment.

2.3.2 Setup Schematic



Sending Idle Channel Noise - Setup

2.3.3

Procedure

- 1 Run Sending Idle Channel Noise test case

6.2.4 Side Tone Masking Rating

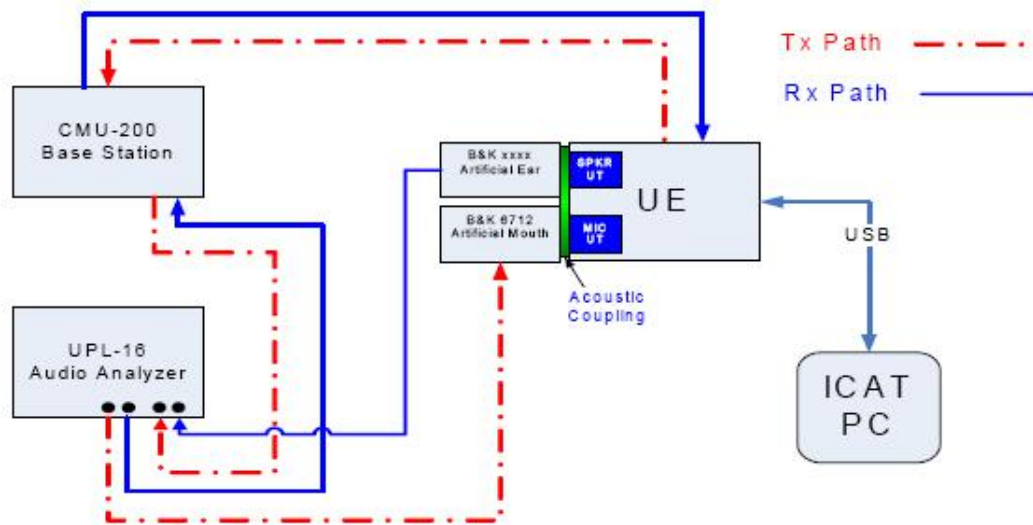
2.4.1

Definition

Side tone loudness ratings are a means to express the path loss from the artificial mouth to the artificial ear based on objective single tone measurements in a way that relates to how a speaker will perceive his own voice when speaking (talker side tone, expressed by the Side Tone Masking Rating - STMR), or how a listener will perceive the background noise picked up by the microphone (Listener Side Tone Rating - LSTR).

2.4.2

Setup Schematic



2.4.3 Procedure

To run the Side Tone Masking Rating test case, perform the following steps:

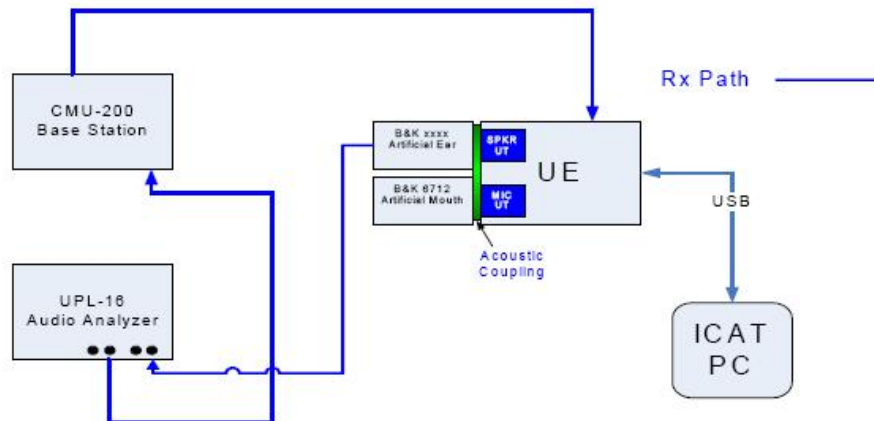
1. Enable the LEVANTE_IN_L_TO_OUT path, set volume to 0%, and then click *Send*.
2. On the UPL-16 or UPV instrument, run Side Tone Masking Rating test case.
3. Wait for the test to end.
4. If the test fails, change the Side Tone ST_GAIN_L offset parameter until the test passes.

6.2.5 Receiving Loudness Rating & Frequency Response

2.5.1 Definition

Receiving Loudness Rating (RLR) is a means to express the receiving frequency response based on objective single tone measurements in a way that relates to how a speech signal would be perceived by a listener.

2.5.2 Setup Schematic



2.5.3 Procedure (Volume Steps)

To run the Receiving Loudness Rating test case, perform the following steps:

1. Make a call to enable VoicePlayToEarphone and set the volume level to 50% in the Set Volume path window.
2. Make sure all gain for this path are set to 0dB. If they are not, change the appropriate gain (dB).
3. On the ACQUA or UPV instrument, run RLR Normal Volume test case.
4. Wait for the test to end.
5. View the results on the ACQUA or UPV monitor. If the test case fails, continue to step 7, if not continue to step 9.
6. Change the digital gain offset (no more than 0dB) in 50% and repeats the test until it passes.
7. If digital gain was not enough change DPGA and PGA gain for that path.
8. Enable VoicePlayToEarphone and set the volume level to 100% in the Set Volume path window.
9. On the ACQUA or UPV instrument, run RLR Max Volume test case.
10. Wait for the test to end.
11. View the results on the ACQUA or UPV monitor. If the test case fails, continue to step 11.
12. Change the digital gain in 100% until the test passes (no more than 0dB).
13. If digital gain was not enough change DPGA and PGA gain for that path.
14. Set the volume of path VoicePlayToEarphone to 0% in the Set Volume path window.
15. Run RLR Min Volume test on the ACQUA or UPV instrument
16. Change the digital gain offset (no more than 0dB) in 0% and repeats the test until it passes.
17. If digital gain was not enough change DPGA and PGA gain for that path.
Input other gain for other volume steps.
18. Enable VoicePlayToEarphone and set the volume level to 50% in the Set Volume path window.
19. On the ACQUA or UPV instrument, run RLR Normal Volume test case.
20. Wait for the test to end and make sure it passes.
21. Make sure you leave this path with a 50% volume setting. It is essential for upcoming tests.
22. At this point we stop to see if Receiving Frequency Response test passed. If not continue to step 21. If it did this test case is complete.
23. Save RLR test results to a USB disk.
24. Update equalizer Rx taps (refer to [Section 3.1.5](#)). (Assuming Matlab(R) or MCR installed).
25. On the UPL-16 or UPV instrument, press the RLR Normal Volume test case.
26. Wait for the test to end.
27. If all pass, test is complete.



6.2.6 Receiving Distortion

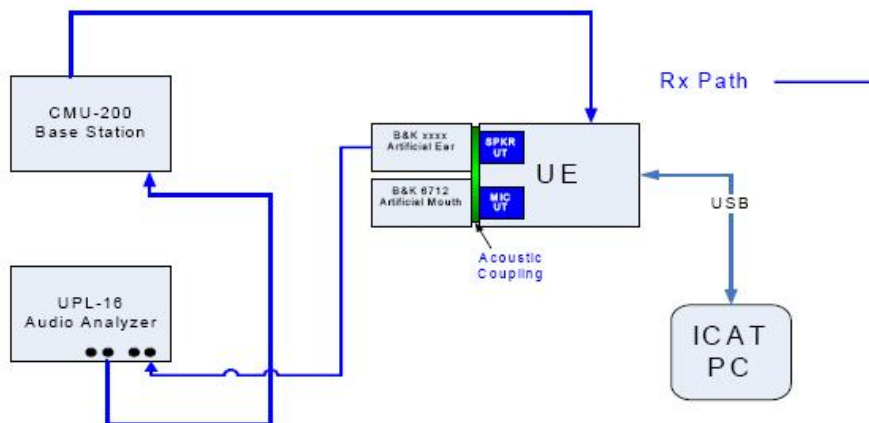
2.6.1

Definition

The receive signal to total distortion ratio is a means to measure the linearity in the receive equipment (excluding the speech decoder).

2.6.2

Setup Schematic



2.6.3

Procedure

Run receive distortion test case.

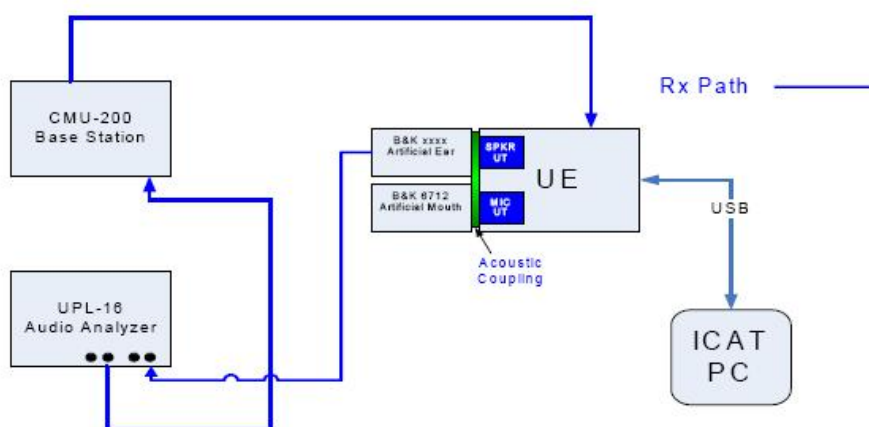
Note: To improve performance, decrease Rx analog gain, but keep the total gain (analog and digital) as calibrated during Receiving Loudness Rating test case step.

6.2.7 Receiving Idle Channel Noise

2.7.1

Definition

The idle channel noise in receiving direction is the acoustic sound pressure in the artificial ear when the digital input signal does not exist.



2.7.2

Setup Schematic

2.7.3

Procedure

Run Receiving Idle Channel Noise

6.2.8 Echo Loss

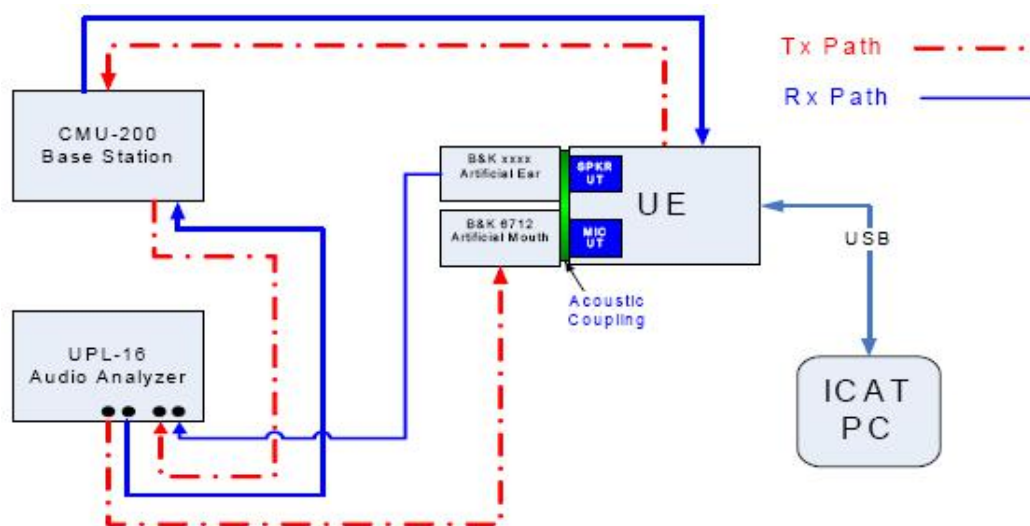
2.8.1

Definition

Echo loss is the path loss from the input of the reference speech encoder of the SS (System Simulator, that is, the CMU-200) to the output of the reference speech decoder of the SS.

2.8.2

Setup Schematic



2.8.3

Procedure

Run ECHO LOSS without training test case.

Run ECHO LOSS with training test case.

Note: The echo canceller is activated by default.

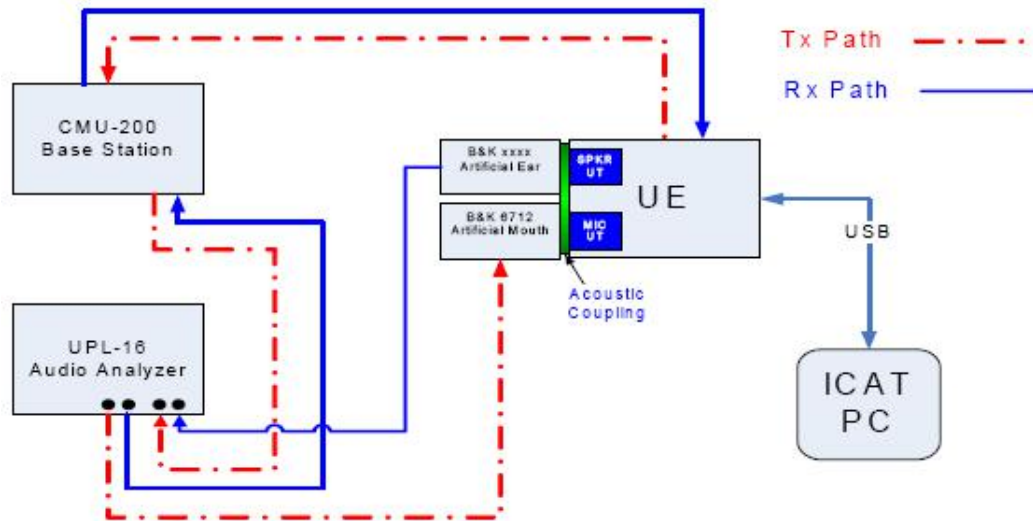
6.2.9 Stability Margin

2.9.1

Definition

The transmit-receive stability margin is a measure of the gain that would have to be inserted between the go and return paths of the reference speech coder in the system simulator for oscillation to occur.

2.9.2 Setup Schematic



2.9.3 Procedure

To run a Stability Margin test case, perform the following steps:

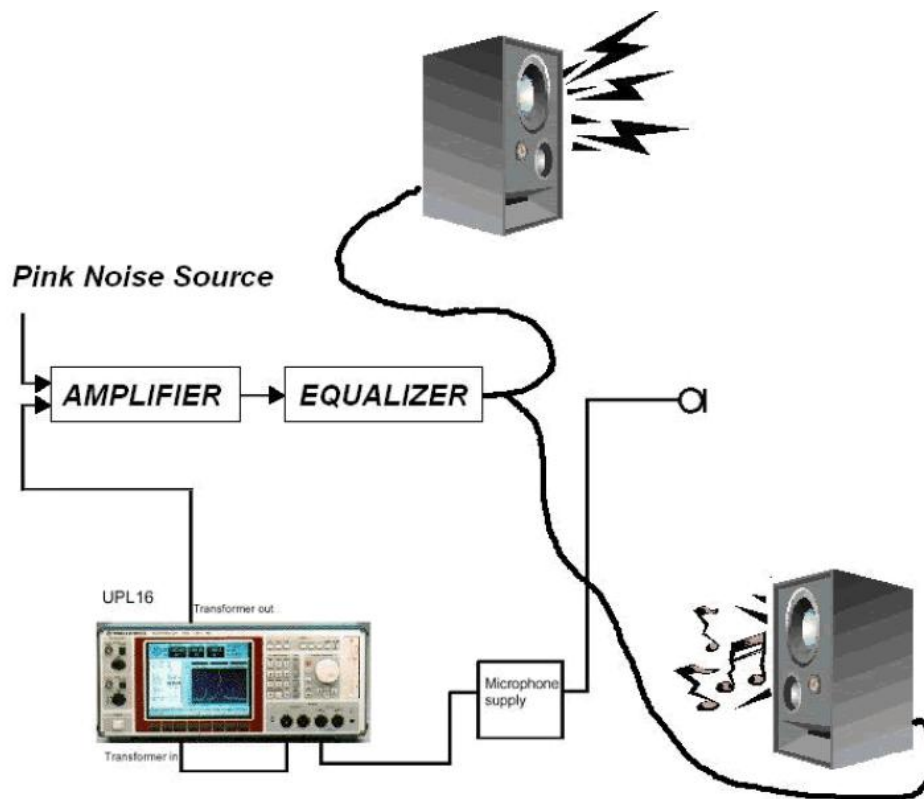
1. Disconnect the UE from the setup.
2. Place the UE speaker and microphone on a rigid flat surface.
3. Make sure that no audible oscillation is heard.
4. On the UPL-16 or UPV, run Stability Margin.

Note: There is no specific parameter improving performance of the Stability Margin test result.

6.2.10 Ambient Noise Rejection

2.10.1 Definition

Ambient noise rejection (ANR) describes the weighted ratio of voice signal transmission to ambient noise. An ANR value >0 dB means that voice as the useful signal is transmitted more loudly than any ambient noise.



2.10.2 Setup Schematic

2.10.3 Procedure

To run an Ambient Noise Rejection test case, perform the following steps:

1. Make sure a 31-band equalizer and a pink noise CD are available.
2. Upon first time use, calibrate the environment as stated in the UPL-16 or UPV manual. In other cases, make sure that the speakers are activated.
3. Run the test case, follow the UPL-16 or UPV on-screen instructions, and view the results on the UPL-16 or UPV monitor.

Noise suppression is activated by default.

7 Calibration Files

7.1 General

NVM audio files are generated during the audio calibration process.

When calibration files do not exist, then the default values which have been hard coded in the software are used.

Refer to the CT Hall Tool Set User and Reference Guide for more information regarding audio calibration tools.

Default NVM files can be generated when explicitly requested by CAT-Studio command.

Table 16 summarizes the NVM audio files. Each file is briefly described in the following subsections.

Table 6: NVM Files (PXA920s)

NVM Files		
File	Location	Remarks
audio_levante.nvm	CP folder	Codec calibration file
audio_MSAmain.nvm	CP folder	Generated by Calibration tools (mostly uses default values)
audio_ec.nvm	CP folder	Echo loss module calibration data
audio_ns.nvm	CP folder	Noise Suppression module calibration data
audio_avc.nvm	CP folder	AVC module calibration data
audio_HLPF.nvm	CP folder	High pass and low pass filter calibration data
audio_DualMic.nvm	CP folder	Dual Mic Calibration data
audio_eq.nvm	CP folder	Equalizer calibration data
audio_ctm.nvm	CP folder	Generated by Calibration tools
audio_hifi_eq.nvm	CP folder	For HIFI Equalizer
audio_hifi_volume.nvm	CP folder	Software volume calibration file

audio_levante.nvm saves audio path information and calibrated gain for each path.

Table 7: NVM Files (PXA978)

NVM Files		
File	Location	Remarks
audio_gain_calibration.xml	CP folder	Codec calibration file
CP NVM files		
File	Location	Comments
audio_MSAmain.nvm	CP folder	MSA calibration data index table
audio_ec.nvm	CP folder	Echo loss module calibration data
audio_ns.nvm	CP folder	Noise Suppression module calibration data
audio_avc.nvm	CP folder	AVC module calibration data
audio_HLPF.nvm	CP folder	High pass and low pass filter calibration data
audio_DualMic.nvm	CP folder	Dual MIC Calibration data
audio_eq.nvm	CP folder	Equalizer calibration data
audio_ctm.nvm	CP folder	CTM data

Table 8: NVM Files (PXA1920)

AP NVM files		
File	Location	Comments
audio_MSAmain_ap.nvm	AP folder	MSA calibration data index table
audio_ec_ap.nvm	AP folder	Echo loss module calibration data
audio_ns_ap.nvm	AP folder	Noise Suppression module calibration data
audio_avc_ap.nvm	AP folder	AVC module calibration data
audio_HLPF_ap.nvm	AP folder	High pass and low pass filter calibration data
audio_DualMic_ap.nvm	AP folder	Dual MIC Calibration data
audio_eq_ap.nvm	AP folder	Equalizer calibration data
CP NVM files		
File	Location	Comments
audio_voice_calibration.nvm	CP folder	Voice call calibration data
audio_voice_calibration_ap.nvm	AP folder	VOIP calibration data
audio_gain_calibration.xml	AP folder	Audio Gain Calibration Data
audio_swvol_calibration.xml	AP folder	Software Volume Calibration Data
audio_effect_config.xml	AP folder	HIFI Audio Enhancement Calibration Data

**Table 9:
NVM Files
(PXA1908
PXA1928
PXA1936)**

7.2 MSA Audio Path Parameters

Figure 1 shows the NVM data parameters table structures that configure the MSA audio path. This configuration is held in the device handle

ACM_DeviceTable in Figure 1 maps devices to components, paths, and entries to NVM calibration data as given in the subclasses below.

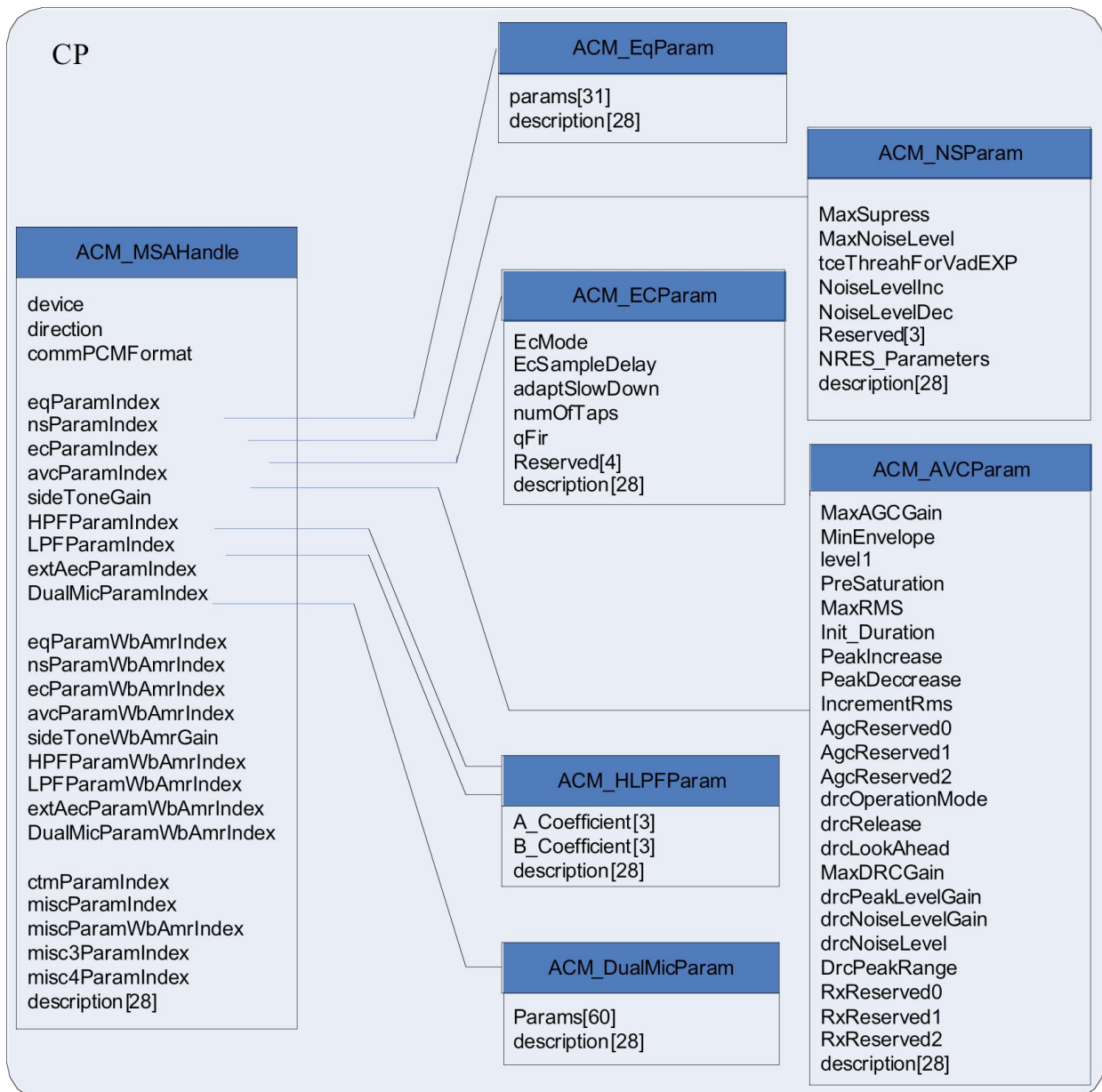


Figure 1: MSA Audio Paths NVM Tables

- 1) In ACM_DeviceHandle, Side Tone and CTM parameter indexes are set to 0xffff (turn off).

We suggest using Codec side tone, not MSA digital side tone.

- 2) MSA disable side tone and CTM
- 3) Most of 0 and 1 paramIndex is used for default value, so please check which paramIndex could use carefully.
- 4) The parameter index is 0xffff means the module is turned off

7.2.1 audio_MSAMain.nvm

Table 8: ACM_MSAMain Structure

ACM_MSAMain Structure		
Member	Type/Range	Notes
device	ACM_AudioDevice	
direction	ACM_PathDirection	
commPcmFormat	ACM_CommPcmFormat	
eqParamIndex	unsigned long	NB parameter index for EQ NVM .
nsParamIndex	unsigned long	NB parameter index for NS NVM .
ecParamIndex	unsigned long	NB parameter index for EC NVM .
avcParamIndex	unsigned long	NB parameter index for AVC NVM .
sideToneGain	unsigned long	NB parameter index for side tone NVM .
HPFParamIndex	unsigned long	NB parameter index for HPF NVM .
LPFParamIndex	unsigned long	NB parameter index for LPF NVM .
extAecParamIndex	unsigned long	Reserved parameter index
DualMicParamIndex	unsigned long	NB parameter index for Dual MIC
eqParamWbAmrIndex	unsigned long	WB parameter index for EQ NVM .
nsParamWbAmrIndex	unsigned long	WB parameter index for NS NVM .
ecParamWbAmrIndex	unsigned long	WB parameter index for EC NVM .
avcParamWbAmrIndex	unsigned long	WB parameter index for AVC NVM .
sideToneWbAmrGain	unsigned long	WB parameter index for side tone NVM .
HLPFParamWbAmrIndex	unsigned long	WB parameter index for HLPF NVM .
LPFParamWbAmrIndex	unsigned long	
extAecParamWbAmrIndex	unsigned long	Reserved parameter index
DualMicParamWbAmrIndex	unsigned long	
miscParamIndex	unsigned long	Reserved misc parameter index

miscParamWbAmrIndex	unsigned long	Reserved misc parameter index
misc3ParamIndex	unsigned long	Reserved misc parameter index
misc4ParamIndex	unsigned long	Reserved misc parameter index
description[32]	unsigned char	

See [Section 4.4.3. ACM_MSAHandle](#) for a description of the MSA device handler and the ACM_MSAHandle structure.

The last entry of audio_MSAmain.nvm MUST set device=ACM_NOT_CONNECTED.

ASR Modem can be set to different behavior using configuration NVM ([audio_config.nvm](#)).

Table 9: ACM_Appliance and Audio Device

Audio Device Map		
Audio Mode	ACM Audio Device	MSA EC Mode
Handset	ACM_MAIN_SPEAKER , ACM_MIC	Ear Seal
Hands free	ACM_AUX_SPEAKER , ACM_AUX_MIC	Speaker Phone
Headset	ACM_HEADSET_SPEAKER, ACM_HEADSET_MIC	Ear Seal
Bluetooth	ACM_BLUETOOTH_SPEAKER , ACM_BLUETOOTH_MIC	Ear Seal
Bluetooth_NREC	ACM_BLUETOOTH_NREC_SPEAKER , ACM_BLUETOOTH_NREC_MIC	Ear Seal
Handset_Loop	ACM_MAIN_SPEAKER__LOOP, ACM_MIC__LOOP	Ear Seal
Hands free_Loop	ACM_AUX_SPEAKER__LOOP, ACM_AUX_MIC__LOOP	Speaker Phone
Headset_Loop	ACM_HEADSET_SPEAKER__LOOP, ACM_HEADSET_MIC__LOOP	Ear Seal
Bluetooth_Loop	ACM_BLUETOOTH_SPEAKER__LOOP, ACM_BLUETOOTH_MIC__LOOP	Ear Seal
Dual Mic: Handset	ACM_MAIN_SPEAKER_DUALMIC , ACM_MIC_DUALMIC	Ear Seal
Dual Mic: Hands free	ACM_AUX_SPEAKER_DUALMIC , ACM_AUX_MIC_DUALMIC	Speaker Phone
Dual Mic: Headset	ACM_HEADSET_SPEAKER_DUALMIC, ACM_HEADSET_MIC_DUALMIC	Ear Seal

Dual Mic: Bluetooth	ACM_BLUETOOTH_SPEAKER_DUALMIC , ACM_BLUETOOTH_MIC_DUALMIC	Ear Seal
Dual Mic: Bluetooth_NREC	ACM_BLUETOOTH_NREC_SPEAKER_DUALMIC , ACM_BLUETOOTH_NREC_MIC_DUALMIC	Ear Seal

7.2.2 audio_eq.nvm

This file includes the Equalizer settings that are fed into the Rx/Tx paths.

Table 10: ACM_EqParam Structure

ACM_EqParam Structure		
Member	Type/Range	Notes
params[31]	Signed Short	31 taps coefficients. The coefficient is between -1024 to 1024.
description[28]	Unsigned Char	Array of 28 character for structure description.

7.2.3 audio_ns.nvm

This file configures the noise suppressor.

Table 11: ACM_NSParam Structure

ACM_NSParam Structure		
Member	Type/Range	Notes
MaxSuppression	Unsigned Short	Maximal suppression in TX/RX noise suppression. Translation to dB is done via: $90.3 - 20 \log_{10}(x)$. For example, the value of 13 dB is reached with a value of 0x1ca8.
MaxNoiseLevel	Unsigned Short	Maximal noise level that is fully suppressed in TX/RX noise suppression. Translation to dBm0 is done via: $3x - 102$. For example, the value of -12 dBm0 is reached with a value of 0x001e.

tceThreshForVadEXP	Unsigned Short	Signal energy used for VAD detection. Translation to dBm0 is done via: $3x - 102$.
NoiseLevelInc	Unsigned Short	Noise update alpha factor for increment. Translation to dBm0 is done via: $x/32767$.
NoiseLevelDec	Unsigned Short	Noise update alpha factor for decrement. Translation to dBm0 is done via: $x/32767$.
cnCeil**	Unsigned Short	Max TX CNG amplitude. This limit max CN can be injected.
updateNoiseMatrix	Unsigned Short	A matrix for noise update. Larger value means more noise suppression ability, but may cause some speech distortion.
Refvadhang	Unsigned Short	Far-end vad hang

NRES Table (Only for Tx)

Member	Type/Range	Notes
VmThresDT(*)	Unsigned Short	Double Talk Detection thresh hold
Hang(*)	Unsigned Short	Far-end voice release time.
GammaDT(*)	Unsigned Short	Decide how much residual echo been subtract.
resInitPeriod	Unsigned Short	Algorithm initial training period time.
resOutputGain	Unsigned Short	Gain that amplifier final output of NRES.
TxnsCnGain(*)	Unsigned Short	Comfort noise injection gain
TxnsCnFloor	Unsigned Short	Minimum Comfort noise
cnHighFreqBoost	Unsigned Short	CNG High Frequency boost slop.
vmThresMus(*) (Affects SP mode only)	Unsigned Short	Affect Music Tone noise of spectral subtraction
musGainThres[2]Only when (vmThresMus > 0) (*)	Unsigned Short	Music tone noise rejection factor
resInitialW[16]	Unsigned Short	371,320,611,226,656,000,000,000,000,000,000,000,000,000,000,000
resAdaptEnable	Unsigned Short	Control resInitialW Update on/off Residual echo Attenuation method control
NonLinearModel	Unsigned Short	Residual echo remove method
BetaFest(*)	Unsigned Short	Only far-end speaking residual echo Suppression gain
BetaBrutal	Unsigned Short	Initial stage residual echo suppression gain
resBetaDT(*)	Unsigned Short	Double talk residual echo suppression gain
resMaxAdaptStep	Unsigned Short	Control resInitialW update speed of increment
resSlowAdaptWeight	Unsigned Short	Control resInitialW update speed of decrement
resInitAdaptDelay	Unsigned Short	Control resInitialW update speed

ResGainSmoothAttack	Unsigned Short	Residual echo suppression gain smooth factor
musGainFloor(*)	Unsigned Short	Minimum music tone noise rejection gain
refresFactor	Unsigned Short	Used for far-end VAD detection.
resBurst	Unsigned Short	Far-end voice detection time.

7.2.4 audio_ec.nvm

This file configures the EC and AVC parameters.

Table 12: ACM_ECPParam Structure

ACM_ECPParam Structure		
Member	Type/Range	Notes
sampleDelay	Unsigned Short	Compensate the echo path delay.
adaptSlowDown(*)	Unsigned Short	Controls the slow-down of LMS adaptation during DT
numOfTaps(*)	Unsigned Short	Number of LMS filter taps
qFir(*)	Unsigned Short	fix-point representation of the FIR

7.2.5 audio_ns.nvm

Table 13: ACM_NSParam Structure

ACM_NSParam Structure		
Member	Type/Range	Notes
MaxAGCGain	Unsigned Short	Upper limit on the AGC gain
MinEnvelope	Unsigned Short	Signals with envelope smaller than this threshold do not affect the AGC
Level1	Unsigned Short	Target level for the RX-signal envelope
MAX_RMS	Unsigned Short	The AGC gain is limited so that the signal RMS after amplification does not exceed MAX_RMS. Can be modify only in test commands
Init_Duration	Unsigned Short	The duration for smoothing the initial envelope (Units in voice frames).
PeakIncrease	Unsigned Short	Signal envelop increase tracing speed.

PeakDecrease	Unsigned Short	Signal envelop decrease tracing speed
IncrementRms	Unsigned Short	Signal RMS tracing speed

DRC Param

Member	Type/Range	Notes
DrcOperationMode	Unsigned Short	DRC operation mode
DrcRelease	Unsigned Short	RX signal envelop tracing decrement speed
DrcLookAhead	Unsigned Short	RX signal envelop tracing prediction time
MaxDRCGain	Unsigned Short	Upper limit on the DRC gain. Decreasing this parameter will decrease the effect of the DRC
DrcPeakLevelGain	Unsigned Short	Rx signal Peak level (AGC LEVEL1)gain
DrcNoiseLevelGain	Unsigned Short	Rx signal Noise level gain
DrcNoiseLevel	Unsigned Short	Rx signal noise level
DrcPeakRange	Unsigned Short	Drc Peak level range

7.2.6 audio_sidetone.nvm

This file configures the side tone parameters.

Table 14: ACM_SideToneParam Structure

ACM_SideTonParam Structure

Member	Type/Range	Notes
SidetoneGain	Unsigned Short	-36~12 dB

7.2.7 audio_HLPF.nvm

This file configures the HPF/LPF parameters.

Table 15: ACM_HLPFParam Structure

ACM_EqParam Structure		
Member	Type/Range	Notes
A_Coefficient[3]	Signed Short	Coefficients for LPF(Low Pass Filter)
B_Coefficient[3]		Coefficient for HPF(High Pass Filter)
description[28]	Unsigned Char	Array of 28 character for structure description.

7.3 audio_levante.nvm

This file contains the PMIC(Levante) hardware Codec slope and offset of each defined audio path. It use in 920s platform.

{{Gain_Id}, (Path_Id), (Slope), (Offset)}

Gain_Id - each PGA in the Codec is numbered in the NVM database with a unique Gain ID.

The digital gain in the MSA voice path is numbered with Gain_ID=0.

Table 16: Gain ID Table

Gain ID	Gain Name	Description	Comment
0	DIGITAL_GAIN	Digital gain of the MSA	
1	PGA_MIC1_GAIN	MIC1or MIC2 Analog Driver	
2	PGA_MIC2_GAIN	MIC3 Analog Driver	
3	PGA_AUX1_GAIN	AUX1 Analog Driver	
4	PGA_AUX2_GAIN	AUX2 Analog Driver	
5	HP1_A2A_GAIN		
6	HP2_A2A_GAIN		
7	ST_R_GAIN	Side Tone Right Gain	
8	ST_L_GAIN	Side Tone Left Gain	
9	CHANNEL_1_GAIN	Right HiFi Left Digital Gain	
10	CHANNEL_2_GAIN	LOFI Right Digital Gain	
11	CHANNEL_3_GAIN	Left HiFi Right Digital Gain	
12	CHANNEL_4_GAIN	Right HiFi Right Digital Gain	

Path_Id - each path in the Codec is numbered in the NVM database with a unique Path ID.
The following table lists all PMIC path IDs and names.

Table 17: Path IDs

Path ID	Path Name
PCM 1 output	
0	ELBA_AnaMIC1_TO_PCM_L
1	ELBA_AnaMIC1_LOUD_TO_PCM_L
2	ELBA_ExtAnaMIC_TO_PCM_L
3	ELBA_DigMIC1_TO_PCM_L
4	ELBA_DigMIC1_LOUD_TO_PCM_L
5	ELBA_DigMIC2_TO_PCM_L
6	ELBA_AnaMIC2_TO_PCM_L
7	ELBA_AUX1_TO_PCM_L
8	ELBA_AUX2_TO_PCM_L
9	ELBA_AnaMIC1_TO_PCM_R
10	ELBA_AnaMIC1_LOUD_TO_PCM_R
11	ELBA_ExtAnaMIC_TO_PCM_R
12	ELBA_DigMIC1_TO_PCM_R
13	ELBA_DigMIC1_LOUD_TO_PCM_R
14	ELBA_DigMIC2_TO_PCM_R
15	ELBA_AnaMIC2_TO_PCM_R
16	ELBA_AUX1_TO_PCM_R

17	ELBA_AUX2_TO_PCM_R
I2S 1 output	
18	ELBA_AnaMIC1_TO_I2S_L
19	ELBA_AnaMIC1_LOUD_TO_I2S_L
20	ELBA_ExtAnaMIC_TO_I2S_L
21	ELBA_DigMIC1_TO_I2S_L
22	ELBA_DigMIC1_LOUD_TO_I2S_L
23	ELBA_DigMIC2_TO_I2S_L
24	ELBA_AnaMIC2_TO_I2S_L
25	ELBA_AUX1_TO_I2S_L
26	ELBA_AUX2_TO_I2S_L
I2S Right output	
27	ELBA_AnaMIC1_TO_I2S_R
28	ELBA_AnaMIC1_LOUD_TO_I2S_R
29	ELBA_ExtAnaMIC_TO_I2S_R
30	ELBA_DigMIC1_TO_I2S_R
31	ELBA_DigMIC1_LOUD_TO_I2S_R
32	ELBA_DigMIC2_TO_I2S_R
33	ELBA_AnaMIC2_TO_I2S_R
34	ELBA_AUX1_TO_I2S_R
35	ELBA_AUX2_TO_I2S_R
From PCM Left	
36	ELBA_PCM_L_TO_EAR
37	ELBA_PCM_L_TO_SPKR
38	ELBA_PCM_L_TO_HS1
39	ELBA_PCM_L_TO_HS2
40	ELBA_PCM_L_TO_HS
41	ELBA_PCM_L_TO_LINEOUT1
42	ELBA_PCM_L_TO_LINEOUT2
43	ELBA_PCM_L_TO_LINEOUT
From PCM Right	

44	ELBA_PCM_TO_EAR
45	ELBA_PCM_TO_SPKR
46	ELBA_PCM_TO_HS1
47	ELBA_PCM_TO_HS2
48	ELBA_PCM_TO_HS
49	ELBA_PCM_TO_LINEOUT1
50	ELBA_PCM_TO_LINEOUT2
51	ELBA_PCM_TO_LINEOUT

From I2S

52	ELBA_I2S_TO_EAR
53	ELBA_I2S_TO_SPKR
54	ELBA_I2S_TO_HS1
55	ELBA_I2S_TO_HS2
56	ELBA_I2S_TO_HS
57	ELBA_I2S_TO_LINEOUT1
58	ELBA_I2S_TO_LINEOUT2
59	ELBA_I2S_TO_LINEOUT

Side Tone

60	ELBA_PCM_L_TO_OUT_L
61	ELBA_PCM_R_TO_OUT_R
62	ELBA_IN_L_TO_OUT_L
63	ELBA_IN_R_TO_OUT_R

Digital Loops

71	ELBA_PCM_INT_LOOP
72	ELBA_PCM_EXT_LOOP
73	ELBA_I2S_INT_LOOP
74	ELBA_I2S_EXT_LOOP
75	ELBA_DAC1_IN_EC_LOOP
76	ELBA_DAC2_IN_EC_LOOP
77	ELBA_DAC1_DAC2_IN_EC_LOOP
78	ELBA_PCM_L_IN_TO_I2S_L_OUT_LOOP

79	ELBA_PCM_L_IN_TO_I2S_R_OUT_LOOP
80	ELBA_PCM_L_IN_TO_I2S_Both_OUT_LOOP
81	ELBA_PCM_R_IN_TO_I2S_L_OUT_LOOP
82	ELBA_PCM_R_IN_TO_I2S_R_OUT_LOOP
83	ELBA_PCM_R_IN_TO_I2S_Both_OUT_LOOP
84	ELBA_I2S_IN_TO_I2S_L_OUT_LOOP
85	ELBA_I2S_IN_TO_I2S_R_OUT_LOOP
86	ELBA_I2S_IN_TO_I2S_Both_OUT_LOOP
87	ELBA_I2S1_IN_TO_I2S_L_OUT_LOOP
88	ELBA_I2S1_IN_TO_I2S_R_OUT_LOOP
89	ELBA_I2S1_IN_TO_I2S_Both_OUT_LOOP

Analog Loops

90	ELBA_AUX1_TO_HS2
91	ELBA_AUX1_TO_LOUT2
92	ELBA_AUX2_TO_HS1
93	ELBA_AUX2_TO_LOUT1

Loopback including codec and MSA

94	ELBA_PCM_L_TO_EAR_LOOP
95	ELBA_PCM_L_TO_HS_LOOP
96	ELBA_PCM_L_TO_SPKR_LOOP
97	ELBA_AnaMIC1_TO_PCM_L_LOOP
98	ELBA_ExtAnaMIC_TO_PCM_L_LOOP
99	ELBA_AnaMIC1_LOUD_TO_PCM_L_LOOP

Aux to speaker using side tone path

100	ELBA_AUX1_TO_SIDETONE_L
101	ELBA_AUX2_TO_SIDETONE_R
102	ELBA_SIDETONE_IN_TO_OUT
103	ELBA_SIDETONE_TO_SPKR

Force Speaker

104	ELBA_I2S_TO_SPKR_HS (Loud-Speaker with Headset)
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Record via GSSP

105	ELBA_AnaMIC1_TO_PCM_I2S_L
106	ELBA_AnaMIC1_TO_PCM_I2S_R
107	ELBA_AnaMIC1_LOUD_TO_PCM_I2S_L
108	ELBA_AnaMIC1_LOUD_TO_PCM_I2S_R
109	ELBA_ExtAnaMIC_TO_PCM_I2S_L
110	ELBA_ExtAnaMIC_TO_PCM_I2S_R
111	ELBA_AUX1_TO_PCM_I2S_L
112	ELBA_AUX2_TO_PCM_I2S_R

Slope/Offset - Slope and Offset are gain values which are generated during the calibration process. The linear combination of Slope and Offset generates the gain value in the audio path, which is required for volume settings.

Gain value result is generated in the analog path (within the PMIC) and in the digital path (in the MSA) according to the following equation:

$$\text{Gain [dB]} = \text{Volume settings [\%]} * \text{Slope}/1024 \text{ [dB]} + \text{Offset}/128 \text{ [dB]}$$

Slope and Offset parameters are calculated during the calibration process by solving linear equations (for example, define which gain corresponds to 50% volume and which gain corresponds to 100% volume).

7.4 audio_hifi_volume.nvm

It save audio hifi software gain volume calibration for 920s platform.

Table 18: hifiParam Structure

hifiParam Structure		
Member	Type/Range	Notes
paramType	Unsigned Short	0: playback 1: record
deviceType	Unsigned Short	output or input device
streamType	Unsigned Short	stream type for playback, application style for record.
formatType	Unsigned Short	file format for playback, encoding format for record.
volumeIndex	Unsigned Short	only valid for playback, it will be fixed to 0 for recording.
dBIndex	Unsigned Short	dB value index.

7.5 audio_gain_calibration.xml

The default “volume-gain” mapping information of all components is summarized in this XML file. Assume that for one gain, the gain mapping relationship on one path is fixed, but the gain mapping relationship on different paths may be different. ACM can position the gain mapping relationship according to a triple: <path identifier, component identifier, gain name>.

Notes: Only the gain for voice paths should be tuned. For HIFI paths, the gain mapping may be related with application scenarios, so the gain should be fixed, and use software amplifier instead.

The details of each tag will be described in the following sections.

Top Level Tag

Name	Description
Tag Name	ASRAudioGainConfiguration
Parent Tag	N/A
Possible Child Tags	<AudioPath>
Mandatory Attributes	N/A
Optional Attributes	N/A
Text	N/A
Description	top tag of the XML file

Audio Gain Configuration Tag

Name	Description
Tag Name	AudioPath
Parent Tag	<ASRAudioGainConfiguration>
Possible Child Tags	<Gain>
Mandatory Attributes	identifier: path identifier which the gains belong to componenet: component identifier which the gains belong to
Optional Attributes	N/A
Text	N/A
Description	describe the gain configuration of each component on each audio path

Name	Description
Tag Name	Gain
Parent Tag	<AudioPath>
Possible Child Tags	<Value>
Mandatory Attributes	name: gain name

Optional Attributes	N/A
Text	N/A
Description	describe the configuration of each gain

Name	Description
Tag Name	Value
Parent Tag	<Gain>
Possible Child Tags	N/A
Mandatory Attributes	volume: digital volume (percentage in decimal)
Optional Attributes	N/A
Text	gain register value related with each volume (hex)
Description	describe the gain configuration of each volume

Sample

```
<?xml version="1.0" ?>
<ASRAudioGainConfiguration>
  <AudioPath identifier="HiFiPlayToSPKR" component="ELBA">
    <Gain name="VOL_SEL_3[7:0]">
      <Value volume="10">0xb7</Value>
      <Value volume="20">0xa9</Value>
      <Value volume="30">0xab</Value>
      <Value volume="40">0xad</Value>
      <Value volume="50">0xaf</Value>
      <Value volume="60">0xb1</Value>
      <Value volume="70">0xb3</Value>
      <Value volume="80">0xb5</Value>
      <Value volume="90">0xb7</Value>
      <Value volume="100">0xb9</Value>
    </Gain>
  </AudioPath>
</ASRAudioGainConfiguration>
```

7.6 audio_swvol_volume.xml

This file will save the calibration result for software volume.

Top Level Tag

Name	Description
Tag Name	ASRSWVolCalibrationConfig
Parent Tag	N/A
Possible Child Tags	<AudioPath>
Mandatory Attributes	N/A
Optional Attributes	N/A

Text	N/A
Description	top tag of the XML file

Audio Gain Configuration Tag

Name	Description
Tag Name	PlaybackVolumeConfig
Parent Tag	< ASRSWVolCalibrationConfig>
Possible Child Tags	<Stream>
Mandatory Attributes	N/A
Optional Attributes	N/A
Text	N/A
Description	describe playback direction path

Name	Description
Tag Name	RecordVolumeConfig
Parent Tag	< ASRSWVolCalibrationConfig>
Possible Child Tags	<AppStyle>
Mandatory Attributes	N/A
Optional Attributes	N/A
Text	N/A
Description	describe record direction path

Name	Description
Tag Name	Stream
Parent Tag	< PlaybackVolumeConfig >
Possible Child Tags	<Device><CombinedDevices>
Mandatory Attributes	identifier: stream identifier
Optional Attributes	N/A
Text	N/A
Description	describe audio playback streams

7.7

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_voice_calibration.nvm

File audio_voice_calibration.nvm and audio_voice_calibration_ap.nvm are used for single NVM solution.

Member	Type/Range	Notes
profile_id	ACM_PROFILE_ID	The ID of profiles

description	char	Array of 32 characters for logging remarks.
Ssp_para	_ACM_SspAudioConfiguration	GSSP parameters
in_para	InPath_Structure	The parameters for audio input path .
out_para	OutPath_Structure	The parameters for audio output path.

_ACM_SspAudioConfiguration Structure

Member	Type/Range	Notes
shift	unsigned short	Shift of GSSP driver to get data.
SSCR0_HIGH	unsigned short	Value of SSP control register 0's high 16 bit
SSCR0_LOW	unsigned short	Value of SSP control register 0's low 16 bit
SSCR1_HIGH	unsigned short	Value of SSP control register 1's high 16 bit
SSCR1_LOW	unsigned short	Value of SSP control register 1's low 16 bit
SSTSA_LOW	unsigned short	Value of SSP TX time slot active register
SSRSA_LOW	unsigned short	Value of SSP RX time slot active register
SSPSP_HIGH	unsigned short	Value of SSP programmable serial protocol register's high 16 bit

SSPSP_LOW	unsigned short	Value of SSP programmable serial protocol register's low 16 bit
SSACD_LOW	unsigned short	Value of SSP audio clock divider register's low 16 bit
SSACDD_HIGH	unsigned short	Value of SSP audio clock dither divider register's high 16 bit
SSACDD_LOW	unsigned short	Value of SSP audio clock dither divider register's low 16 bit
devicePort	unsigned short	GSSP0_PORT=0, GSSP1_PORT=1
SSP_Rate	unsigned short	0=8 kHz, 1=16 kHz
SSP_Mode	unsigned short	0=unpacked, 1=packed
Dual_Mic	unsigned short	0:Single mic, 1: Dual Mic

InPath_Structure Structure

Member	Type/Range	Notes
HPF	ACM_Module_Switcher	Turn on/off HPF module in TX path
HPF_Para	_ACM_HLPFParam	HPF NB parameters.
HPF_Para_HD	_ACM_HLPFParam	HPF WB parameters.
LPF	ACM_Module_Switcher	Turn on/off LPF module in TX path
LPF_Para	_ACM_HLPFParam	LPF NB parameters.
LPF_Para_HD	_ACM_HLPFParam	LPF WB parameters.

EC	ACM_Module_Switcher	Turn on/off EC module in TX path
EC_Para	_ACM_ECPParam	EC NB parameters.
EC_Para_HD	_ACM_ECPParam	EC WB parameters.
NS	ACM_Module_Switcher	Turn on/off NS module in TX path
NS_Para	_ACM_NSParam	NS NB parameters.
NS_Para_HD	_ACM_NSParam	NS WB parameters.
EQ	ACM_Module_Switcher	Turn on/off EQ module in TX path
EQ_Para	_ACM_EqParam	EQ NB parameters.
EQ_Para_HD	_ACM_EqParam	EQ WB parameters.
AGC	ACM_Module_Switcher	Turn on/off AGC module in TX path
AGC_Para	_ACM_AVCPParam	AGC NB parameters.
AGC_Para_HD	_ACM_AVCPParam	AGC WB parameters.
CTM	ACM_Module_Switcher	Turn on/off CTM module in TX path
CTM_Para	_ACM_CTMPParam	CTM NB parameters.
CTM_Para_HD	_ACM_CTMPParam	CTM WB parameters.
DualMic	ACM_Module_Switcher	Turn on/off DualMic module in TX path
DualMic_Para	_ACM_DualMicParam	DualMic NB parameters.
DualMic_Para_HD	_ACM_DualMicParam	DualMic WB parameters.
MISC	ACM_Module_Switcher	Turn on/off MISC module in TX path

MISC_Para	_ACM_MISCParmTx	MISC NB parameters.
MISC_Para_HD	_ACM_MISCParmTx	MISC WB parameters.

OutPath_Structure Structure

Member	Type/Range	Notes
HPF	ACM_Module_Switcher	Turn on/off HPF module in RX path
HPF_Para	_ACM_HLPFParam	HPF NB parameters.
HPF_Para_HD	_ACM_HLPFParam	HPF WB parameters.
LPF	ACM_Module_Switcher	Turn on/off LPF module in RX path
LPF_Para	_ACM_HLPFParam	LPF NB parameters.
LPF_Para_HD	_ACM_HLPFParam	LPF WB parameters.
AVC	ACM_Module_Switcher	Turn on/off AVC module in RX path
AVC_Para	_ACM_AVCParm	AGC NB parameters.
AVC_Para_HD	_ACM_AVCParm	AGC WB parameters.
NS	ACM_Module_Switcher	Turn on/off NS module in RX path
NS_Para	_ACM_NSParam	NS NB parameters.
NS_Para_HD	_ACM_NSParam	NS WB parameters.
EQ	ACM_Module_Switcher	Turn on/off EQ module in RX path
EQ_Para	_ACM_EqParam	EQ NB parameters.
EQ_Para_HD	_ACM_EqParam	EQ WB parameters.
CTM	ACM_Module_Switcher	Turn on/off CTM module in RX path

_ACM_EqParam Structure

Member	Type/Range	Notes
params[31]	signed short	Up to 31 coefficients. The coefficient is between -1024 to +1024.

_ACM_NSParam Structure

Member	Type	Legal Range	Notes
MaxSupression	unsigned short	0 – 32767	Maximal suppression in TX/RX noise suppression. Translation to dB is done via: $90.3 - 20 \log_{10}(x)$. For example, the value of 13 dB is reached with a value of 0x1ca8.
MaxNoiseLevel	unsigned short	0 – 30	Maximal noise level that is fully suppressed in TX/RX noise suppression. Translation to dBm0 is done via: $3x - 102$. For example, the value of -12 dBm0 is reached with a value of 0x001e.
tceThreshForVadEXP	unsigned short	0 – 30	Signal energy used for VAD detection. Translation to dBm0 is done via: $3x - 102$.
NoiseLevelInc	unsigned short	327 – 32767	Noise update alpha factor for increment.

			Translation to dBm0 is done via: $x/32767$.
NoiseLevelDec	unsigned short	327 – 32767	Noise update alpha factor for decrement. Translation to dBm0 is done via: $x/32767$.
cnCeil	unsigned short	0 – 32767	Max TX CNG amplitude. This limit max CN can be injected. Translation to dB is done via: $90.3 - 20 \log_{10}(x)$.
updateNoiseMatrix	unsigned short	0 – 35	A matrix for noise update. Larger value means more noise suppression ability, but may cause some speech distortion.
Refvadhang	unsigned short	0 – 50	Far-end vad hang.
NRES_Parameters			
adaptEnable	unsigned short	0,1	Control resInitialW Update on/off
nonLinearModel	unsigned short	0,1	Residual echo remove method, 0 for ES and 1 for BT/SP

betaFest	unsigned short	0 – 32767	Only far-end speaking residual echo Suppression gain, ideally it's set to 3276 for ES/BT and 327 for SP. Translation to dB is done via: $20 \log_{10} (x/32768)$
betaBrutal	unsigned short	0 – 32767	Initial stage residual echo suppression gain. Ideally it's set to 32. Translation to dB is done via: $20 \log_{10} (x/32768)$
betaDT	unsigned short	7336	Double talk residual echo suppression gain. Ideally it's set to 7336.
gammaDT	unsigned short	512 – 4096	Decide how much residual echo been subtract. $\text{gammaDT} = x/1024$
vmThresDT	unsigned short	50 – 200	Double Talk Detection thresh hold
burst	unsigned short	0 – 10	Far-end voice detection time. Time = $x \cdot 10$ ms Ideally it's set to 0.

Hang	unsigned short	0 – 100	Far-end voice release time. Time = $x \cdot 10$ ms Ideally it's set to 20 for ES, 40 for BT, 100 for SP.
cnGain	unsigned short	0 – 32767	Comfort noise injection gain. Translation to dB is done via: $20 \log_{10} (x/1024)$.
cnFloor	unsigned short	0 – 32767	Minimum Comfort noise Translation to dB is done via: $20 \log_{10} (x/32767)$.
cnHighFreqBoost	unsigned short	0 – 1023	CNG High Freq boosts slop. Translation to dB is done via: $x \cdot 60 / 1024$
vmThresMus	unsigned short	<VmThresDT	Affect Music Tone noise of spectral subtraction, affects SP mode only
maxAdaptStep	unsigned short	8192	Control resInitialW update speed of increment
slowAdaptWeight	unsigned short	3 ES/BT 4 SP	Control resInitialW update speed of decrement
initPeriod	unsigned short	50 – 500	Algorithm initial training period time.

			Time = x*10 ms
initAdaptDelay	unsigned short	5	Control InitialW update speed. Time = x*10 ms
gainSmoothAttack	unsigned short	4200	Residual echo suppression gain smooth factor
outputGain	unsigned short	0 – 2	Gain that amplifier final output of NRES. Db = X*6dB
musGainThres[2]	unsigned short	0,1	Music tone noise rejection factor, Only when (vmThresMus >0). Factor = X/16384
musGainFloor	unsigned short	32	Minimum music tone noise rejection gain. Translation to dB is done via: $20 \log_{10}(x/32767)$.
refresFactor	unsigned short	0-3	Used for far-end VAD detection. Db = X*6dB
InitialW[22]	unsigned short	[11066,8000,4313,5125,5106,4353,6370,4481,3582,6436,6171,9036,2928.3640,5101,565,0,0,0,0,0] for ES [3713,2061,1226,655,999,1370,1792,2094,2087,2926,1845,1094,695,750,1442,1416,1212,4709,5217,5466,3788,16] for SP, [20040,11132,13029,12011,19114,16965,14460,9735,9521,10551,11340,17034,18689,15627,9800,3798,0,0,0,0,0] for BT	

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_ACM_ECParm Structure

Member	Type	Legal Range	Notes
EcMode	unsigned short		See ACM_MSA_ECMode.
EcSampleDelay	unsigned short	0 – 1600	Compensate the echo path delay. Range: 0-800 (for 8 kHz) and 0-1600 (for 16 kHz)
adaptSlowDown	unsigned short	0 – 31	Hangover duration for the RES in SP mode.
numOfTaps	unsigned short	64 – 128	Numbers of AEC filter taps.
qFir	unsigned short	14 – 18	Fix-point representation of the FIR.
Reserved[4]	unsigned short	N/A	Reserved parameters.

_ACM_AVCParm Structure

Member	Type	Legal Range	Notes
MaxAGCGain	unsigned short	0 – 32767	Upper limit on the AGC gain. Value translated to dB is reached via $3x/1024$. For example, the value of 21 dB is reached by using the value 7143.
MinEnvelope	signed short	-32768 – level1	Signals with envelope smaller than this threshold do not affect the AGC. Value translated to dB is reached via $3x/1024$.

			For example, the value of -23 dBov is reached by using the value -7824.
Level1	signed short	-4096 – 0	Target level for the envelope. Value translated to dB is reached via $3x/1024$. For example, the value of -1 dBov is reached by using the value -340.
BoostEnable	unsigned short	0,1	Enable boost function (for tx path only). Boost function Off = 0; Boost function On = 1.
MaxRMS	signed short	-32768 – 0	This parameter determines the amplification of non-speech signals such as holding tones. It should be tuned so that, for a given volume setting, typical holding-tones would sound with a similar loudness as the transmit speech.

			Value translated to dB is reached via $3x/1024$ For example, value=-5102, $Db = -5102 * 3/1024 = -15dB$
Init_Duration	unsigned short	0 – 5	The duration for smoothing the initial envelope (units in voice frames) in AGC path.
PeakIncrease	unsigned short	1 – 4096	Signal envelop increase tracing speed. Value translated to dB/frame is reached via $3x/1024$ For example, value=3410, $PeakIncrease = 3410 * 3/1024 = 10dB/frame$
PeakDecrease	unsigned short	1 – 68	Signal envelop decrease tracing speed Value translated to dB/Second is reached via $150x/1024$ For example, value=20, DSP will set $150 * 20/1024 = 3dB/Second$

IncrementRms	unsigned short	1 – 341	Signal RMS tracing speed Value translated to dB/Second is reached via $150x/1024$ For example, value=68, DSP will set $150 \times 68 / 1024 = 9 \text{ dB/Second}$
BoostGain	unsigned short	1364	Upper limit on the boost gain. [0db,6db] Db = $x/341$ (If Boost function is enabled, this parameter is available. It is also only for tx path.)
BoostRelease	unsigned short	65	Tracking speed factor for signal amplitude, and is typically of the order of 10-100 ms.(for tx path only) Translation Formula : $x / 32767$
BnoiseGain	unsigned short	-1024	Noise suppression ability for boost function.(for tx path only) [0db,-6db] Db = $x/341$
DRC_Parameters (for RX only)			

drcOperationMode	unsigned short	0,1,2	0: DRC Function off 1: DRC enabled without response to ambient noise 2: DRC enabled and response to ambient noise
drcRelease	unsigned short	0 – 655	The time scale over which the level is estimated is configurable by the parameter drcRelease, and is typically of the order of 10-100 ms. Translation Formula : $x / 32767$
drcLookAhead	unsigned short	1 – 4	A short delay (drcLookAhead) of typically 1ms is introduced in order to predict the signal level change direction. Translation Formula : $\text{Log}_2(\text{ms} \times 8)$
MaxDRCGain	unsigned short	0 – 7143	Upper limit on the DRC gain. Decreasing this parameter will decrease the effect of the DRC Value translated to dB is reached via $3x/1024$.

			For example, the value of 21 dB is reached by using the value 7143.
drcPeakLevelGain	signed short	-2048 – 0	The peak level is a property of the signal after the AGC: for speech signal it is equal to Level1, while for simple tones it is equal to maxRms+3dB. Translation Formula : $\text{dB}/3.01 \times 1024$
drcNoiseLevelGain	signed short	-4096 – 0	The noise level is a dynamic property of the signal, and is continuously tracked by the DRC. Translation Formula : $\text{dB}/3.01 \times 1024$
drcNoiseLevel	signed short	-30000 – -15000	Rx signal noise level Translation Formula : $\text{dB}/3.01 \times 1024$
DrcPeakRange	unsigned short	$(1 \sim 3) \times (\text{MaxDrcGain} - \text{PeakLevelGain})$	Drc Peak level range Translation Formula : $\text{dB}/3.01 \times 1024$
RxReserved0	unsigned short	N/A	
RxReserved1	unsigned short	N/A	

RxReserved2	unsigned short	N/A	
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_ACM_HLPFParam Structure

Member	Type/Range	Notes
A_Coefficient	ACM_FilterCoef	Coefficients for LPF(Low Pass Filter)
B_Coefficient	ACM_FilterCoef	Coefficient for HPF(High Pass Filter)

_ACM_CTMPParam Structure

Member	Type/Range	Notes
baudotRate	unsigned short	Baud rate
baudotInterface	unsigned short	Transmit interface
forceNegotiationIsOn	unsigned short	Whether force negotiation on
disableNegotiation	unsigned short	Whether disable negotiation
disableVcoOrHco	unsigned short	Whether disable Vco or Hco
volumeDifferenceIndex	signed short	The index for volume difference

_ACM_DualMicParam Structure

Member	Type/Range	Notes
DmConstant	signed short	Pickup insulation of Dual-Mic's voice default: 0x0005
DmVmThresh	signed short	Lower limit of the DM_VM curve. default: 0x0006
DmVmPeak	signed short	Upper limit of the DM_VM curve. default: 0x0080

DmAttnPeak	signed short	Parameter for restraining in noise area default: 0x0700
DmAttnMin	signed short	Parameter for restraining in voice area default: 0x0200
DmDecAlpha	signed short	The speed of VM smooth tracking(downward) default: 0x1000
AmbientNoiseLevel	signed short	Upper limit of NRES tracking noise in the noisy environment. default: 0x000e
StaticNoiseFactor	signed short	MicL estimation noise steady factor default: 0x7000
MicR_Gain	signed short	Linear gain complement for auxiliary microphone default: 16384
Bottom_Noise	signed short	Bottom noise energy,adjusted according to noise floor of amplifier in codec Range: -72dB ~ -68dB. default: -72

MinUnstableNoiseFactor	signed short	Minimum unstable noise factor. default:0x0080
MaxUnstableNoiseFactor	signed short	Maximum unstable noise factor. default:0x0100
Reserved3	signed short	
AmbientNoiseDiff	signed short	Ambient noise trace threshold. Db=x/341. Range:34~512 Default:171
OverSuppressionMatrix	signed short	Extra restraining parameter Default:0x0205
StaticNoiseFactor2	signed short	MicR estimation noise steady factor default: 0x2000
filter[0]---filter[15]	signed short	Coefficients for LMS filter.

ACM MISCParamTX Structure

Member	Type/Range	Notes
LoopbackDelay	short	Range: 0~50. For loopback test, AP will enable loopback device, then CP will send this delay value into DSP.

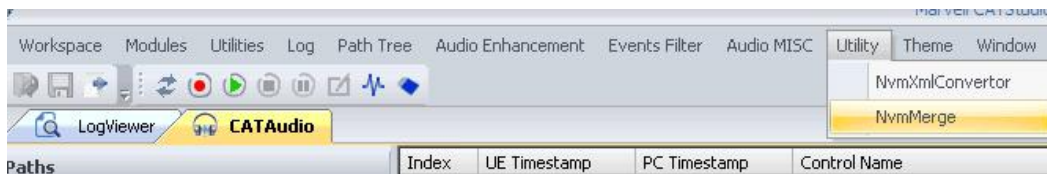
		The unit is 20ms, for example, LoopbackDelay=30 means DSP will add 20ms*30=600ms delay.
Parameters[31]	short	Reserved

_ACM_MISCParmRX Structure

Member	Type/Range	Notes
Parameters[32]	short	Reserved

NvmMerge Tool

You can open it by Open CAT-Audio, and then open Utility->NvmMerge.



8.1 Purpose and Scope

This document describes how to use NvmMerge to merge calibrated values in old nvm files into new nvm files.

8.2 How to use?

8.2.1 Rules

The steps are pretty self-explanatory. But first you should understand the rules it uses to merge the value from old nvm files to new nvm files.

The merge rules are as following.

- (1) All the merging process are doing based on audio_MSAmain.nvm, without this file, this software cannot work.
- (2) There should be two sets of nvm files. One we call "Source" here, which is the nvm files containing the values we want to migrate. The other we call "Target" here, which is the nvm files that the values would be written into.
- (3) The software will merge by scanning the MSAHandle in audio_MSAmain.nvm one by one, each MSAHandle contains a set of index which points to module nvm files.(such as EC, EQ, AVC etc.)

For each index in the same device MSAHandle:

1. If the source index is 0xffff and the target index is not 0xffff, then the target index will be changed to 0xffff.
2. If the source index is not 0xffff and the target index is 0xffff, then the target index will stay untouched.

3. If neither the source index nor the target index is 0xffff, the actual value in source module nvm file (audio_ec.nvm, audio_eq.nvm etc.) which the source index points to will be written into the target module nvm file the target index points to.

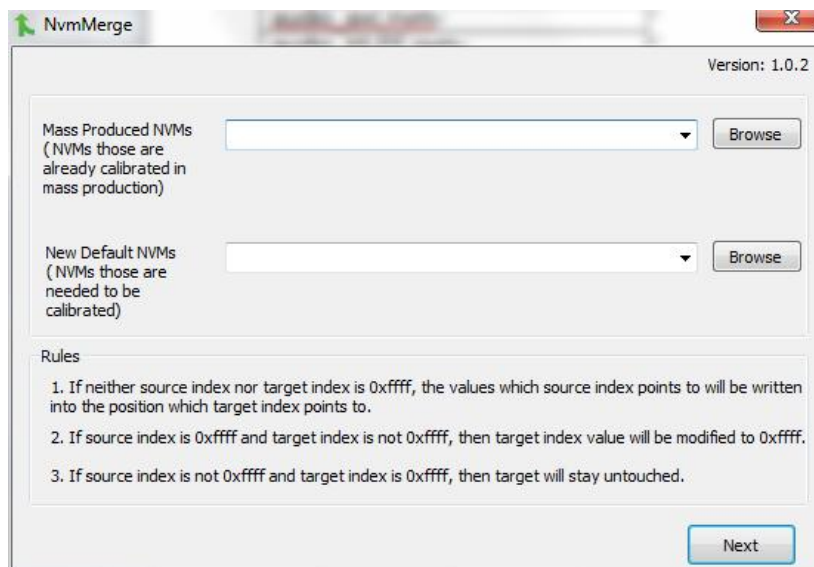
8.2.2 Support NVM files

So far, we support nvm files as following.

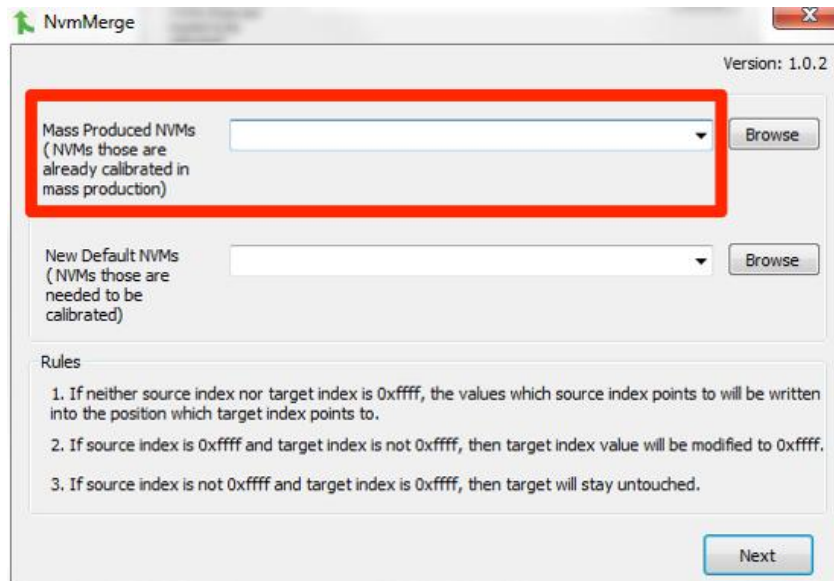
audio_MSAmain.nvm (must exist)
audio_eq.nvm
audio_ns.nvm
audio_avc.nvm
audio_HLPF.nvm
audio_Diamond.nvm
audio_Misc.nvm
audio_dualmic.nvm

8.2.3 Steps

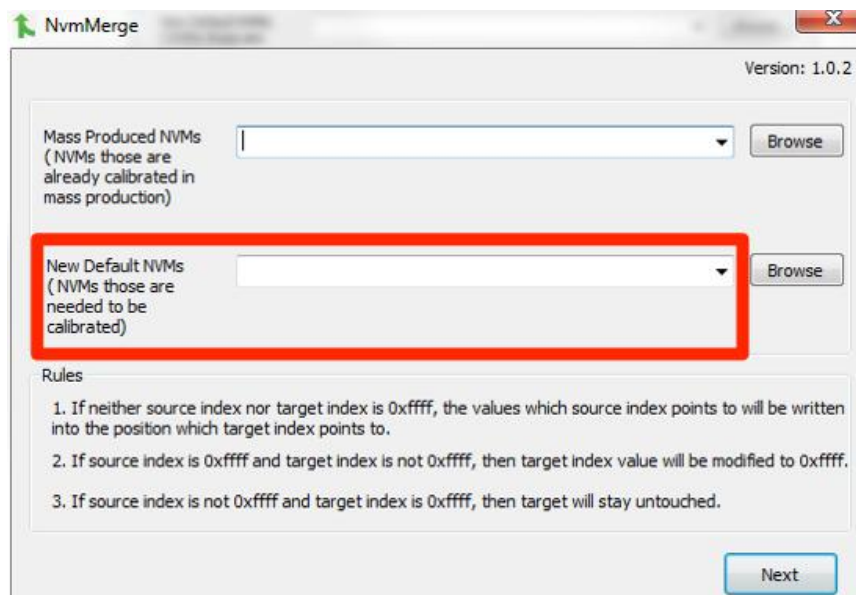
- 1) Open NvmMerge.exe



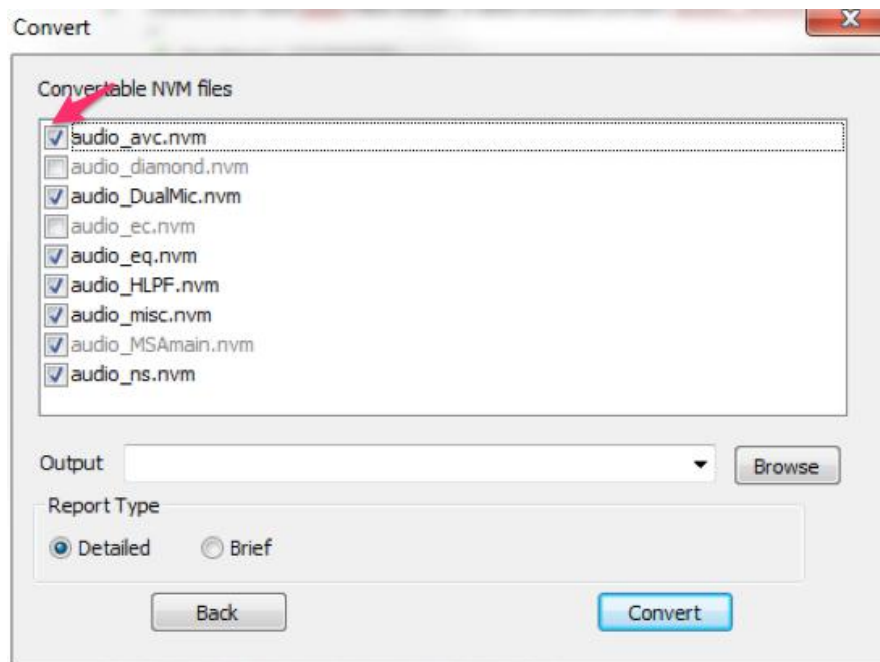
- 2) Select the "Mass Produced NVM" files folder, it should contain audio_MSAmain.nvm, audio_ec.nvm etc.



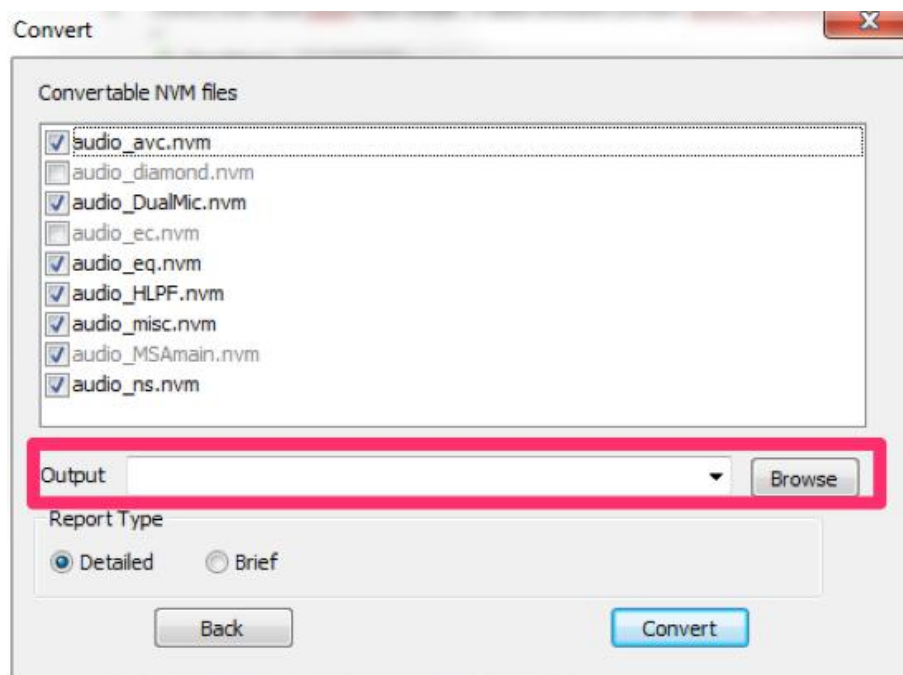
- 3) Select the “New Default NVM” files folder, it also should contain audio_MSAmain.nvm, audio_ec.nvm etc.



- 4) Select the NVM files you want to merge by checking it or not.
 Notice: audio_MSAmain.nvm must be selected. That's why you cannot unselect audio_MSAmain.nvm



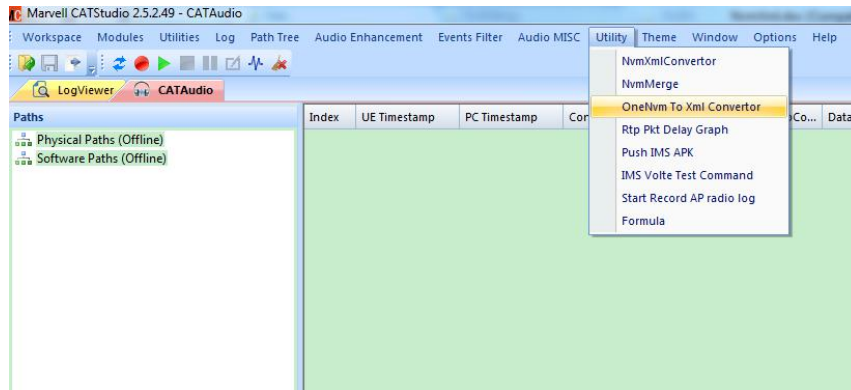
- 5) Select the output folder where you want to store the merged nvm files



- 6) Click "Convert" to do the merge, you will find the merged nvm files in the selected folder, and at the same time, a txt report with more converting details will pop up.

OneNvm to Xml Tool

You can open it by Open CAT-Audio, and then open Utility-> OneNVM to XML convertor.



9.1 Purpose and Scope

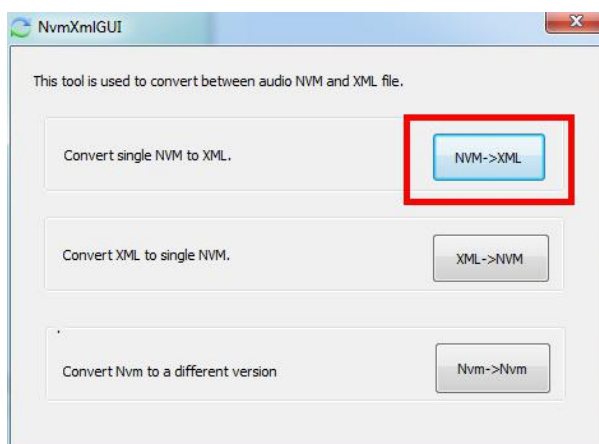
This document describes how to use OneNVM to XML convertor to convert single NVM file to XML and the other way.

9.2 How to use it?

Usually there are two scenarios. One is to convert NVM file to XML file, the other is to convert XML file back to NVM file. Let's illustrate them one by one.

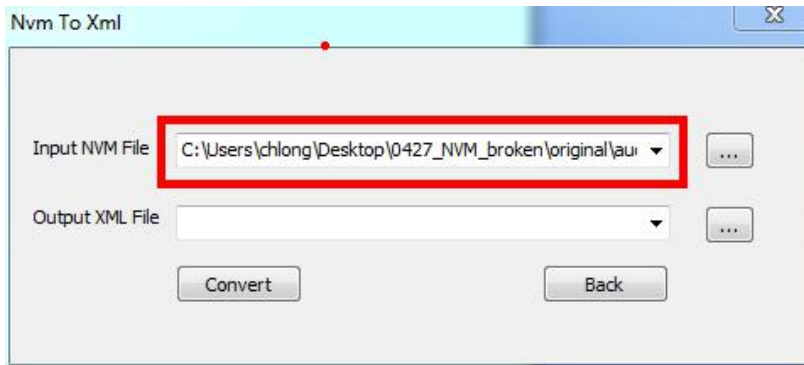
9.2.1 Nvm → Xml

1. First click "NVM->XML".

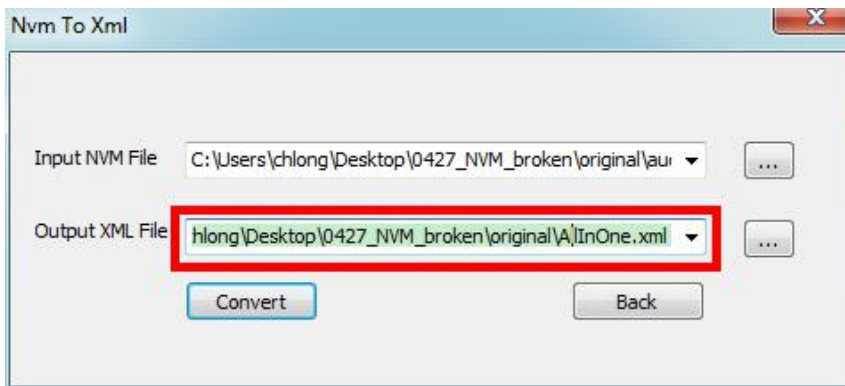


2. Select the input

audio_voice_calibration.nvm which you want to convert to XML.



3. Select an output XML file path, the name could be altered according to personal preference, default is "AllInOne.xml".

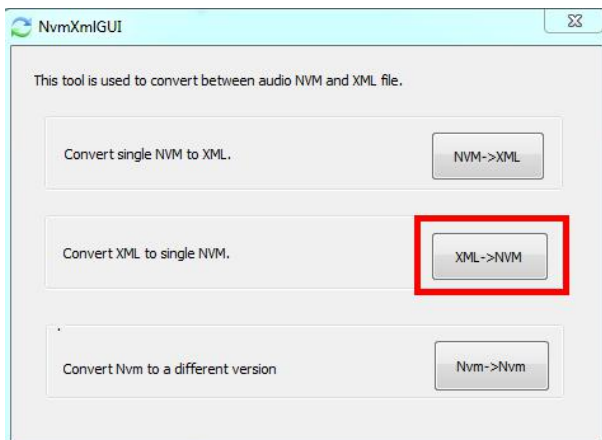


4. Click "Convert", and after a few seconds you will find the output XML file in the path you provide previously.

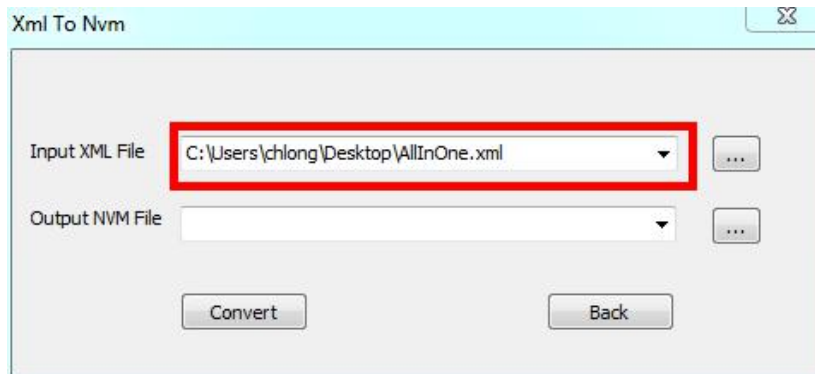
9.2.2 Xml → Nvm

The steps are quite similar to "NVM->XML".

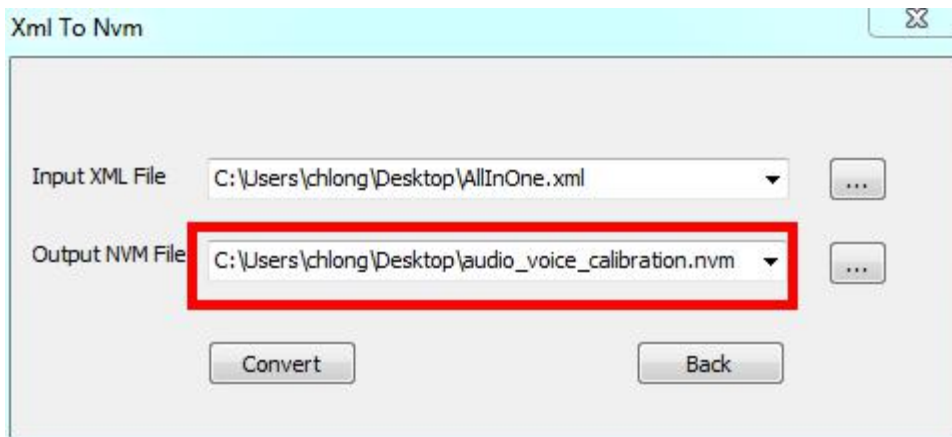
1. Click "XML->NVM" at the main GUI.



2. Select the input XML file which you want to convert to NVM.



3. Select an output NVM file path, the name could be altered according to personal preference, default is "audio_voice_calibration.nvm".

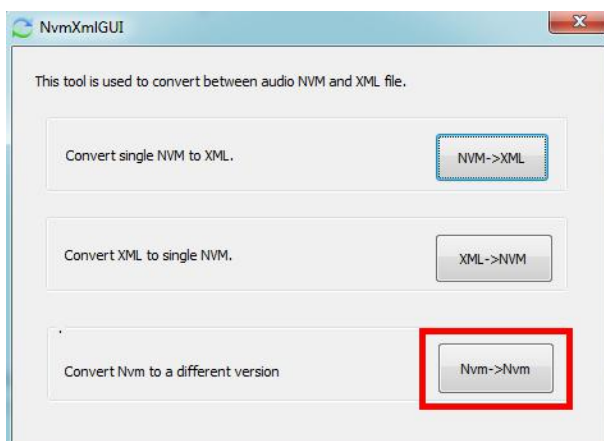


4. Click "Convert", and after a few seconds you will find the output NVM file in the path you provide previously.

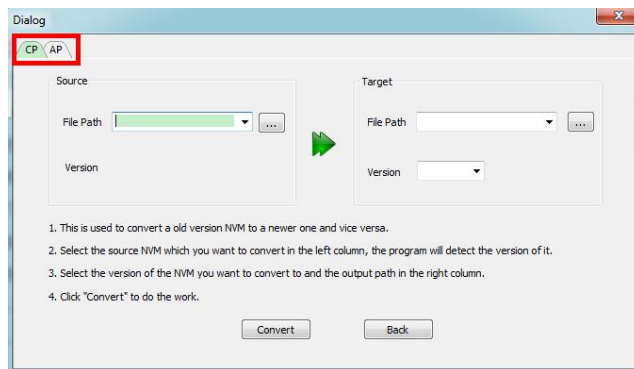
9.2.3 Nvm → Nvm

The steps are quite similar to "NVM->XML".

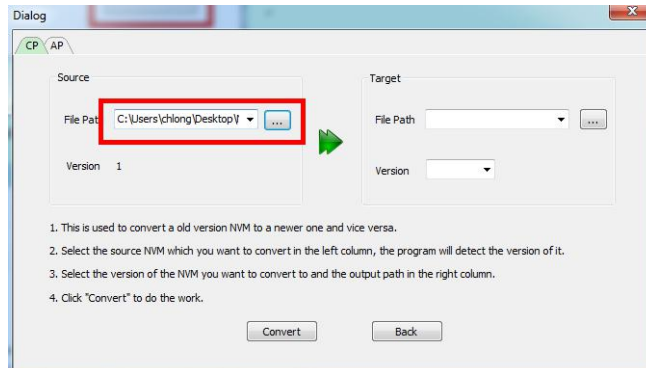
1. Click "XML->NVM" at the main GUI.



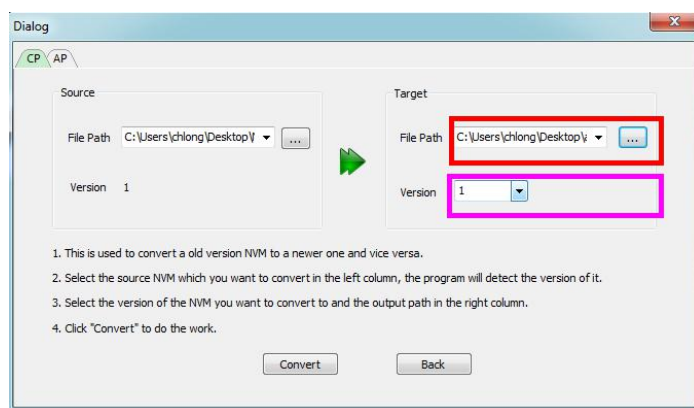
2. Select the tab page for current NVM file, if it is audio_voice_calibration.nvm, then you should choose CP tab, else if it is audio_voice_calibration_ap.nvm, then you should choose AP tab.



3. Select the input NVM file in the "File Path" combo box in the "Source" panel, the version of the NVM file will be detected automatically.



4. Select the output file path at the "File Path" combo box in the "Target" panel, and choose the version of the NVM you want to convert to in the "Version" combo box.



5. Click the button "Convert" to do the converting, after a few seconds you will find the output file at the path you choose.

The ASR Audio Processor (MAP) is a highly customized digital signal processor for voice and music applications. With one arithmetic logic unit (ALU) and one coprocessor, the MAP provides a simplified architecture with the following qualities:

-

10.2 Map Enhancement Tuning

10.2.1 MAP enhancement tuning conception

Configure MAP Audio enhancements' parameters by Audio Profile.

In current voice call scenarios, AP will send Audio Profile to CP side. And then CP uses this audio Profile to select related MSA enhancement configuration. And we use Audio Profile as MAP audio enhancement entry. We expand them for AP scenarios, for example, HIFI scenarios and VOIP scenarios.

10.2.2 MAP enhancement tuning

To access the CAT Audio Tool (online mode):

1. Open CAT-Studio and select online mode.

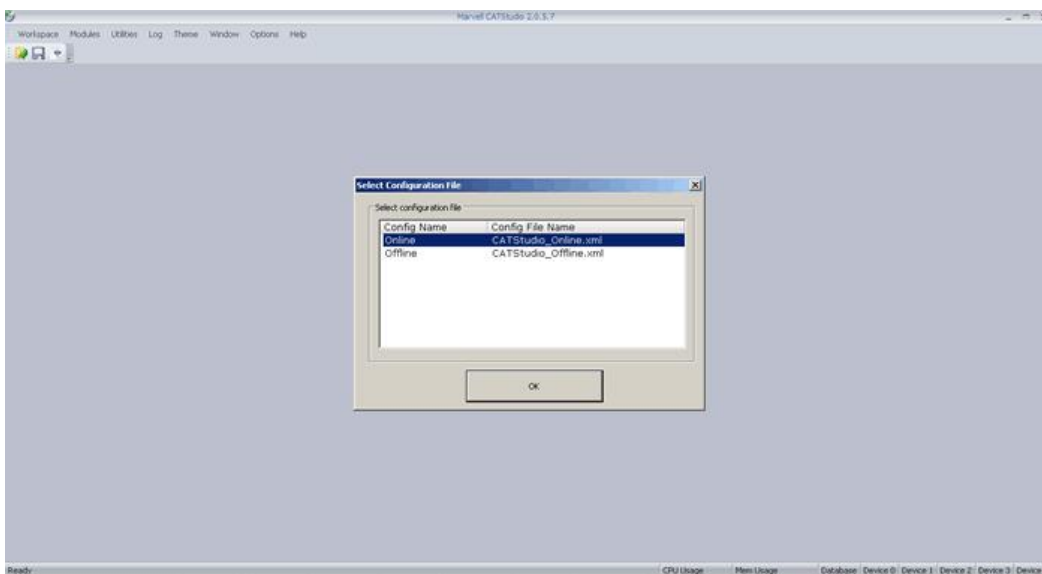


Figure: Open CAT-Studio online mode.

5. Connect CAT-Studio with board, and check whether both AP and CP Database are matched and can receive DIAG messages correctly.

Figure: Check CP and AP database.



6. Open the CAT-Audio from CAT-Studio menu as next figure.

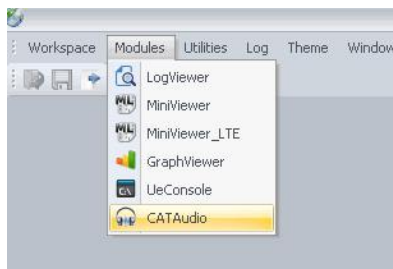
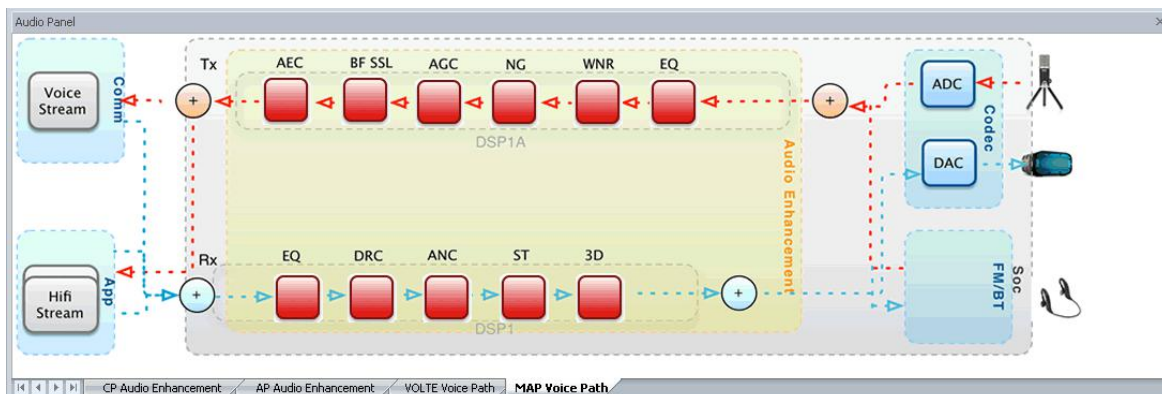
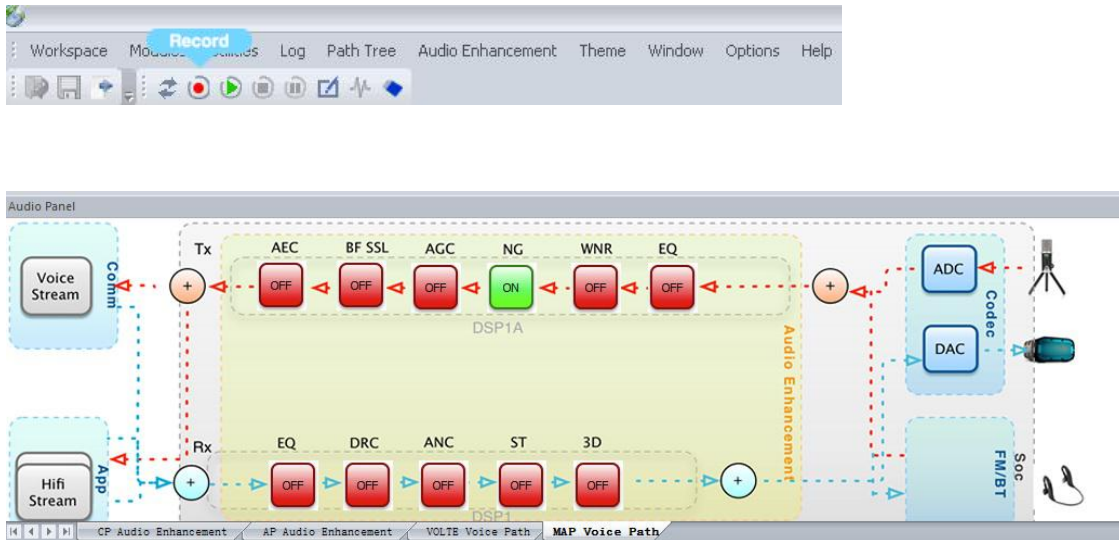


Figure: Open CAT-Audio

7. Select MAP Voice Path Tab.

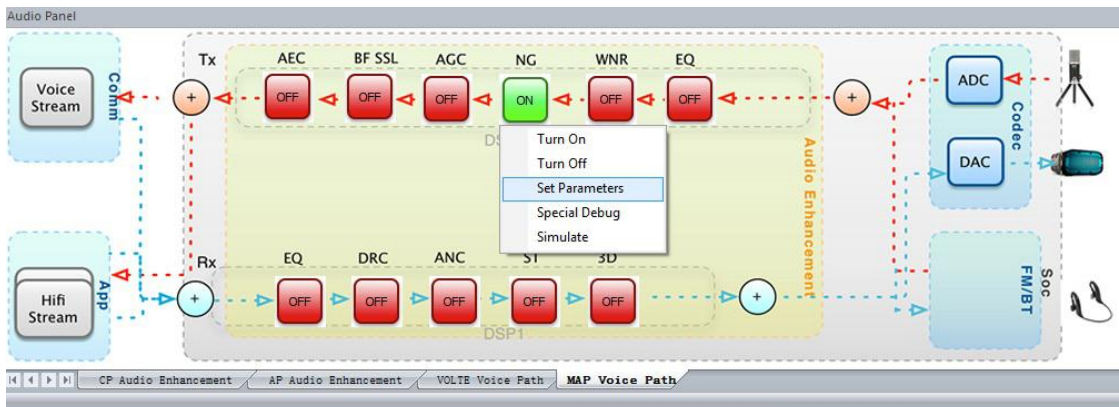


5 Click Record button and make a voice call.



Related module is enabled. It based on your default configuration and calibration data.

6 Right click on module. It will pop up a menu, select Set Parameters



7 Modify the parameters and click "Send" or "Send&Save". If custom clicks "Send&Save", the parameters will be saved into calibration data.

Revision History

Table 19: Revision History

Document No and Revision	Int Rev	Author	Description	Date
	0.1	Peter Shao	Initial version.	2012-09-03
	0.2	Peter Shao	Add chip lab chapter	2012-12-06
	0.3	Peter Shao	Update for CAT-AudioNT	2013-08-12
	0.4	Leon Cheng	Update for NvmMerge and NvmXml tool	2014-03-06
	0.5	Peter Shao	Update for MAP calibration	2014-08-05