Qualcomm Technologies International, Ltd.

BSWS-CVC-4-8-2 1M-HF

Tuning Guide

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Contents

Revision history . . . 7

1 Introduction . . . 8

1.1 General . . . 8

1.2 Software versions supported . . . 9

1.3 8th Generation new features . . . 9

1.4 Before you begin . . . 9

2 Prerequisites . . . 10

2.1 SPI communication protocol drivers . . . 10

2.2 Hardware interfaces . . . 11

2.3 Parameter Manager tool and user guide . . . 11

2.3.1 For Flash-based ICs . . . 11

3 Tuning overview . . . 14

3.1 Tuning stages . . . 15

3.2 Tuning flowchart . . . 16

4 Tuning preparation . . . 17

4.1 Before you start tuning . . . 17

4.2 Handsfree system in-vehicle . . . 17

4.3 Phones models and network types . . . 18

4.4 When using the in-vehicle audio system . . . 18

4.4.1 Vehicle HVAC settings . . . 19

4.4.2 Automotive factors . . . 19

4.4.3 Level speech phrase . . . 20

5 Instrumentation . . . 21

5.1 SPL meter . . . 21

5.2 Head and Torso Simulator . . . 21

6 Tuning preparation checklist . . . 22

7 Quick start guide . . . 23

7.1 Set Receive Gain path . . . 23

7.2 Set Send Gain path . . . 23
7.3 Echo/doubletalk performance . . . 23
8 Tuning procedures . . . 24
8.1 Objective measurement . . . 24
8.2 Receive Path tuning . . . 24
8.2.1 Load Preset Parameters . . . 24
8.2.2 Packet Loss Concealment . . . 25
8.2.3 Receive AGC . . . 25
8.2.4 SPKR Gain . . . 26
8.3 Send Path tuning . . . 29
8.3.1 MIC Gain . . . 29
8.3.2 Send AGC . . . 30
8.3.3 Send Noise Suppression (includes Wind Noise Reduction) . . . 32
8.3.4 Acoustic Echo Canceller . . . 34
8.3.5 Non-linear Processing . . . 36
8.3.6 Howling Control . . . 37
8.3.7 Comfort Noise . . . 38
9 Fine-tuning . . . 40
9.1 Receive Path fine tuning . . . 40
9.1.1 Setting minimum speaker gain loudness . . . 40
9.1.2 Receive AGC . . . 40
9.1.3 Receive EQ . . . 41
9.1.4 Clipper . . . 41
9.1.5 Auxiliary stream mix . . . 42
9.2 Send Path fine tuning . . . 43
9.2.1 Send EQ . . . 43
9.2.2 Send AGC . . . 43
9.3 Auxiliary stream mix 4-8-2 1M-HF . . . 44
10 Advanced tuning . . . 45
10.1 Advanced tuning features . . . 45
10.1.1 Noise-Dependent Volume Control . . . 45
10.1.2 Adaptive EQ (AEQ) . . . 46
10.1.3 AEQ with high frequency emphasis or expansion . . . 51
Document references . . . 54
Terms and definitions . . . 55

Tables

Table 1-1: Part number matrix.......... 9
Table 5-1: Recommended meter settings.......... 21
Table 8-1: Acoustic Echo Cancellation settings fields.......... 34

Figures

Figure 2-1: QTIL ADK software setup window with driver installation option selected..10
Figure 2-2: Accessing Headset Parameter Manager from UFE... 12
Figure 2-3: Parameter Manager window, Static mode (narrow band example).. 13
Figure 3-1: Processing blocks on the Parameter Manager window, Monitoring mode.. 14
Figure 3-2: Six cVc HF tuning stages.. 15
Figure 3-3: Tuning flowchart.. 16
Figure 4-1: Recorded waveform of the English phrase "one two three four five".. 20
Figure 8-1: Receive Path processing blocks..24
Figure 8-2: Packet Loss Concealment Settings window.. 25
Figure 8-3: Receive Automatic Gain Control Settings window, default...25
Figure 8-4: Adjusting the speaker gain..27
Figure 8-5: Headset configuration tool, audio gains.. 27
Figure 8-6: Receive Path Noise Suppression Settings window..28
Figure 8-7: Send Path Processing block diagram.. 29
Figure 8-8: Microphone Gain settings window...30
Figure 8-9: Send Path Noise Suppression Settings window... 32
Figure 8-10: Acoustic Echo Cancellation Settings window...34
Figure 8-11: Acoustic Echo Cancellation Settings fields...36
Figure 8-12: Non-linear Processing Settings window...37
Figure 8-13: Howling Control Settings window..38
Figure 8-14: Comfort Noise Settings window... 38
Figure 9-1: Hard Clipper Settings window..42
Figure 9-2: Auxiliary stream mix setting window..43
Figure 9-3: Auxiliary stream mix setting window..44

Figure 10-1: Noise Dependent Volume Control Settings window 46
Figure 10-2: Adaptive Equalization Settings window 48
Figure 10-3: Adaptive Equalization switching to mid noise tier 49
Figure 10-4: Adaptive Equalization switching to high noise tier 50

Revision history

Revision	Date	Description
1	December 2013	Original publication of this document. Alternative document number CS-004309819-UG.
2	December 2013	Editorial updates
3	May 2014	Editorial updates
4	May 2014	Updated graphics
5	July 214	Updated for ADK 3.5
6	May 2015	Updated for ADK 4.0
7	September 2016	Updated for ADK 4.1
8	April 2017	Updated for ADK 4.2
AJ	November 2017	Updated to reflect new document and revision numbering scheme.

1 Introduction

This document is designed for the Handsfree device manufacturers' audio developers and describes how to tune Clear Voice Capture (Qualcomm® cVc™) 1-mic Handsfree (HF) audio processing software running on Qualcomm® BlueCore-Multimedia Integrated Circuit (IC).

A Windows PC-based configuration tool (Parameter Manager) that communicates with Qualcomm’s BlueCore IC, simplifies the tuning process. This tool monitors audio signal statistics and can be used to adjust cVc HF audio processing block parameters to achieve optimal audio performance.

1.1 General

cVc software is a sophisticated application that enables users to compensate for environmental and acoustic variables to improve a product’s sound quality performance. The cVc software tuning procedure a series of acoustic and electro-acoustic tests and measurements performed in a specific sequence. The test results are used to modify cVc software audio processing parameters. The main purpose of the cVc software tuning procedure is to achieve an optimum level of Handsfree sound quality.

The product developer may perform the cVc tuning process at several stages during the Handsfree system’s development cycle. Typically, a developer fabricates a prototype Handsfree system that packages audio, power, communication, and processing components. Tuning can begin when a prototype system is available. Final tuning to verify optimal performance should be completed when the final production components and packaging are available.

The cVc HF application includes default settings that can be used for its audio processing. These defaults may require gains adjustment to compensate for variations in the hardware design, such as microphones and speakers.

1.2 Software versions supported

This Tuning Guide covers the audio adjustments of the cVc BCSW-CVC-HF-4-8-2 algorithm. The same audio tuning procedure is recommended when the algorithm is used on the ICs listed in Table 1.1.

Table 1-1 Part number matrix

IC Supported	cVc Product Code	Version SysID	NB (8k)	WB (16k)	cVc License Key Part Number
CSR8670 (Flash)	BCSW-CVC-HF-4-8-2	0xA012	Yes	Yes	BCSW-CVC-HF-4-8-2
CSR8675 (Flash)	BCSW-CVC-HF-4-8-2	0xA012	Yes	Yes	BCSW-CVC-HF-4-8-2

NOTE CSR8670/CSR8675 support narrow band (NB) using CVSD and includes wide band (16 kHz sample rate) using modified sub band coding (mSBC).

CSR86xx UFE installer is available at www.csrsupport.com for download.

1.3 8th Generation new features

cVc's 1-mic Handsfree solution is in constant development. Qualcomm Technologies International, Ltd. (QTIL) is adding features and making improvements driven by the market place.

This section lists improvements made since the previous release (BCSW-CVC-HF-4-8-1, A008) that improve performance and/or affect the tuning process.

New/improved features include:

- cVc Generation 8 feature support
- OMS re-factored processing to reduce MIPs
- Updated AGC Module that improves tracking of target level
- Updated CNG Module that enables selectable colored noise
- PEQ Master Gain independent form bi-quad stages
- Tone volumes have been normalized between processing modes with and without cVc
- Full Band Echo Cancellation added to AEC Module for greater echo reduction and better double talk performance in highly/close coupled speaker type applications.

1.4 Before you begin

QTIL recommends that product developers become familiar with the principles of acoustic performance and the tunable parameters supported by the Parameter Manager tool before starting to tune their devices. See Tuning flowchart for a flowchart of the tuning process.

2 Prerequisites

This section describes the prerequisites for cVc Handsfree tuning.

2.1 SPI communication protocol drivers

The cVc HF Parameter Manager tool requires the Serial Peripheral Interface (SPI) drivers. The appropriate drivers are included in the QTIL ADKs, if required. Install the ADK from www.csrsupport.com.

NOTE The SPI connection does not work if the SPI device drivers are missing.

To ensure these drivers are installed during the ADK installation, select the **Install the SPI device driver** option.

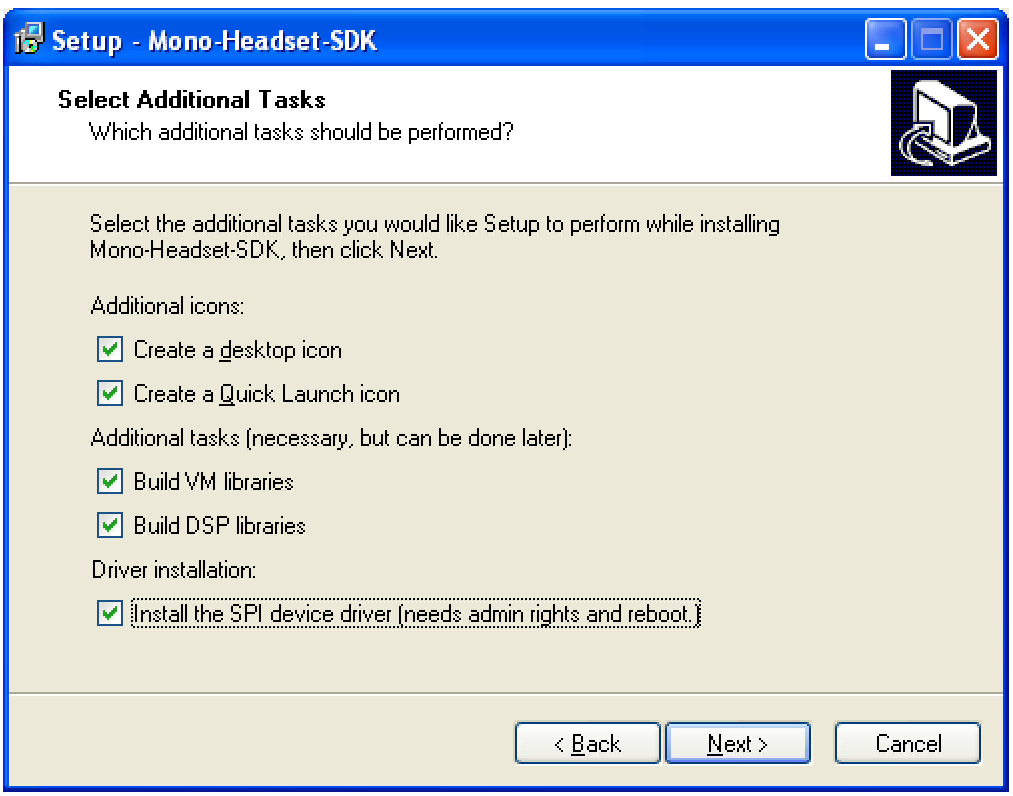


Figure 2-1 QTIL ADK software setup window with driver installation option selected

2.2 Hardware interfaces

The Parameter Manager and **PSTool** require a SPI connection to communicate with the target hardware. The Handsfree device under development must support a SPI interface to a PC in to tune the cVc software for a specific product.

2.3 Parameter Manager tool and user guide

Parameter Manager assists the tuning process in the following ways:

- Provides a graphical user interface (GUI)
- Displays live signal statistics
- Allows parameters to be easily adjusted using the Windows interface
- Makes changes to the tuning parameters storing them as PS Keys in the BlueCore Persistent Store.

The *BCSW-CVC-HF-4-8-2 1M-HF Parameter Manger User Guide* describes how to use the tool and explains the cVc parameters, their configuration, valid parameter ranges and their number formats.

The Parameter Manager is accessible through the Universal Front End (UFE) Application.

2.3.1 For Flash-based ICs

By default the ADK installation creates a subdirectory on the **root drive** of the PC:

C:\<ADK Name>\Tools\UFE\CSR™\UnviversalFrontEnd.exe

A corresponding Start Menu link is created during the installation process.

The Headset Parameter Manager is accessible from the HTML Start Page of Universal Parameter Manager application.

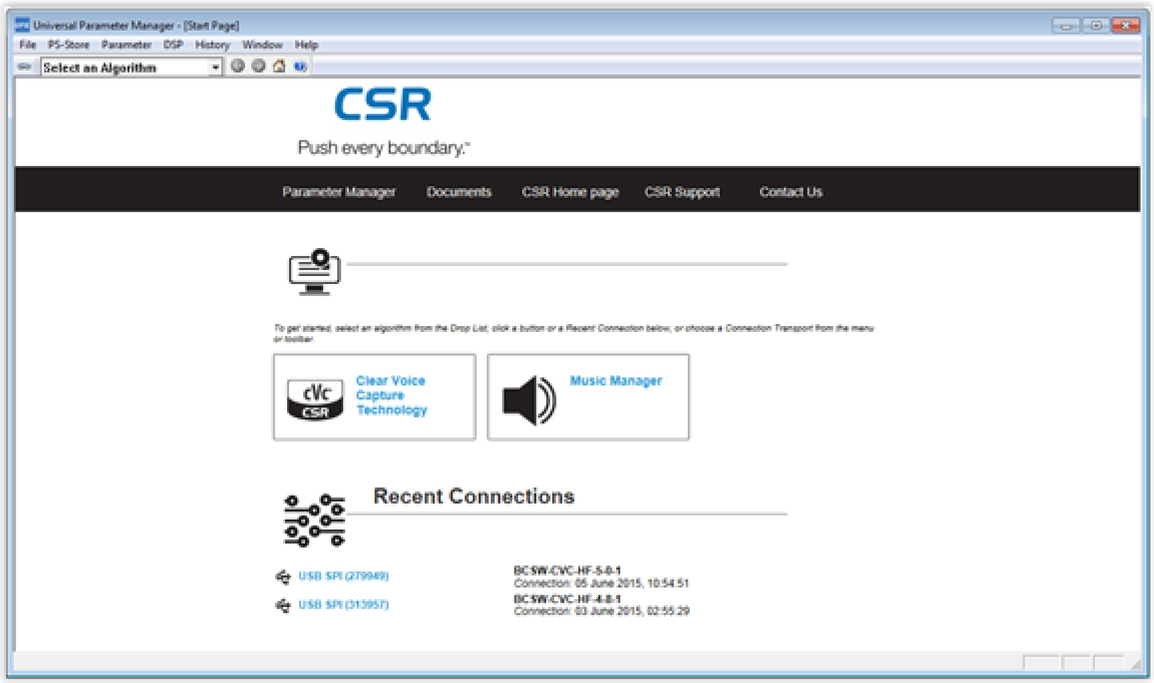


Figure 2-2 Accessing Headset Parameter Manager from UFE

Figure 2-3 shows the 1-mic CVC-HF-4-8-2 Handsfree Parameter Manager window that appears when the application is started i.e. the application is connected and in the Static mode. The application is in a connected mode for a NB (narrow band, 8000 Hz sample rate) system.

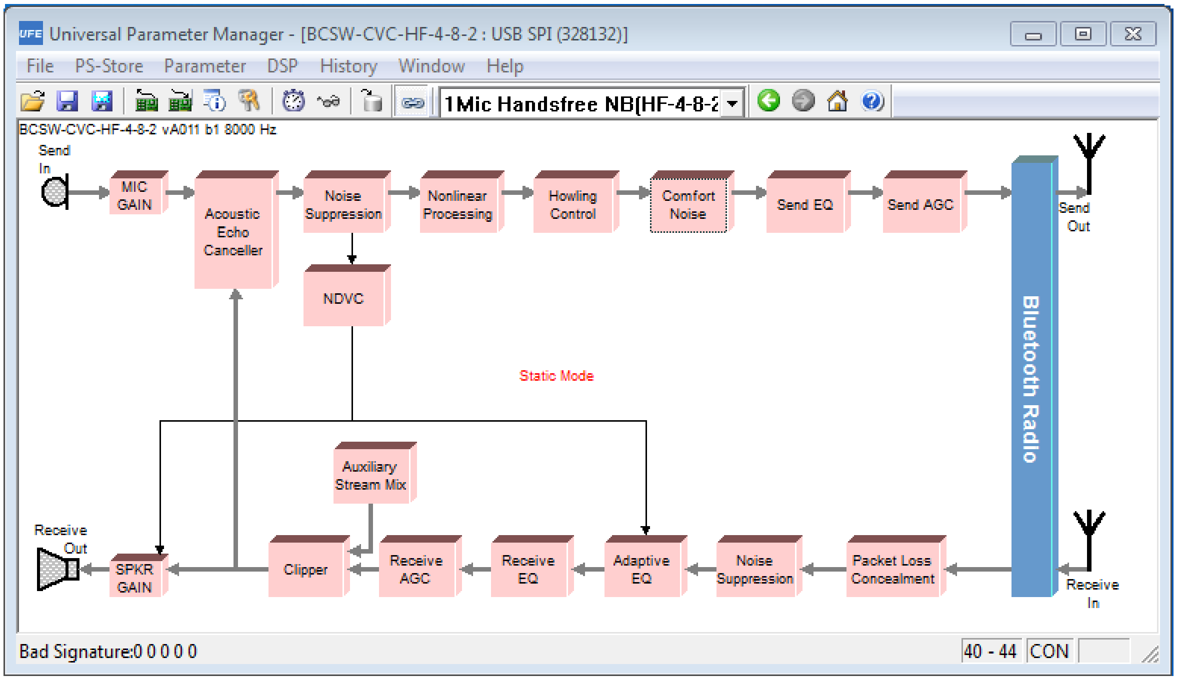


Figure 2-3 Parameter Manager window, Static mode (narrow band example)

3 Tuning overview

This section describes the overall process for tuning the cVc HF audio processing subsystem and presents the suggested sequence of the tuning process for cVc Handsfree applications.

Figure 3-1 shows the cVc processing blocks in the **Parameter Manager** window. The tuning process involves adjusting each of these major processing blocks and setting the gains at each cVc interface point.

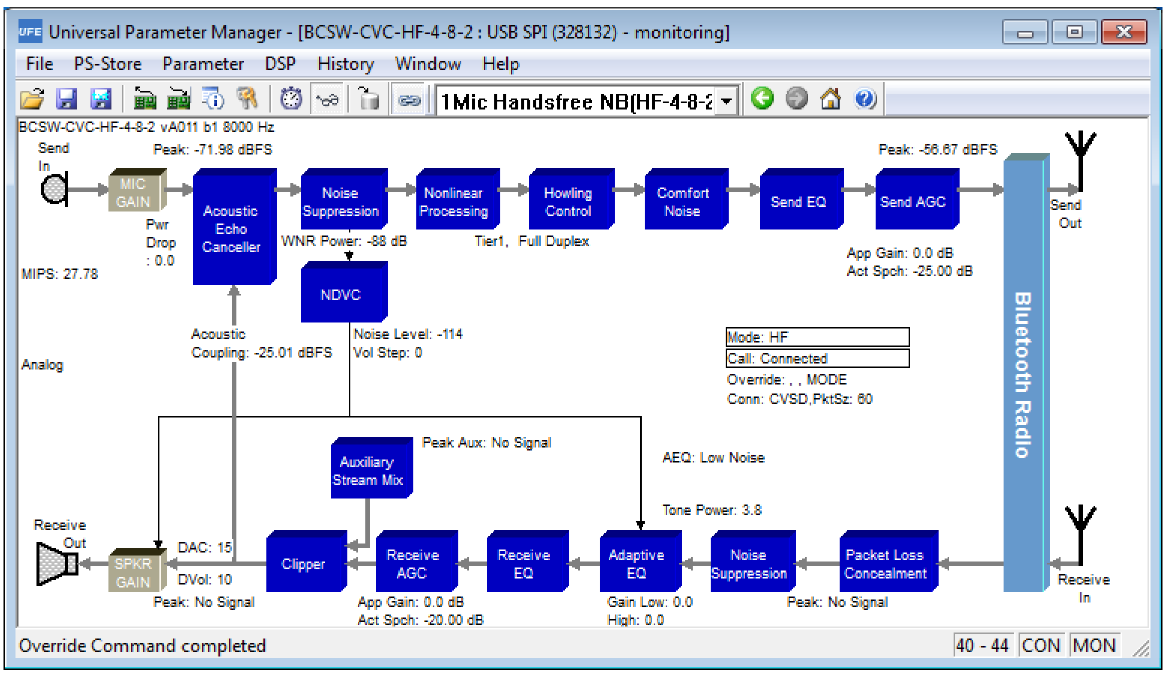


Figure 3-1 Processing blocks on the Parameter Manager window, Monitoring mode

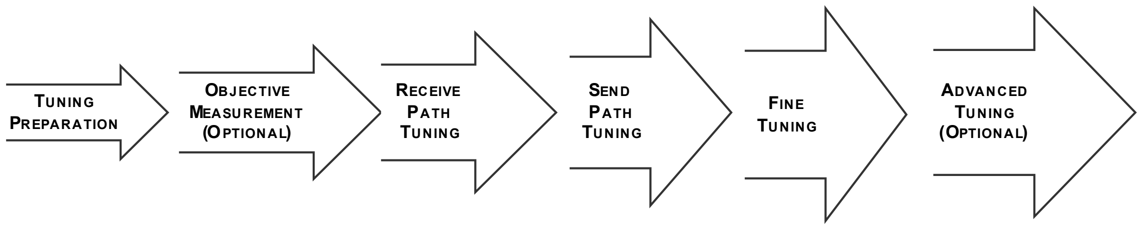


Figure 3-2 Six cVc HF tuning stages

NOTE Some stages may be unnecessary, if a specific cVc feature block is not used in a particular design.

3.1 Tuning stages

- **Tuning Preparation**: The process of preparing the test environment and setting up the test equipment.
- **Objective Measurement** (optional): Makes use of a HATS system to characterize frequency response, loudness rating, and distortion characteristics of the DUT.
- **Receive Path Tuning**: Focuses on tuning the Receive path processing blocks of the cVc HF algorithm (PLC, Receive AGC, Speaker Gain, and Noise Suppression).
- **Send Path Tuning**: Focuses on tuning the Send path processing blocks of the cVc HF algorithm (Microphone Gain, Send AGC, Noise Suppression, Acoustic Echo Canceller and Comfort Noise).
- **Fine Tuning**: The fifth stage at which minor adjustments are made to the processing blocks as required (Receive AGC, Receive EQ, Clipper, Auxiliary Stream Mix, Send EQ, Send AGC and Send Noise Suppression)
- **Advanced Tuning** (optional): The final stage is adding/tuning the advanced feature processing blocks.

NOTE Some of these features have been added as part of the 8th Generation cVc.

The following features may be enabled and tuned to enhance the audio performance. However, power consumption slightly increases as a result.

Enabling and tuning the following processing blocks should be made as required:

- Adaptive EQ with Frequency Expansion
- NDVC

NOTE Use the *BCSW-CVC-HF-4-8-2 1M-HF Parameter Manger User Guide* as a reference during the tuning process.

QTIL recommends that developers periodically save the best tuning settings to either the Persistent Store (PS) memory or to a `.psr` file to revert to for later use if required.

3.2 Tuning flowchart

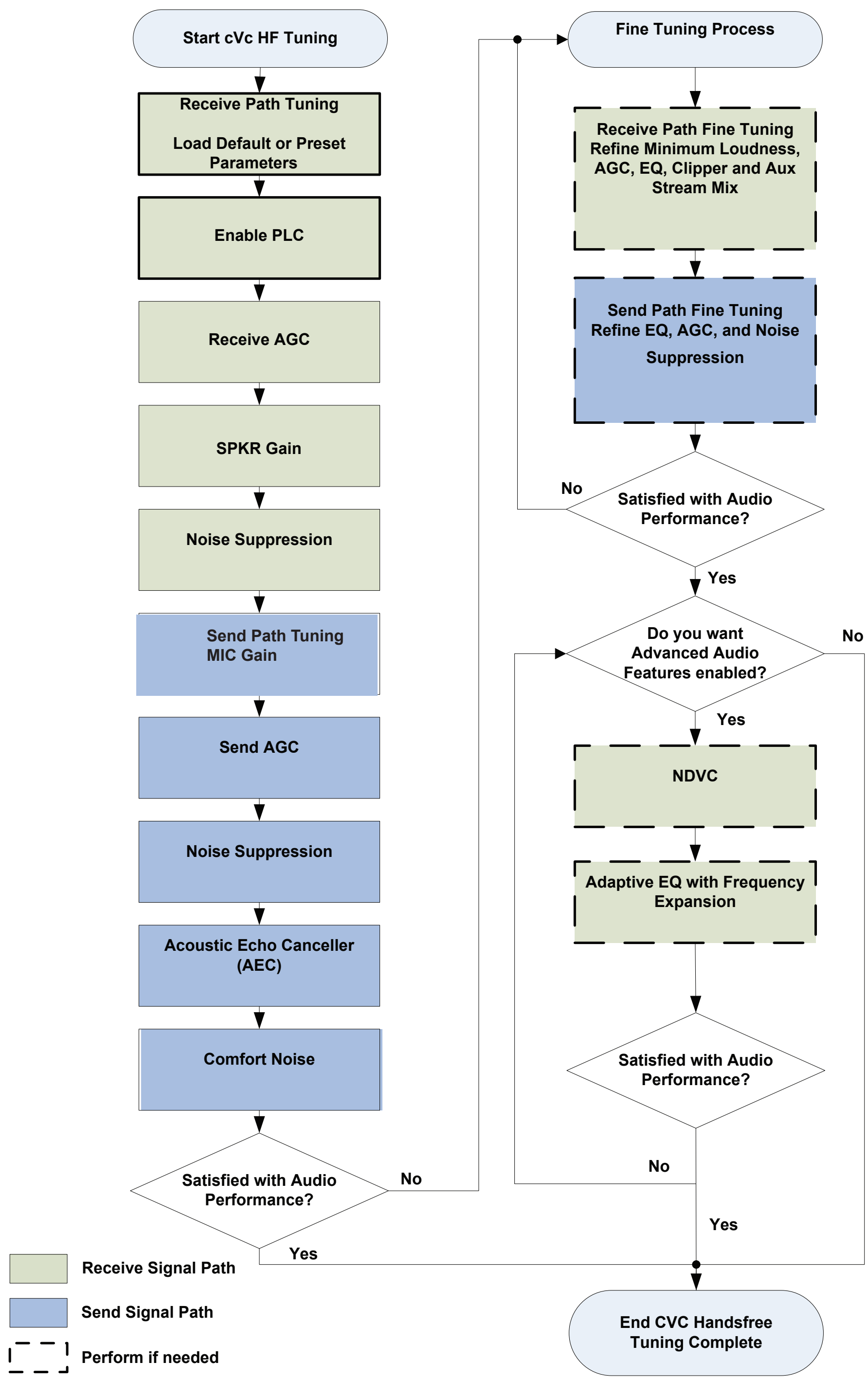


Figure 3-3 Tuning flowchart

4 Tuning preparation

This version of the cVc software is designed for use with Handsfree devices that have a one microphone input channel.

Because each cVc software application has a unique acoustical, electrical and mechanical environment in which it functions, you must consider the environment in which it will be used and the factors that may affect its performance.

4.1 Before you start tuning

Before starting the tuning process, review the preparation information in this section and complete the Tuning Preparation Checklist in section Tuning preparation checklist.

4.2 Handsfree system in-vehicle

Before tuning the Handsfree system check the following:

- The Handsfree audio components and plastics are production-level or production-intent.
- Any microphone wind noise reduction filters are in place.
- The device should include SPI connectivity to a PC running the Parameter Manager software tool.
- The SPI breakout cabling does not obstruct or interfere with the Handsfree microphone and receiver.
- Before testing, the battery is fully charged or has an external power source.
- Pair the Handsfree kit to a mobile telephone supporting the Bluetooth Handsfree profiles.
- During testing, the Handsfree kit is positioned in its intended location.
- Make a short call with the default cVc HF parameters to verify the performance before making any adjustments.

4.3 Phones models and network types

It is important to understand the general performance of the phones that the Handsfree product will support:

- Phone models and local networks vary and affect the sound quality, while also affecting the Handsfree product's performance.
- Phones or chipsets may have industry-known issues such as only supporting partial duplex.
- Local networks may have known noise suppression characteristics that cause fluctuations in the background noise during a call.

Because network types and phone models vary in the Receive SCO Audio level, QTIL recommends performing initial tuning using the mobile phone and network that has the lowest Receive SCO Audio level and has no known or discovered audio issues. To check the lowest Receive SCO Audio level, monitor the **Peak SCO in** statistic. When the initial tuning is complete and validated, test other phones to verify the performance (further fine adjustments may be required based on test results).

Perform tuning during cellular off-peak hours (10:00 to 16:00 and 19:00 to 07:00). This limits the amount of cellular network effects (such as aggressive routing schemes, comfort noise generation, bandwidth limiting, and compression) that may influence telephone sound quality.

QTIL recommends using a GSM mobile telephone as the primary tuning phone. Upon completing tuning, check the Handsfree sound quality with a CDMA mobile phone for similar results. In CDMA-dominant countries, tuning using only a CDMA phone is adequate.

While on a Handsfree call, it is normal to hear a buzz noise while tuning with a GSM mobile. You may need to move the GSM mobile to another location so that the buzz noise does not couple to the test hardware, especially the Handsfree microphone(s) and speakers.

NOTE Moving the Bluetooth-paired phone to different locations may degrade the Bluetooth link between the phone and the Handsfree kit. This affects the sound quality of the Handsfree call.

For best interoperability between the phone and headset product, QTIL recommends the Bluetooth Handsfree Profile (HFP) with an implementation of AT commands for turning off the mobile phone's noise reduction and echo cancellation processing.

4.4 When using the in-vehicle audio system

When using an audio/navigation head-unit, ensure that all in-vehicle audio system settings are nominal, such as:

- Customizable EQ is flat (Bass, Treble, Mid)
- Positioning is set to middle (Fade, Balance)
- DSP functions are off (such as concert hall effect, bass extension, and so on)

In automotive OEM solutions, tuning one Handsfree kit packaged with multiple types of audio systems may be required. For example, a vehicle may offer a Handsfree kit packaged with:

- An eight-speaker premium system
- A six-speaker mid-grade system
- A non-branded six-speaker base system.

In these cases, first tune the system that has the highest Sound Pressure Level (SPL) output at the Handsfree microphone position. When this tuning is complete, check the other audio systems for any problems in Handsfree sound quality.

NOTE To help minimize sound exposure, use earplugs when testing moderate to high sound pressure levels.

4.4.1 Vehicle HVAC settings

Follow the Heating Ventilation and Air Conditioning (HVAC) fan settings guidelines for each tuning procedure. If the HVAC setting is not listed, set the HVAC fan speed to either Low or Off. The wind direction should be set towards the face and foot. Test other conditions when the initial tuning is complete.

4.4.2 Automotive factors

Many automotive factors contribute to Handsfree sound quality. When testing, take into consideration the following conditions and any modifications required made to the tests:

- Noise, Vibration, and Harshness effects
- Wind Noise
- Vehicle Camouflage (pre-production cars)
- Engine location (Front/rear-mounted)
- Tire Noise
- Drive Train (AWD, FWD, RWD)
- Body Type (Sedan, Coupe, Convertible, SUV)
- Seat Type (Leather, Cloth)
- Prototype plastics
- Seat Position
- Exhaust Noise
- Road Condition (Wet, Dry, Bumpy)
- Road Surface (Asphalt/Concrete)
- Road Traffic
- Driving Speed
- Electromagnetic Effects, Wiring
- Microphone position
- Speaker position related to microphone position

4.4.3 Level speech phrase

To obtain a proper signal for measuring speech levels in the processor, QTIL recommends using a steady speech pattern. For example, repeat the English phrase “one two three four five” or “a b c d e” at a quick rate, with no pause between each word, for the specified measurement period. This technique proves a reasonably stable speech signal and reduces the dependence on sophisticated test equipment.

NOTE For more precise tuning, a laboratory-based test is required. QTIL can carry out laboratory-based tests on request.

When the near-end subject speaks the level speech phrase, measurement of the **Sound Pressure Level** should be approximately “90 dB SPL Average (C) Weighted Fast” measured 25 mm from the near end subject’s mouth.

Figure 4-1 shows a recorded waveform of the near-end subject’s voice while speaking the test phrase “one two three four five” twice.

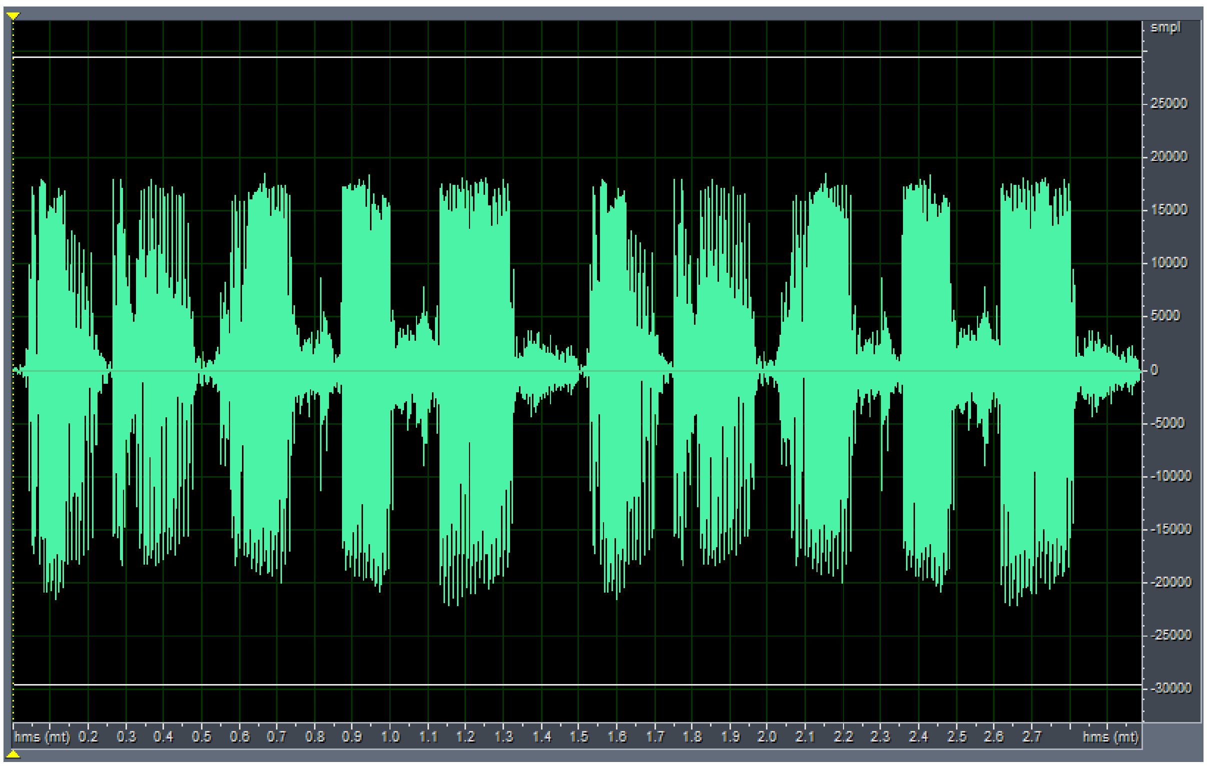


Figure 4-1 Recorded waveform of the English phrase “one two three four five”

5 Instrumentation

5.1 SPL meter

For measuring Sound Pressure Level (SPL), QTIL recommends a high quality digital sound meter, such as a Bruel & Kjaer (B&K) Type 2239 Sound Level Meter or equivalent.

Table 5-1 Recommended meter settings

Meter measurements	Meter settings
Measurement Type	Sound Pressure Level
Weighting	C-weighted (according to IEC-179)
Filter	Random incidence
Detector Type	RMS
Averaging	Fast
Units	dBSPL

5.2 Head and Torso Simulator

QTIL also recommends the using a Head and Torso Simulator (HATS) capable of electro-acoustic measurements on telephones per ITU-T recommendations. The HATS system should be capable of testing send/receive frequency response, send/receive loudness ratings and receive-path THD+N.

6 Tuning preparation checklist

Production or Production-intended Audio Components and Plastics	Check
SPI communication to PC	❑
■ Break-out SPI wiring does not obstruct or interfere with the Handsfree microphone and receiver	❑
■ Fully charged Handsfree battery or external power source	❑
■ Paired to a mobile phone	❑
GSM Mobile Phone with Bluetooth Headset Profile	
■ Fully charged phone battery or external power source	❑
■ Noise Suppression and Echo Cancellation disabled on the phone	❑
■ Mobile phone in close proximity to the headset	❑
■ No GSM buzz noise coupling on the headset	❑
Control of Noise Environment	
■ Near-end and Far-end subjects are acoustically separated	❑
■ Low network traffic test time	❑
■ Set the SPL Meter to the settings listed in SPL meter	❑
■ Calibrate the Sound Pressure Level Meter to a 1 kHz 94dB re 20µPa sine tone	❑
■ Ensure all cables and power supplies are in proper working order	❑
■ Parameter Manager tool connected to the Handsfree system	❑
Accessories	
■ Ear Plugs	❑
Personal	
■ Far-end (landside) subject	❑
■ Near-end (headset) subject	❑
Documentation	
■ Parameter Manager User Guide	❑

7 Quick start guide

This section describes how to get started with your cVc algorithm and provide a simple tuning method to set receive/send gain path and tune for acceptable noise/echo cancellation and doubletalk performance. You may need to perform additional tuning to achieve optimal performance, depending on the hardware and design of the product.

Before you begin, you must have an active SCO connection, with SPI communication, using the UFE in Monitoring Mode. To establish this, see Prerequisites.

7.1 Set Receive Gain path

Adjust **SPKR Gain** to the required maximum volume of the HFK and measure the echo signal at the microphone with a SPL meter or by measuring the Receive Loudness Rating (RLR). Maximum volume should not exceed 115 dB peak measured at the microphone to prevent clipping of microphone.

7.2 Set Send Gain path

Determine if the echo path signal (loudspeaker at maximum volume) is louder that the speech signal at the microphone.

If echo is louder than speech:

1. Adjust the Mic Gain so that the echo signal does not exceed -6 dBFS as seen in the SND IN statistics.
2. Monitor send speech level as seen in the SND IN statistics. Add Pre-Gain in the SND AGC so this level enters the SND AGC at approximately -18 dBFS.

If the speech is louder than echo, adjust the microphone gain so speech signal is approximately -18 dBFS as seen in the SND IN statistics.

See section 7.3.1 for details on tuning the Mic Gain, see MIC Gain.

7.3 Echo/doubletalk performance

If less echo is required, increase RER Power and/or decrease RER Adjustment.

If increased doubletalk is required, increase RER Adjustment and/or decrease RER power.

See Section 7.3.4 for details on tuning the Acoustic Echo Canceller.

> **NOTE** It could take several attempts to find the best ratio of RER Power/RER Adjustment to achieve optimal doubletalk and echo performance.

8 Tuning procedures

8.1 Objective measurement

After completing the tuning preparation process, the Handsfree device can be acoustically characterized using an objective telephone sound quality test system. This system normally uses HATS, a PC audio interface, professional-grade measurement microphones, reference phone system, and measurement software containing standardized test methods, such as ITU-T and TIA/EIA.

These objective telephone sound quality test systems are useful to perform future tuning activities:

- Send/Receive frequency response (TIA/EIA 810-A): Helpful for tuning Receive EQ and Send EQ.
- Send/Receive loudness rating (ITU-T P.79, ITU-T P.50): Helpful for tuning **SPKR Gain**, **MIC Gain**, and **Send Gain**.
- Send/Receive Distortion (THD/THD+N): Helpful for tuning the MIC Gain to a level limiting distortion and setting the **SPKR Gain** to a level limiting distortion .

8.2 Receive Path tuning

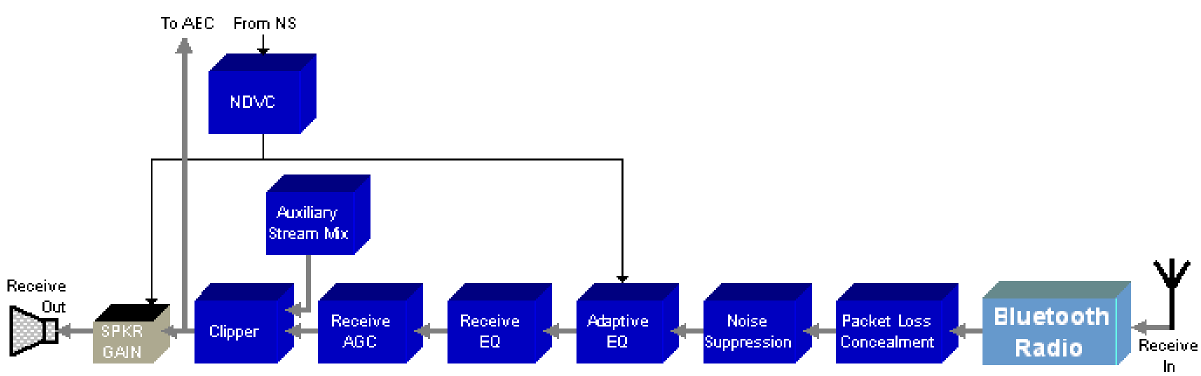


Figure 8-1 Receive Path processing blocks

NOTE Repeat all tuning measurements at least twice.

8.2.1 Load Preset Parameters

If the Handsfree device is being tuned for the first time, begin tuning with the default parameters provided with the cVc HF release. The defaults can be loaded using the Parameter Manager application by selecting **Use Default Parameters** on the **Parameters** menu. When the defaults have been loaded, bypass the advanced processing blocks (Bypass Adaptive EQ, Clipper and NDVC) to simplify tuning.

Alternately, if you have previously tuned the Handsfree device and have saved the parameters, you can preload the saved parameters and continue with the tuning process. The loading of saved parameters is described in the Parameters Manager integrated documentation. Click the **Quick Start** link on Parameter Manager Documentation opening window (home page) and read **Managing Parameter Settings and PS Key**.

8.2.2 Packet Loss Concealment

The Packet Loss Concealment block improves the receive path audio quality only in the presence of bit and packet errors within the Bluetooth link by using a variety of techniques such as pitch based waveform substitution. For the best audio quality, leave the Packet Loss Concealment enabled. To disable the module, select **Bypass Packet Loss Concealment**.

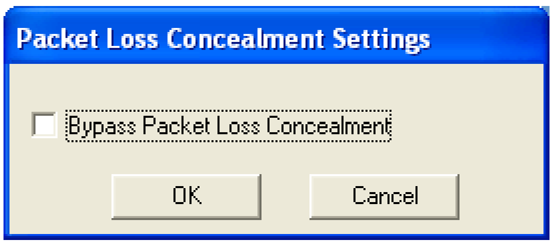


Figure 8-2 Packet Loss Concealment Settings window

8.2.3 Receive AGC

The Receive Automatic Gain Control (AGC) automatically adjusts the receive path **Receive SCO** signal to a specific level determined by the **AGC Target Level** parameter. This parameter compensates for variance in **Receive SCO** signal levels. By default, the **AGC Target Level** is -20 dB, and needs no additional tuning. If additional fine tuning is required, see Receive AGC.

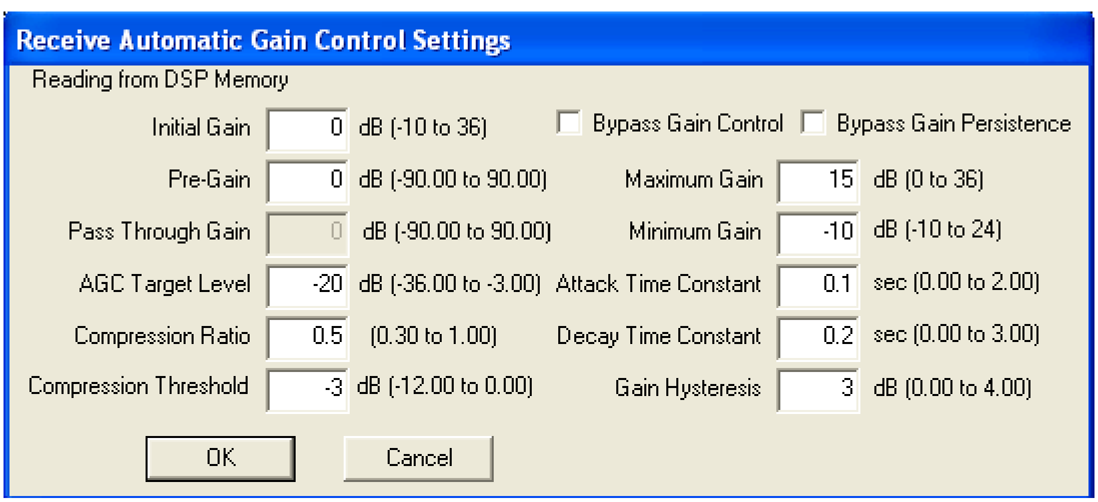


Figure 8-3 Receive Automatic Gain Control Settings window, default

NOTE Although it is possible to bypass the Receive AGC by selecting **Bypass Receive AGC**, QTIL does not recommend doing so.

8.2.4 SPKR Gain

Tuning the speaker involves determining the maximum receiver volume that the Handsfree device supports. As the Handsfree volume increases, the acoustic coupling between the receiver and the microphone increases. A large amount of acoustic coupling causes echo, which worsens when either the microphone or loudspeaker distorts.

Although you cannot eliminate distortion, you can minimize it to an acceptable level. Use better quality loudspeakers, amplifiers, microphones, leak-tolerant packaging, or improved acoustic separation of the microphone and receiver help to reduce echo and distortion.

The amount of distortion in the microphone and speaker determines the maximum volume. In general, limit the maximum receiver volume to the microphone's saturation level (approximately "110 dB SPL Average Fast Peek" at the Handsfree microphone position).

Before tuning, determine whether the Handsfree system controls the volume of the loudspeaker (via the Bluetooth volume control) or if an external amplifier (such as an audio/navigation head-unit) controls the volume of the loudspeaker.

NOTE In kits where the Handsfree system itself controls the volume of the loudspeaker, ensure that the Handsfree system is at maximum volume when tuning the receive path output level.

In Handsfree kits where an external amplifier (such as an audio head-unit (radio)) controls the volume of the loudspeaker, the Bluetooth phone should not affect the loudspeaker volume. For these product types, the Speaker Gain level remains fixed across all phone volumes.

To tune the **SPKR Gain**:

1. Initiate a Handsfree call.
2. The far-end subject speaks the level speech phrase.
3. The near-end subject measures or listens to the receiver volume.
4. The near-end subject adjusts the loudspeaker volume by clicking the **SPKR GAIN** block, selecting **Temporarily override DAC** and adjusting the **SPKR Gain** to the highest level that does not cause distortion and passes objective and subjective loudness judgment.

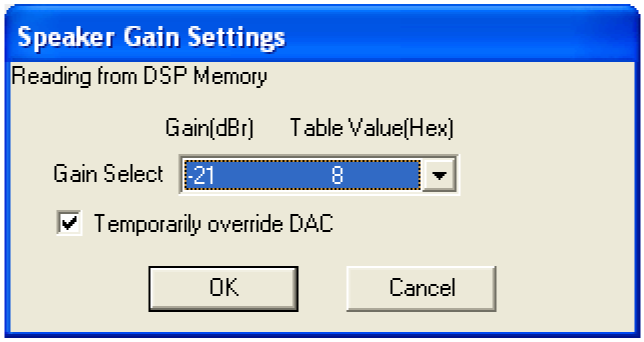


Figure 8-4 Adjusting the speaker gain

a. The near-end subject lowers the **SPKR Gain** or enables and adjusts the Clipper:

b. If the clipper is tried, follow the Clipper setup described in Clipper

c. Optional: The near-end subject adjusts the Boost so that the required receiver loudness is maintained.

5. In the Headset Configuration Tool, select the **Audio Gains** tab to configure the required number of volume steps and set the maximum volume.

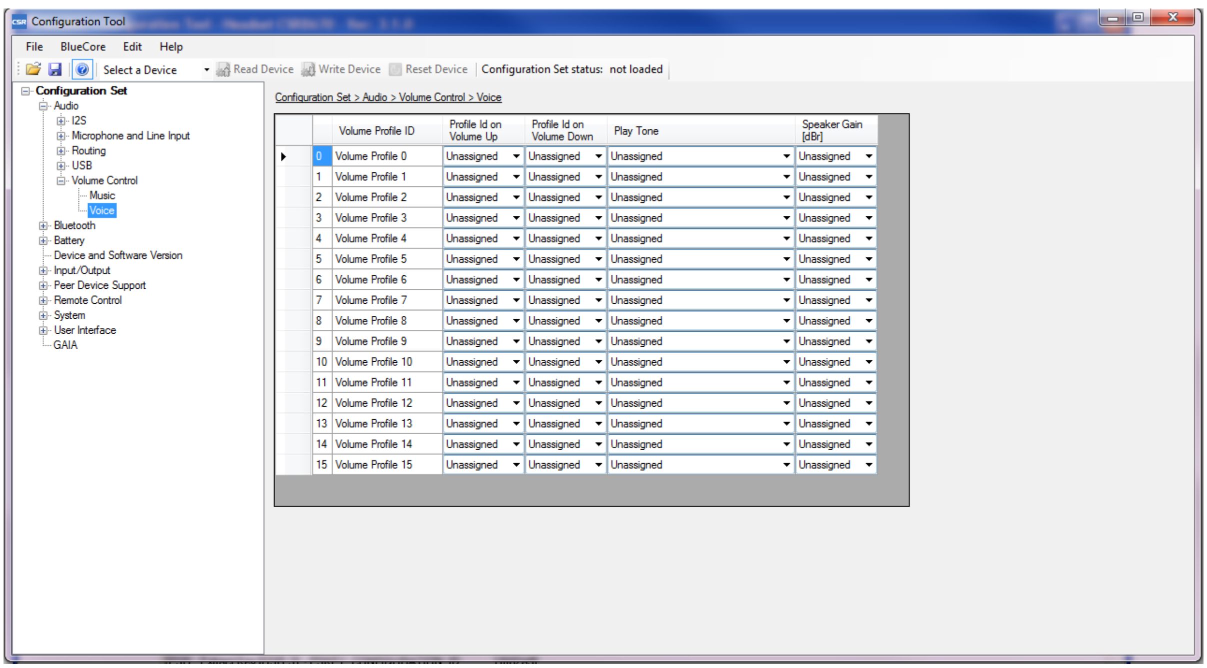


Figure 8-5 Headset configuration tool, audio gains

To tune the **SPKR Gain** with Audio/Navigation Head-Unit Volume Control:

1. Initiate a Handsfree call.
2. The far-end subject speaks the level speech phrase.
3. The near-end subject adjusts loudspeaker volume by clicking on the **SPKR Gain** block, selecting **Temporarily Override DAC**, and adjusting the **SPKR Gain** to the highest level that does not overload the audio input to the audio/navigation head-unit and cause distortion.

4. The far-end subject speaks the level speech phrase.
5. The near-end subject adjusts the audio/navigation head-unit's volume level to a level that does not cause distortion and passes subjective loudness judgment, no larger than "110 dB SPL Average Fast Peek" at the Handsfree microphone.

Use the Headset Configuration Tool shown in Figure 8-5. Select the **Audio Gains** tab to configure the **SPKR Gain [dB]** values in the VM volume table to be the highest level established in step 3. The **SPKR Gain [dB]** configuration should be the same for all **HFP Levels**. Receive Noise Suppression

The **Aggressiveness** parameter is a primary tuning parameter which controls the amount of noise suppression that is applied to the receive signal. Setting this parameter to 80% suppresses ~6 dB of noise (recommended) and 100% suppresses up to 20 dB of noise. However, the receive signal has been processed by the cellular network and transmitted over Bluetooth. To avoid overprocessing the voice, set the aggressiveness conservatively.

To tune for Noise Suppression:

1. Initiate a Handsfree call.
2. The far-end subject introduces background noise into the testing environment.
3. The far-end subject speaks a test phrase or a normal conversational phrase, continually.
4. Under different noise conditions, the near-end subjectively evaluates the noise level and its quality with and without far-end speech.
5. Increase the **Aggressiveness** for more noise suppression (at the cost of voice quality). Decrease the **Aggressiveness** for less noise suppression.

NOTE The far-end subject should avoid using the level speech phrase while tuning the Receive Noise Suppression Aggressiveness.

Normal conversational speech or phonetically-balanced phrases and passages are better for judging speech intelligibility in the presence of noise.

High Quality Mode is selected (enabled) by default. This does not affect the noise suppression but provides improved speech quality.
If voice quality is not critical this option can be un-selected to reduce the processing load. However, because high quality mode only consumes about 1 MIPS, leave this enabled.

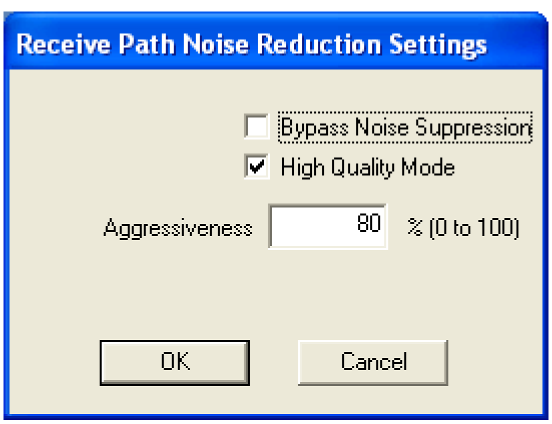


Figure 8-6 Receive Path Noise Suppression Settings window

8.3 Send Path tuning

The send path processes speech, echo and noise entering the Handsfree microphone. The echo signal is the result of acoustic coupling from the loudspeaker to the **Send In** microphone.

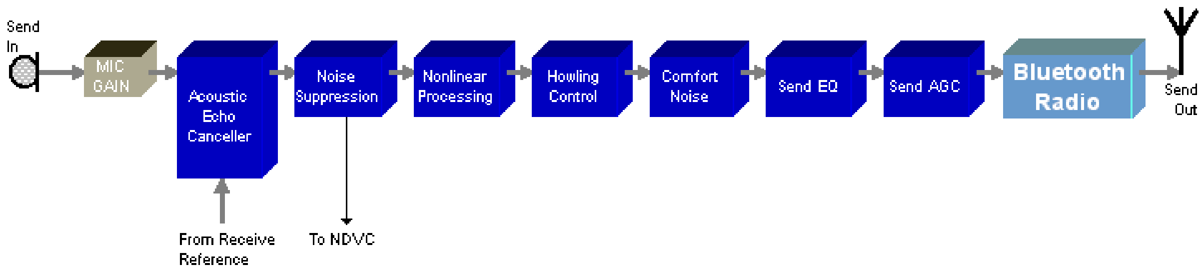


Figure 8-7 Send Path Processing block diagram

8.3.1 MIC Gain

The MIC Gain is an analogue and a digital gain stage that boosts or reduces the signal picked up by the microphone. If the microphone is low-level (microphone level), the integrated 21.0 dB Pre-amp can be applied to boost the signal to line level.

To tune the MIC Gain (HFK Mode):

1. Initiate a Handsfree call.
2. The near-end subject sets the vehicle's HVAC Fan setting to either Low or Off.
3. The near-end subject places the Handsfree microphone at the closest specified operating distance from the near-end subject.
4. If tuning a Bluetooth Handsfree kit whose volume is controlled by the mobile phone, set the phone's volume to maximum. If tuning a Bluetooth Handsfree kit integrated with the car's stereo system, set the audio head-unit's volume to maximum.
5. The far-end subject speaks the speech level phrase while the near-end subject monitors the MIC GAIN **Peak** statistics.
6. The near-end subject speaks the level speech phrase at approximately "90 dB SPL Average (C) Weighted Fast" measured 25 mm from the speaker's mouth and monitors the MIC GAIN **Peak** statistics.
7. Is the MIC GAIN **Peak** statistics larger during the far-end speech (echo) or the near-end speech? If the far-end speech (echo) is greater continue. If not, go to step 10.
8. The far-end subject speaks the speech level phrase while the near-end subject monitors the MIC GAIN **Peak** statistics.
9. The near-end subject adjusts the HFK Mode **MIC GAIN** so that the MIC GAIN **Peak** statistics reads no more than -6 dBFS. End **MIC GAIN** tuning for far-end speech (echo) is larger than near-end speech.

10. The near-end subject speaks the speech level phrase while the near-end subject monitors the MIC GAIN **Peak** statistics.
11. The near-end subject adjusts the HFK Mode **MIC GAIN** so that the MIC GAIN **Peak** statistics reads no more than -9 dBFS (-15 to -9 dBFS). End **MIC GAIN** tuning for near-end speech is larger than far-end speech (echo).

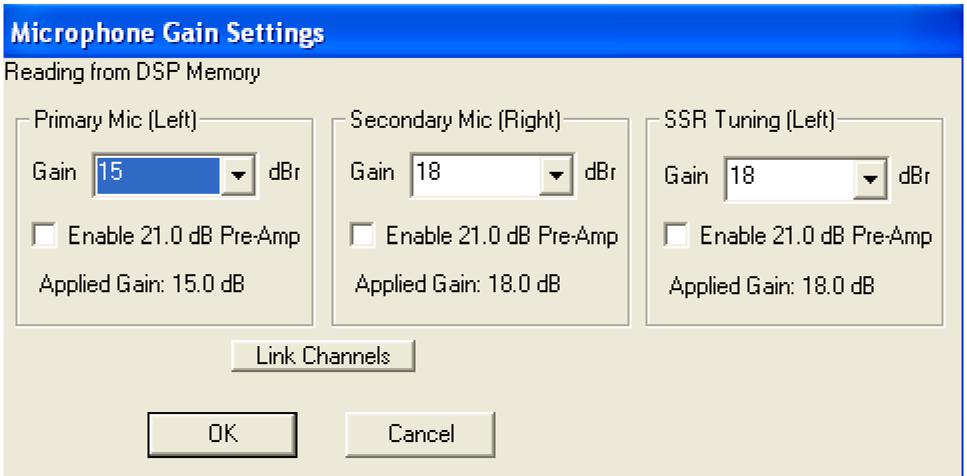


Figure 8-8 Microphone Gain settings window

8.3.2 Send AGC

The Send AGC automatically adjusts the send path send out signal to a specific level determined by the **AGC Target Level** parameter. This parameter compensates for variance in the send out signal levels.

By default, the **AGC Target Level** is -25 dB and needs no additional tuning.

NOTE The Send **AGC Target Level** is the level that the automatic gain control attempts to reach when modifying the send signal.

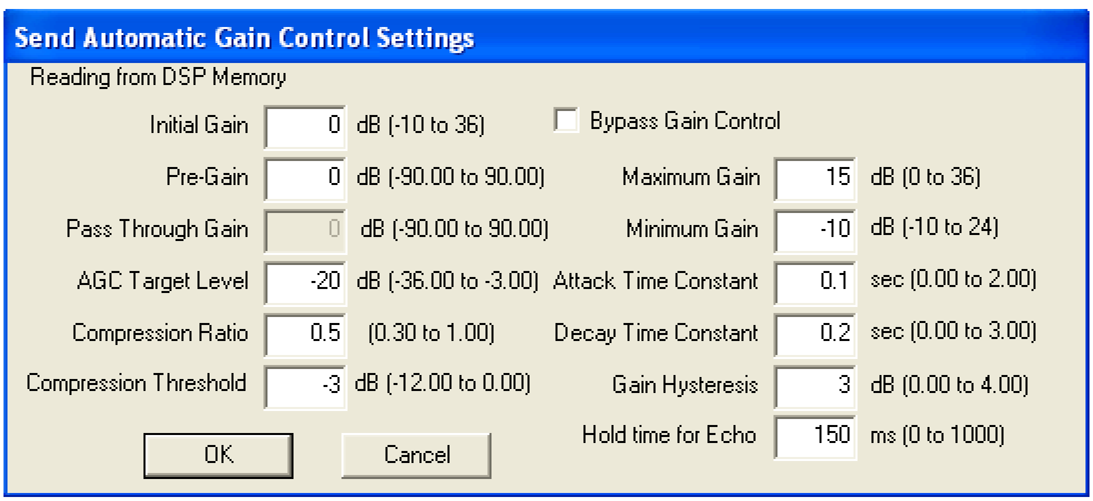
To tune the **AGC Target Level** for Low Echo Coupled Systems (Near-End Send Speech > Speaker Echo):

1. Initiate a Handsfree call.
2. The near-end subject places the primary Handsfree microphone at the closest specified operating distance from the near-end subject.
3. The near-end subject sets the vehicle's HVAC Fan setting to either Low or Off.
4. Next, the near-end subject speaks the level speech phrase at approximately 90 dB SPL Average (C) Weighted Fast, measured 25 mm from the speaker's mouth.
5. The near-end subject adjusts the **AGC Target Level** to achieve the required listening level at the far-end. Ensure the speech is not clipping by monitoring the Send Out **Peak** statistic, avoid saturation. Typically, the Send AGC should not be raised above -10 dB target scale to allow for overshoot, processing in the event of saturation or clipping.
6. The **Maximum Gain** that can be applied to the signal can also be limited.

7. The **Compression Ratio** can also be specified to suit the needs of the application.
8. Make sure that the far-end subject never hears clipped or saturated speech.

To tune the AGC Target Level for High Echo Coupled Systems (Loudspeaker Echo > Near-end Send Speech):

1. Initiate a Handsfree call.
2. The near-end subject sets the vehicle’s HVAC Fan setting to either Low or Off.
3. The near-end subject places the Handsfree microphone at the closest specified operating distance from the loud speaker.
4. The far-end subject speaks the level speech phrase at approximately “90 dB SPL Average (C) Weighted Fast” measured 25 mm from the speaker’s mouth.
5. The near-end subject monitors the Send In **Peak** MIC statistic.
6. The near-end subject adjusts the **Pre-Gain** or **Initial Gain** so that the Send In **Peak** MIC statistic added to the **Pre-Gain** or **Initial Gain** yields levels between (-15 to -9 dBFS).
7. The near-end subject continues to speak the level speech phrase.
8. The near-end subject adjusts the **AGC Target Level** to achieve the required listening level at the far-end. Ensure the speech is not clipping by monitoring the Send Out **Peak** statistic, avoid Saturation. The Send **AGC Target Level** is limited to -3 dB allowing for overshoot.
9. The **Maximum Gain** that can be applied to the signal can also be limited.
10. The **Compression Ratio** can also be specified to suit the needs of the application.
11. Make sure that the far-end subject never hears clipped or saturated voices.



NOTE The Send AGC has two statistics, the Send AGC Applied gain and the Active Speech Level, located directly below the Send AGC tuning block. They are visible while in monitoring mode to aid in the tuning process.

- App Gain: The App. Gain Statistic indicates how much gain the Send AGC is actively applying to the input signal. It also shows how the Send AGC reacts to changes in speech levels.
- Act Spch: The Active Speech Statistic indicates the input level for the active portion of the speech signal as determined by cVc. Together the Active Speech Level and the

Applied Gain are used by cVc to check if the output level of the Send AGC is near the Send AGC Target Level.

8.3.3 Send Noise Suppression (includes Wind Noise Reduction)

This interface controls both the noise suppression and wind noise reduction (WNR) parameters.

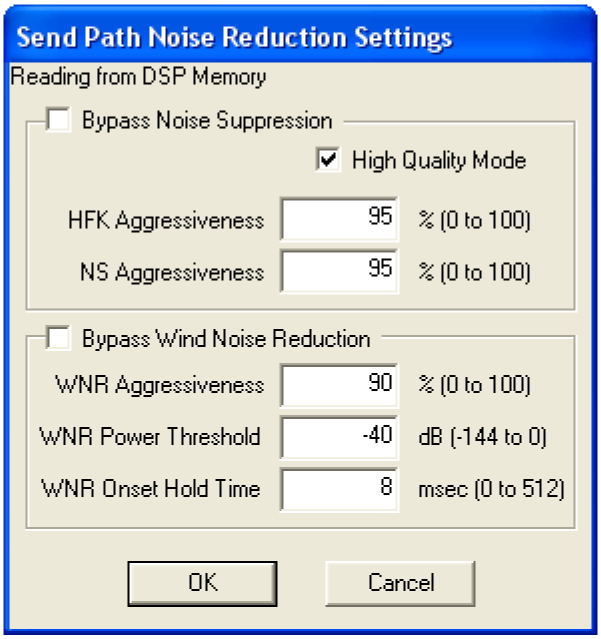


Figure 8-9 Send Path Noise Suppression Settings window

To tune for Noise Suppression:

The **HFK Aggressiveness** parameter is a primary tuning parameter which controls the amount of noise suppression that is applied to the send signal.

1. Initiate a Handsfree call.
2. The near-end subject introduces background noise into the testing environment.
3. The near-end subject continually repeats a test phrase or a normal conversational phrase,.
4. Under different noise conditions, the far-end subjectively evaluates the noise level and its quality with and without near-end speech.
5. Increase the **HFK Aggressiveness** for more noise suppression (at the cost of voice quality). Decrease the **HFK Aggressiveness** for less noise suppression.

NOTE The near-end subject should avoid using the level speech phrase while tuning the HFK Aggressiveness.

Normal conversational speech or phonetically-balanced phrases and passages are better for judging speech intelligibility in the presence of noise.

High Quality Mode is selected (enabled) by default. This does not affect the noise suppression but provides improved speech quality. If un-selected, MIPSis reduced by ~1, but the voice quality slightly degrades.

To tune for Wind Noise Reduction:

The **WNR Aggressiveness** is the primary tuning parameter. It controls the amount of wind noise reduction that is applied to the send signal. The **WNR Power Threshold** controls the trigger point when wind is detected and the **WNR Onset Hold Time** parameter determines how long to wait once wind is detected before WNR is applied.

1. The near-end subject stops, the background noise should be quiet.
2. To begin tuning WNR, set the **WNR Aggressiveness** to 100%, set both **WNR Power Threshold** and **WNR Onset Hold Time** to `0`.
3. The near-end subject adds wind using a controlled source (i.e. fan), attempting to maintain a constant moderate wind speed of approximately 4 mph blowing directly onto the front face of the microphone.
4. Decrease the **WNR Power Threshold** until the wind noise is removed. To ensure the threshold is correct, the near-end subject should switch the WNR on and off by toggling the **Bypass Wind Noise Reduction** button, and checking that the wind noise is removed when WRN is on.
5. Stop the wind, the near-end subject speaks a test phrase or a normal conversational phrase, continually.
6. Record or subjectively evaluate the speech quality. If the speech quality (mainly the on-set of speech) is noticeably degraded, raise the **WNR Onset Hold Time** until good audio is achieved. To ensure the hold time is correct, the near-end subject should switch the WNR on and off by toggling the **Bypass Wind Noise Reduction** button, and checking the speech quality when the WNR is on.
7. Now the near-end subject combines both the moderate wind noise and speech phrases.
8. You can decrease the **WNR Aggressiveness** until the required level of WNR is achieved, or you can leave at maximum (100%).
9. You may need to refine the parameter after the initial tuning.

NOTE Increasing WNR Aggressiveness and decreasing WNR Onset Hold Time achieves more wind noise reduction.

8.3.4 Acoustic Echo Canceller

This section describes areas to fine-tune on the **Acoustic Echo Cancellation Settings** window.

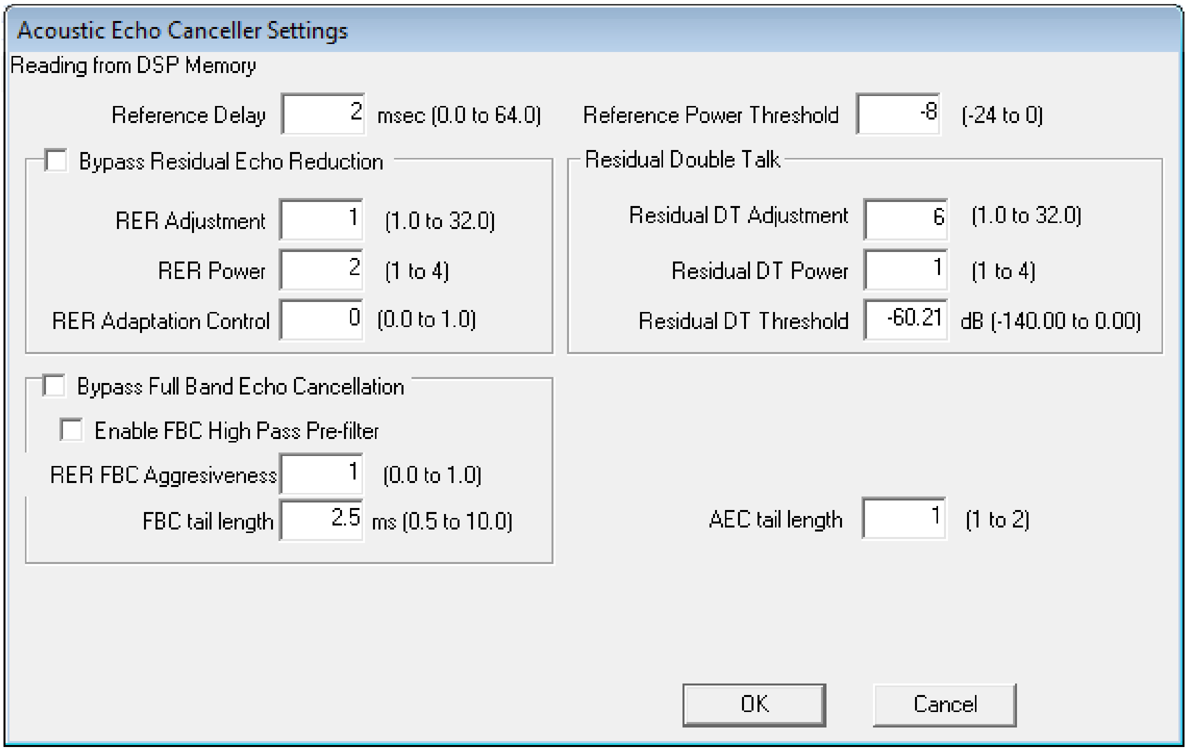


Figure 8-10 Acoustic Echo Cancellation Settings window

Table 8-1 Acoustic Echo Cancellation settings fields

Setting	Description
Bypass Residual Echo Reduction	Enables or disables additional echo cancellation built into the AEC. The intent is to reduce the subtle non-linearity's that could exist after the primary adaptive filter. It is enabled by default. QTIL recommends the **Residual Echo Reduction** feature be enabled, to achieve the best echo cancellation performance.
Reference Delay	Compensates for any time delay of the send signal caused by additional processing preceding the Acoustic Echo Canceller. This feature ensures that the AEC triggers at the correct time. The default reference delay value is 2 ms.
Reference Power Threshold	This is the level in which the receive energy must exceed for the AEC to adapt. Increasing this value helps keep the AEC from diverging when only noise is present on the receive path for long periods of time.
RER Adjustment	Controls the amount of attenuation when receive speech is present. Increasing this value increases double talk performance but can degrade the single talk echo cancellation performance RER attenuation is applied

Table 8-1 Acoustic Echo Cancellation settings fields (cont.)

Setting	Description
RER Power	Controls the number of times the RER attenuation is applied. Increasing this parameter raises the number of times the RER attenuation is applied.
RER DT Adjustment	Controls the amount of attenuation during Double Talk. Increasing this value can degrade echo cancellation performance.
RER DT Power	Controls the number of times the RER attenuation is applied. Increasing this parameter raises the number of times the RER attenuation is applied.
Residual DT Threshold	Controls when to bypass the Residual DT Adjustment and Power. If the noise level statistic exceeds this Residual DT Threshold value only the RER Adjust and Power values are used.
RER Adaptation Control	Controls how fast RER adapts when RCV speech is present. 0 is most conservative and 1 is most aggressive. 0 should be used for automotive applications and 1 for speaker type products.
AEC Tail Length	AEC filter length, 1 means 60 ms and 2 means 120 ms. 1 should be used for automotive applications and 2 for speaker type products.
Bypass Full Band Echo Cancellation	Turns FBC on or off.
Enable FBC High Pass Pre-filter	FBC filter length is usually very short and cannot cover the low frequency resonance, which prevents FBC converging. Therefore, we filter out this low frequency resonance and let FBC converge. Low frequency resonance is introduced by the shaking of shelves carrying the portable speaker.
RER FBC Aggressiveness	Controls RER aggressiveness when FBC is on, its value is between 0 and 1, 0 for most aggressive and 1 for most conservative.
FBC Tail Length	How many taps are used in the FBC Filter, this is displayed in milliseconds

To tune the Acoustic Echo Canceller:

1. Initiate a Handsfree call, set the loudspeaker volume to maximum.
2. The far-end subject speaks short bursts of speech (for example, “one”, “two”, “hello”, “ok”, “check”, “echo”, and so on) and checks for echo at the far-end.
3. If echo is heard on the far end, increase **RER Power** by 0.5 and recheck for echo. Increase as needed.
4. The near end speaks continuous while the far-end speaks short bursts. The far end checks for double talk attenuation.
5. First step in tuning for attenuation during double talk is to raise **RER Adjustment** by steps of 0.5 and retest. If no increase in performance is achieved, return **RER Adjustment** to default, which is a value of 10.
6. If too much attenuation is still applied during double talk, de-select the **Bypass Cross Bin Averaging** check box and retest for double talk attenuation.
7. Turn on FBC, set RER Adaptation Control to 1, and AEC Tail Length to 2.

8. The far-end subject continuously speaks while the near end subject monitors the Power Drop Statistic. Optimal power drop is -10dB.
9. If the power drop is not close to -10, adjust the Reference Delay in 0.5 increments until the optimal delay is determined.

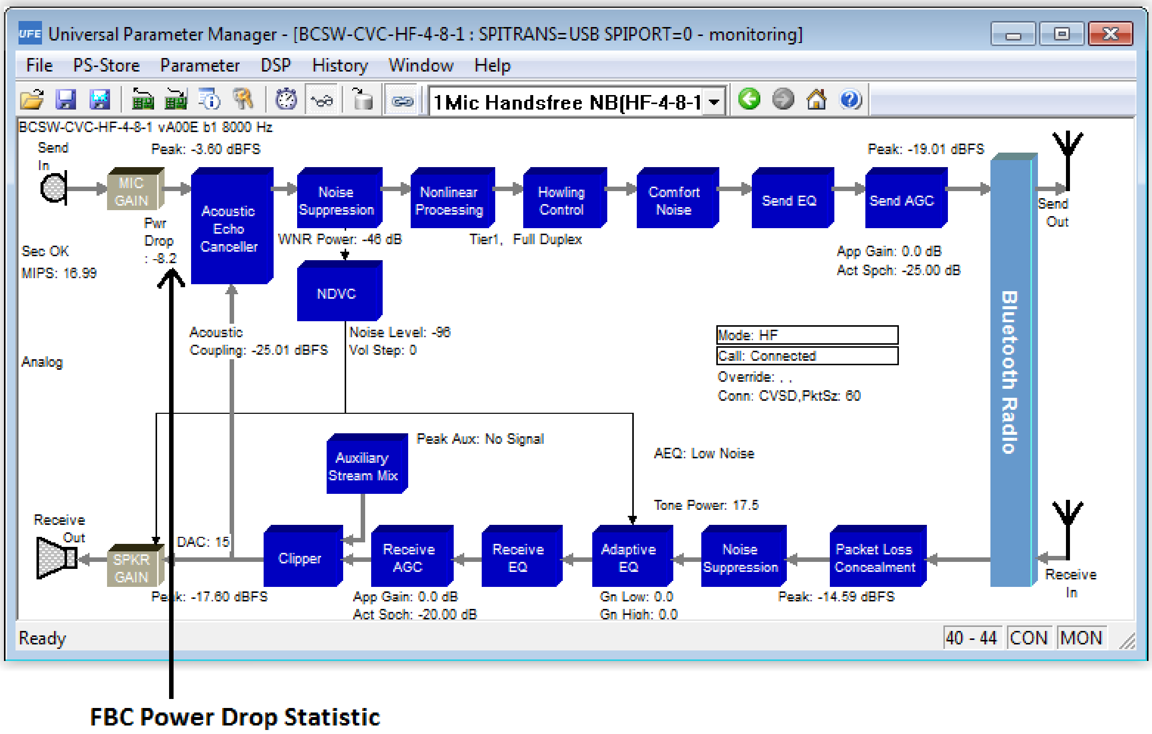


Figure 8-11 Acoustic Echo Cancellation Settings fields

NOTE The **Bypass Residual Echo Reduction** feature enables or disables additional echo cancellation built into the AEC. The echo reduction feature is enabled by default. Leave this feature remain enabled.

8.3.5 Non-linear Processing

The Non-linear Processing block removes residual echo that the primary AEC cannot cancel because f a distortion. It is also compensates for the nonlinearities introduced in the receive path. More often, the reference signal to the AEC is before the volume control of the speaker. Changes in the volume that introduces non-linear effects in the acoustic path is compensated for by using either the Tier 1 or Tier 2 set of heuristic non-linear processing. The Tier 1 set can be used with lower speaker volume levels and the Tier 2 set at higher speaker volume levels. In each tier, the attenuation and the bandwidths, along with the number of bands, can be selected to improve the compensation for the non-linearities introduced.

The **Tier-2 Switching Threshold** of volume level can be selected to transition from Tier 1 to Tier 2 processing. Typically, non-linear processing needs to be tuned to compensate for poor quality speakers and coarse volume control mechanisms. For good quality speakers and volume controls, the non-linear processing is turned off to reduce processing load and improve doubletalk performance.

There are three tunable bounds on the Non-linear Processing Settings window:

- Lower Bound
- Middle Bound
- Upper Bound

There is also a **Number of Bands** and an **Attenuation** field. The Non-linear Processing algorithm looks between the **Lower Bound** and the **Middle Bound** for the average value and compares it to the internal threshold. Based upon this average value, the value in the **Attenuation** field is applied between the Middle Bound and Upper Bound.

Non-linear processing is a two-tier system block. This means that for a receive volume range, two independent non-linear processors can be used. For example, if the receive volume at the Handsfree microphone has a range of 65 dB SPL to 115 dB SPL, the non-linear processing can be set so that Tier One affects residual echo from 65 SPL to 95 dB SPL, and Tier Two from 95 dB SPL to 115 dB SPL. This is set using the setting **Tier-2 Switching Threshold**.

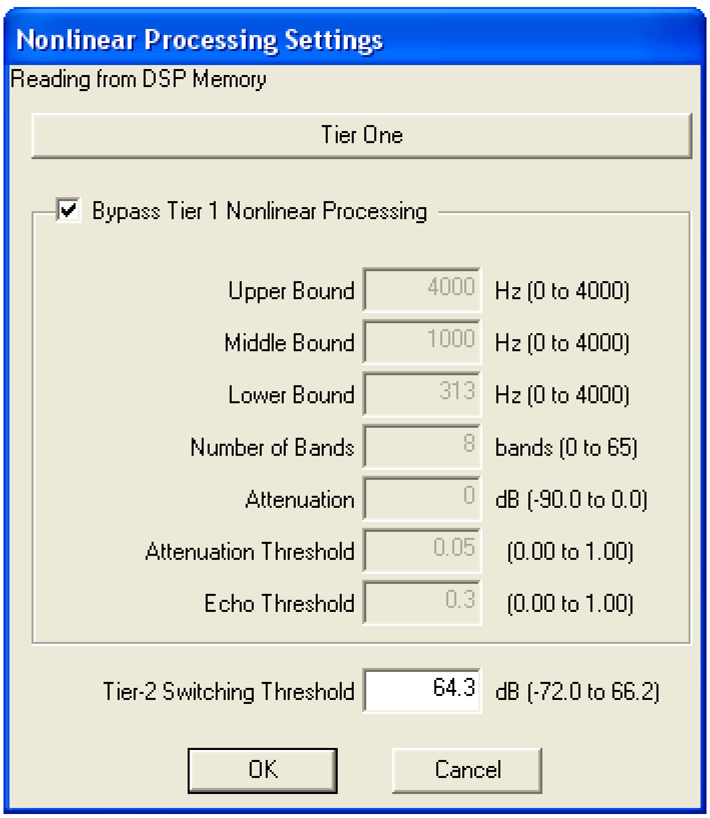


Figure 8-12 Non-linear Processing Settings window

8.3.6 Howling Control

When the echo signal surpasses a certain level, the Howling Control can force the cVc Handsfree device to attenuate the send aggressively (half-duplex).

To fine-tune the Howling Control:

1. Initiate a Handsfree call.
2. While the far-end subject checks for echo, the near-end subject increases the Handsfree system volume until an echo that cannot be cancelled is heard at the far-end.
3. The far-end subject speaks the level speech phrase while the near-end user monitors the **Acoustic Coupling** statistic located below the **Acoustic Echo Canceller** block on the **Parameter Manager** window.
4. To secure an average statistic, repeat this test several times.
5. Click the **Howling Control** block to enter the average statistic in the **Howling Suppression Coupling Threshold** field.
6. The far-end subject checks that there is no echo heard at the volume level determined in Step 2. If echo is present, the near-end reduces the **Howling Suppression Coupling Threshold**.

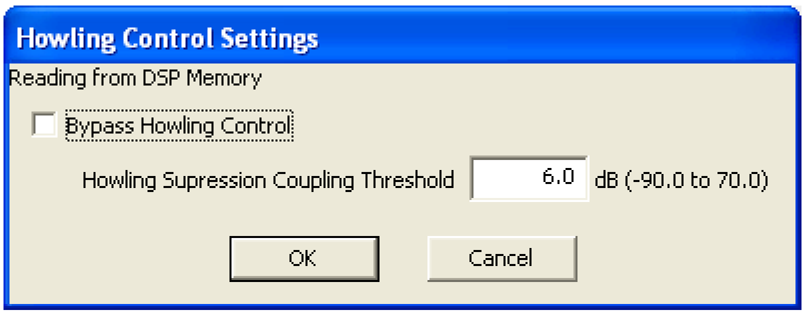


Figure 8-13 Howling Control Settings window

8.3.7 Comfort Noise

The Comfort Noise generator adds noise to the send signal to minimize noise floor fluctuations introduced by the echo cancellation. The Comfort Noise generator has a single gain control.

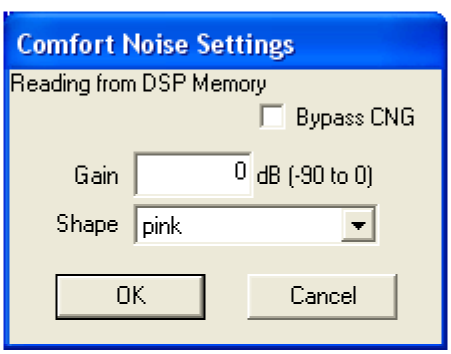


Figure 8-14 Comfort Noise Settings window

To tune the Comfort Noise Generator:

1. Initiate a Handsfree call.
2. The near-end subject introduces steady background noise into the testing environment.
3. The near-end and far-end subjects alternate speaking numbers and letters (for example, “1”, “A”, “2”, “B”, “3”, “C”, and so on) with a 1-second pause between each number and/or letter.

4. The far-end subject listens to any fluctuations in the noise floor during the alternating speech.
5. If fluctuations are heard, increase or decrease the amount of comfort noise.
6. The near-end subject bypasses the Comfort Noise generator.
7. The far-end subject checks for a large decrease in background noise as the Comfort Noise generator is bypassed.
8. If there is a large decrease in background noise, the near-end subject decreases the Comfort Noise generator gain.
9. Shape can be used to choose the weighting of the comfort noise spectrum.

9 Fine-tuning

When the cVc Handsfree receive and send paths are tuned, minor parameter changes may be required to reach a good performance level. Some products have unique acoustic designs or require special Handsfree sound quality requirements for the product. This section describes additional tuning instructions for fine-tuning cVc Handsfree processing modules.

9.1 Receive Path fine tuning

This section describes fine adjustments that can be made to the cVc HF receive path.

9.1.1 Setting minimum speaker gain loudness

The speaker gain tuning to obtain the maximum loudness setting is described in SPKR Gain. Setting the Minimum Loudness for the loudspeaker gain is similar to setting the Maximum Loudness.

1. Initiate a Handsfree call.
2. Adjust the phone volume to minimum.
3. The far-end subject speaks the level speech phrase.
4. The near-end subject measures or listens to the loudspeaker volume. Click on the **SPKR GAIN** block, select the **Temporarily Override DAC** option, and adjust the gain to your required minimum level.
5. Place the **SPKR GAIN** value into the VM volume table using the Headset Configuration Tool. Choose the Audio Gains tab to configure desired number of volume steps as well as minimum.

9.1.2 Receive AGC

To tune the Receive AGC:

1. Adjust the **AGC Target Level** to required value. The default is -20 dB, which provides a good dynamic range with almost full-scale value.
2. Adjust the **Minimum Gain**, which sets the low threshold level for the gain factor. The gain factor does not fall below the **Minimum Gain**.
3. Adjust the **Maximum Gain**, which sets the high threshold level for the gain factor. The gain factor does not exceed above the maximum gain. The **Minimum Gain** and the **Maximum Gain** defines the dynamic range of the gain factor of the AGC.
4. Adjust the **Compression Ratio**, which defines the slope of the compression curve, above the target level. The gain factor follows the compression curve above the target level, while the slope of gain curve below the target level is unity.

9.1.3 Receive EQ

The receive path has a parametric equalizer for enhancing audio quality. By default, the parametric equalizer is set as a high pass filter set to roll off below 125 Hz, but may be used to troubleshoot loudspeaker distortion at specific frequencies or to pass standard measurements (ITU-T). If required, the receive EQ can be used for frequency shaping to fit an appropriate response curve. The GUI enables the Receive EQ parameters to be graphically selected. See *BCSW-cVc-HF-4*-8-2 *1-mic Handsfree Parameter Manager Users Guide* for more details.

9.1.4 Clipper

The Clipper prevents the receive path signal from exceeding a specified maximum level (Clip Point). If the dynamic range of the receive signal is large and causes receiver distortion, the Clipper can be used to limit this distortion. A Clip Point is selected in the Clipper settings to achieve a receive signal limit. The optional Boost adds compression to the clipped signal (such as a loudness boost).

NOTE The Clipper's Boost setting decreases the dynamic range of the receive signal, which degrades speech quality.

To tune the Clipper:

1. Initiate a Handsfree call.
2. Adjust the Handsfree volume to the maximum.
3. The far-end subject speaks the level speech phrase.
4. The near-end subject listens for distortion in the Handsfree receiver.
5. The near-end subject lowers the **Clip Point** by -3 dB steps, until additional distortion is heard.
6. Remove the last -3 dB value of added **Clip Point** and set this as the new **Clip Point**. End tuning the clipper.

Optional Steps:

1. If the Clipper is enabled, the near-end subject can adjust the **Boost** so that the required receiver loudness is maintained. Any boost is hard clipped at the **Clip Point**.
2. If the Clipper is enabled, the near-end subject can adjust the Boost Clip Limit enforcing the maximum digital limit allowed in the path prior to the DAC.

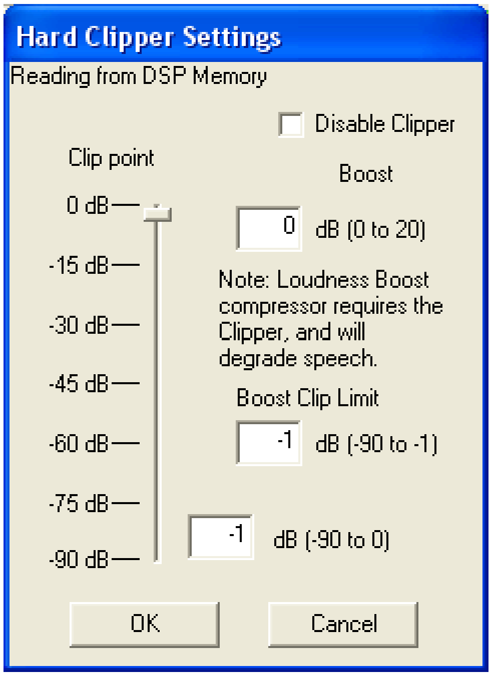


Figure 9-1 Hard Clipper Settings window

9.1.5 Auxiliary stream mix

Auxiliary stream mix is always enabled. Stream mixing only occurs during a stream mix event, such tone play). It mixes an auxiliary signal with the SCO input signal. The auxiliary signal can be tones or another source such as voice prompts. The ratio of the mixture uses the slider controls to achieve the required balance on the receive out signal. Changing a stream mix using a slider inversely controls the other to maintain 100% between the channels.

If the Decouple Gains is checked, the user may separately adjust the mix ratios of the SCO and Auxiliary Streams.

NOTE If the percentage sums >100%, saturation can occur.

The auxiliary signal can be boosted by using the Auxiliary Gain parameter. The maximum limit on the auxiliary gain is 24 dB.

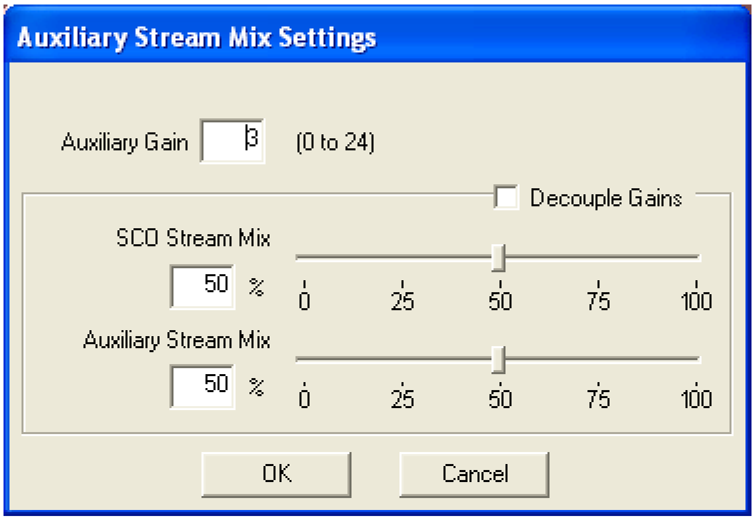


Figure 9-2 Auxiliary stream mix setting window

9.2 Send Path fine tuning

This section describes fine adjustments to the cVc Handsfree send path.

9.2.1 Send EQ

The send path has a parametric equalizer for enhancing audio quality. Normally, the parametric equalization is set flat, but may be applied to the send path signal at specific frequencies or used to pass standard measurements (ITU-T).

If required, the Send EQ can perform some frequency shaping to fit a required response curve. The graphical user interface allows the Send EQ parameters to be graphically selected, see the *BCSW-CVC-HF-4-8-2 1M-HF Parameter Manger User Guide* for details.

Minimize gain (unity) through the EQ to avoid saturation and distortion.

9.2.2 Send AGC

To fine-tune the Send AGC:

1. Adjust the **AGC Target Level** to the required value. The default is -25 dB, which provides a good dynamic range with almost full-scale value.
2. Adjust the **Minimum Gain**, which sets the low threshold level for the gain factor. The gain factor does not fall below the **Minimum Gain**.
3. Adjust the **Maximum Gain**, which sets the high threshold level for the gain factor. The gain factor does not exceed above the maximum gain. The **Minimum Gain** and the **Maximum Gain** defines the dynamic range of the gain factor of the AGC.
4. Adjust the **Compression Ratio**, which defines the slope of the compression curve, above the target level. The gain factor follows the compression curve above the target level, while the slope of gain curve below the target level is unity.

9.3 Auxiliary stream mix 4-8-2 1M-HF

Auxiliary stream mix is always enabled. Stream mixing only occurs during a stream mix event, such tone play). It mixes an auxiliary signal with the SCO input signal. The auxiliary signal can be tones or another source such as voice prompts. The ratio of the mixture uses the slider controls to achieve the required balance on the receive out signal. Changing a stream mix using a slider inversely controls the other to maintain 100% between the channels.

If the Decouple Gains is checked, the user may separately adjust the mix ratios of the SCO and Auxiliary Streams.

NOTE If the percentage sums >100%, saturation can occur.

The auxiliary signal can be boosted by using the Auxiliary Gain parameter. The maximum limit on the auxiliary gain is 24 dB.

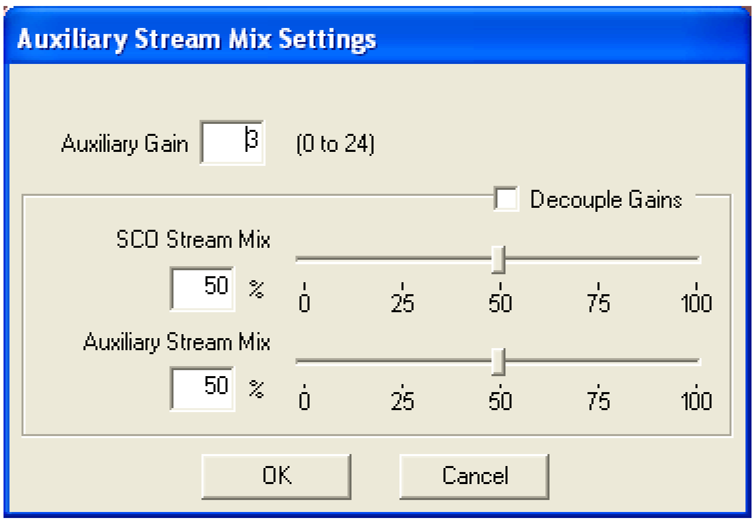


Figure 9-3 Auxiliary stream mix setting window

10 Advanced tuning

10.1 Advanced tuning features

For Handsfree products that offer adequate fidelity you can enable advanced cVc features that include the NVDC and AEQ.

10.1.1 Noise-Dependent Volume Control

The NDVC should be tuned after the MIC Gain.

The NDVC automatically increases or decreases the loudspeaker volume depending on the level of noise in the environment. The gain added from NDVC to the **SPKR Gain** should not exceed the maximum output level.

A properly tuned NDVC results when the DAC proportionally increases with the near end noise level, creating a constant SNR environment for the near end listener.

To tune the NDVC:

1. Initiate a Handsfree call.
2. Click on the NDVC processing block and remove the checkmark from the Bypass NDVC check box. This enables the NDVC.
3. Increase the noise floor inside the vehicle to the required level at which the NDVC should start to adjust the volume (for example, driving at a speed of 20 mph).
4. Monitor the Noise Level statistic and type this value in the Min noise level field.
5. Increase the noise floor inside the vehicle to the required maximum level at which the NDVC should remain turned on (for example, driving at a speed of 60 mph). Monitor the Noise Level statistic and place this value in the Max noise level field.
6. During the high “road” noise condition, the near-end subject determines the maximum gain that the NDVC can apply and enters this number in the Maximum NDVC Gain Limit field.

- The NDVC does not adjust the **SPKR Gain** over 0 dB. For example, if the SPKR Gain is -12 dB and the Maximum NDVC Gain Limit is 15 dB, the NDVC does not apply more than 12 dB of gain. Under the highest noise condition, the maximum loudness is limited to SPKR Gain at 0 dB or the highest level defined in the Volume table.
- The Total SPKR Gain = SPKR Gain + Maximum NDVC Gain Limit
- The Total SPKR Gain is important when tuning the Send Path.
- The NDVC gain change is quantized based on the DAC resolution of roughly 3 dB per step. For example, setting the Maximum NDVC Gain Limit to 15 dB equals 5 steps on the DAC.

1. Set the **Hysteresis** to a value between zero and one. Higher values reduce the NDVC sensitivity when reacting to changes in the background noise and lower values increase the sensitivity. The default value is 0.75.
2. Adjust the **Increasing Noise Attack Time Constant** and **Decreasing Noise Decay Time Constant** to a required level. High time constant values cause the NDVC to react more slowly to changes in the background noise and lower values cause a quicker reaction.

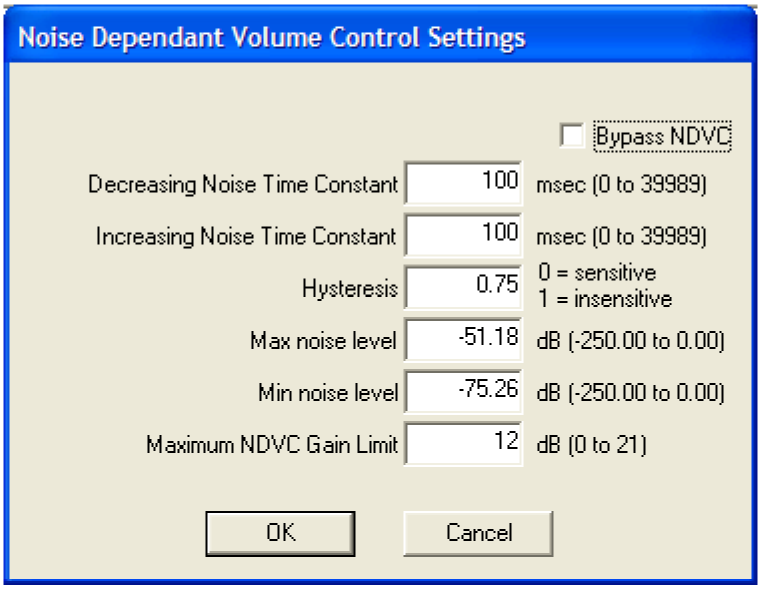


Figure 10-1 Noise Dependent Volume Control Settings window

NOTE Figure 10-1 shows the default settings. Adjust the **Max noise level** and **Min noise level** for your specific Handsfree device.

10.1.2 Adaptive EQ (AEQ)

There are three systems available for the CSR86xx:

- Narrow Band plus High Frequency Emphasis: An AEQ (0 to ~3.5 kHz) plus the addition of an Emphasis Band (~3.5 to 4 kHz).
- Narrow Band plus Frequency Expansion: An AEQ (0 to ~3.5 kHz) plus the addition of Expanded Bands (~3.5 to ~6.2 kHz).
- Wide Band: An AEQ (0 to 8 kHz).

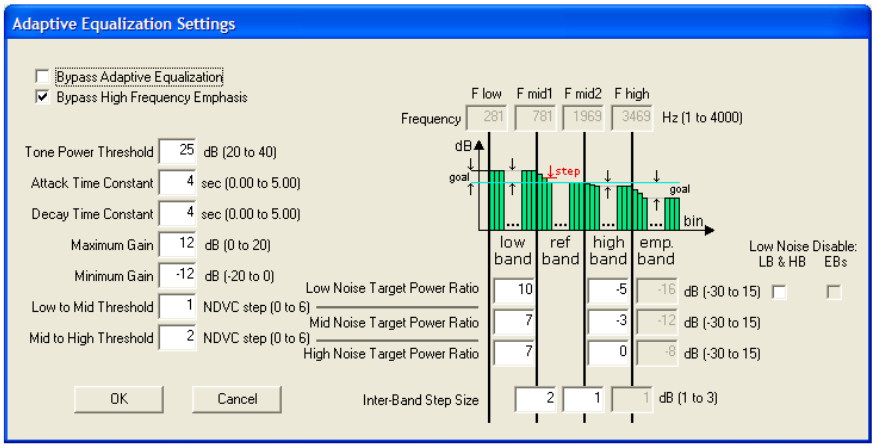
When Adaptive Equalization block is enabled, it improves the intelligibility of the receive path voice signal in the presence of near end noise. It does this by altering the spectral shape of the receive path signal while maintaining the overall power level.

Consonants, which are dominantly high frequency based and much lower in amplitude than vowels, significantly contribute to the intelligibility of the voice signal. In the presence of noise, the lower amplitude consonants become masked by this noise. By increasing the frequency components that contribute to the consonants while in the presence of noise, the intelligibility can be improved.

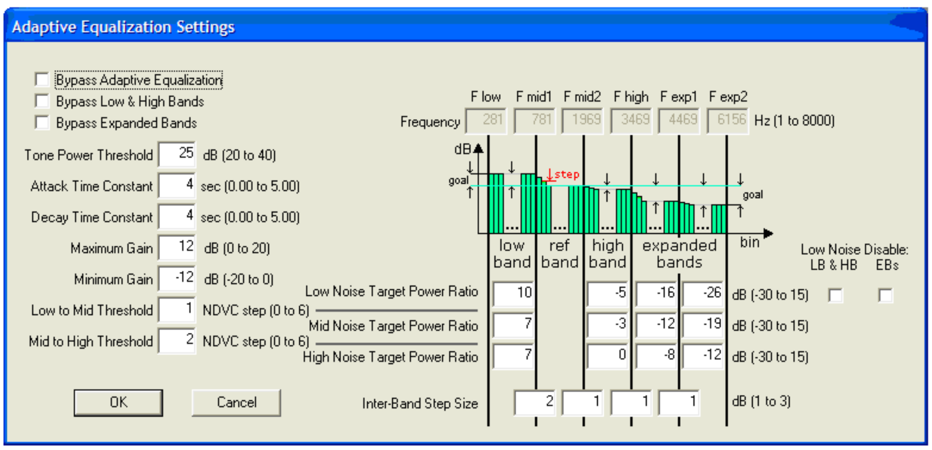
To maintain a consistent amplitude level, the Adaptive Equalization block adaptively increases the high frequencies relative to the middle frequencies while reducing low frequencies accordingly.

The Adaptive Equalization block can compensate for variations in voice transmission channels, which include far-end devices and telecommunication channels.

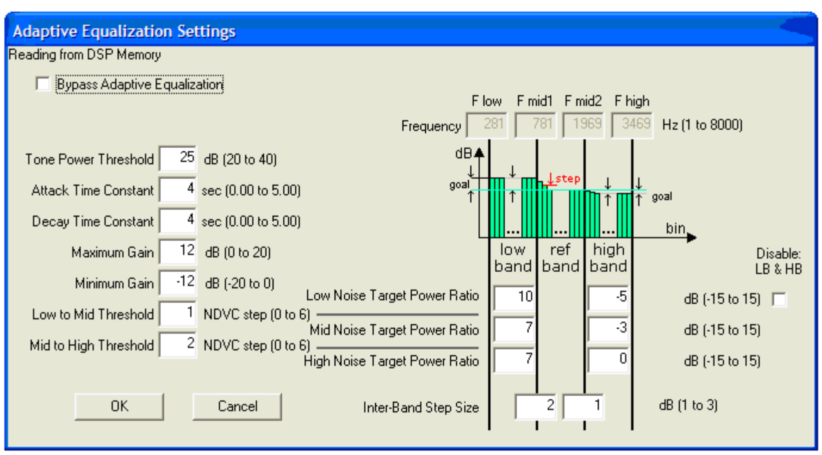
NOTE For the Handsfree device to benefit from this feature, the loudspeaker must provide adequate fidelity delivered to the ear of the user and the NDVC has been enabled and tuned.



Narrow Band AEQ plus High Frequency Enhancement



Narrow Band AEQ plus Frequency Expansion



Wide Band AEQ

Figure 10-2 Adaptive Equalization Settings window

The AEQ applies one of three user shaped curves. These curves are shown as the **Low**, **Mid** and **High Noise Target Power Ratio**. The user can shape the curves by setting the low and high band goals in dB. In quiet conditions the **Low Noise Target Power Ratio** curve is applied. When the **Low to Mid Threshold** is crossed, the **Med Noise Target Power Ratio** curve is applied and finally when the **Mid to High Threshold** is crossed the **High Noise Target Power Ratio** curve is applied.

NOTE There is the option to bypass application of the Adaptive EQ in quiet situations by selecting the **Low Noise Disable LB & HB** option. If this option is selected the adaptive EQ in Mid and High noise situations is still applied.

The AEQ uses the NDVC step (shown as **Vol Step: x** in the Parameter Manager monitor window) statistic to determine the switch points from the Low, Mid to High Noise Target Power Ratio curves. QTIL recommends that the step transitions be placed evenly across the range of NDVC steps available (in this case 0 to 5).

Figure 10-3 shows the NDVC at **Vol Step: 2**, which triggers the Low to Mid Threshold switching from the **Low** to **Mid Noise Target Power Ratio** curve.

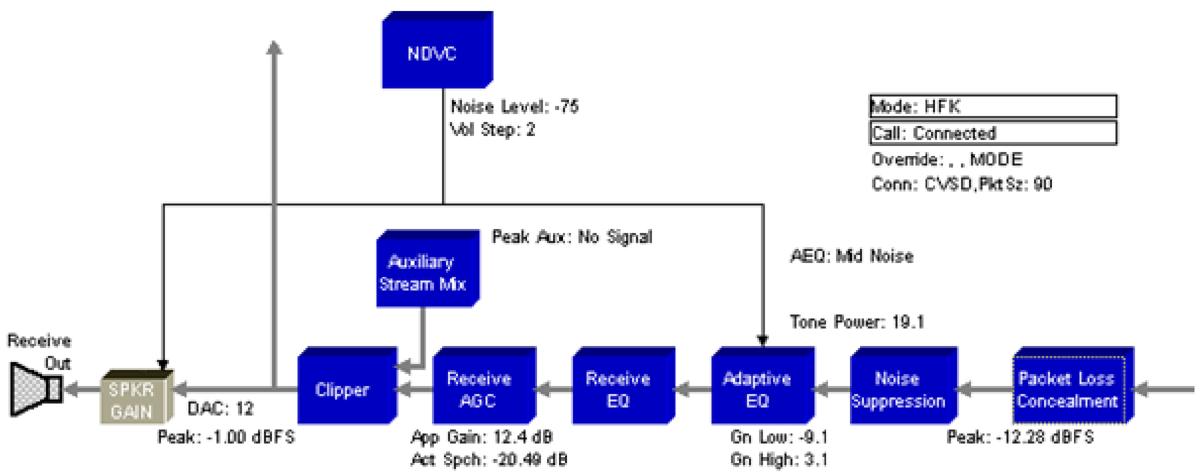


Figure 10-3 Adaptive Equalization switching to mid noise tier

Figure 10-4 shows the NDVC at **Vol Step: 3**, which triggers the **Mid to High Threshold** switching from the **Medium** to the **High Noise Target Power Ratio** curve.

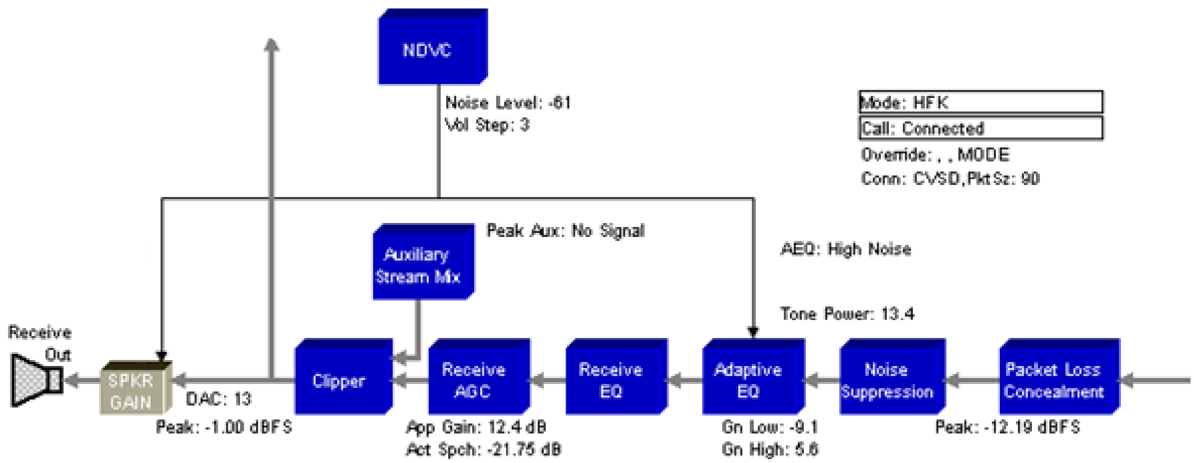


Figure 10-4 Adaptive Equalization switching to high noise tier

To tune the AEQ:

1. Initiate a Handsfree call.
2. To isolate the Adaptive Equalization, select the **Bypass Expanded Bands** or **Bypass High Frequency Emphasis** options (bypassed throughout the Adaptive Equalization tuning) to disable any high frequency expansion/enhancemen.
3. Disable the Adaptive Equalization by selecting the **Bypass Adaptive Equalization** option.
4. The near-end user listens to the original receive speech in a low noise environment (as confirmed by the AEQ noise level statistic).
5. After listening to about thirty (30) seconds of receive speech, de-select the **Bypass Adaptive Equalization** box and listen to the receive speech (again, for about 30 seconds).
6. Adjust the spectral shape of the low and high bands by raising/lowering the **Low Noise Target Power Ratio** parameters under the appropriate low/high band column(s).
7. The low and high bands can be disabled for low noise by selecting the **Low Noise Disable LB & HB** box.
8. Disable the Adaptive Equalization by selecting the **Bypass Adaptive Equalization** option.
9. The near-end user listens to the original receive speech in a medium noise environment (as confirmed by the AEQ noise level statistic).
10. After listening to about thirty (30) seconds of receive speech, de-select the **Bypass Adaptive Equalization** box and listen to the equalized receive speech (again, for approximately 30 seconds).
11. Adjust the spectral shape of the low and high bands by raising/lowering the **Mid Noise Target Power Ratio** parameters under the appropriate low/high band column(s).
12. Disable the Adaptive Equalization by selecting the **Bypass Adaptive Equalization** option.
13. The near-end use listens to the original receive speech in a high noise environment (as confirmed by the AEQ noise level statistic).

14. After listening to about thirty (30) seconds of receive speech, de-select the **Bypass Adaptive Equalization** box and listen to the equalized receive speech (again, for approximately 30 seconds).
15. Adjust the spectral shape of the low and high bands by raising/lowering the **High Noise Target Power Ratio** parameters under the appropriate low/high band column(s).

10.1.3 AEQ with high frequency emphasis or expansion

10.1.3.1 Narrow band plus high frequency emphasis

High Frequency Emphasis can be turned on by un-selecting the **Bypass High Frequency Emphasis** option.

High Frequency Emphasis repairs speech information (3469 Hz to 4000 Hz) lost due to low pass filtering occurring on the PSTN, Cellular Network and Bluetooth connection. Information contained in the original speech from 281 Hz to 3469 Hz reconstructs the lost high frequency content.

Controls are provided in the column **emp. band** to adjust the amount of reconstructed high frequency content that is added to the original speech signal relative to the amount found in the reference speech band (781 Hz to 1969 Hz).

For example, if **emp. band** has a value of -16 dB, the reconstructed high frequency signal added to the original speech signal is 16 dB lower than what is found in the reference speech band.

As with Adaptive Equalization, the amount of reconstructed high frequency speech can be adjusted depending on the level of the acoustic background noise. Adjusting the **Noise Target Power Ratios** defines how much of the reconstructed speech signal is added based on what the NDVC has set the value of **Vol.. Step** to.

10.1.3.2 Narrow Band plus Frequency Expansion

To turn on Frequency Expansion, un-select the **Bypass Expanded Bands** option.

Frequency Expansion repairs speech information (3469 Hz to 6156 Hz) lost due to low pass filtering occurring on the PSTN, Cellular Network and Bluetooth connection. Information contained in the original speech from 281 Hz to 3469 Hz reconstructs the lost high frequency content.

Controls are provided in the **expanded bands** columns to adjust the amount of reconstructed high frequency content that is added to the original speech signal relative to the amount found in the reference speech band (781 Hz to 1969 Hz).

For example, if **expanded bands** has values of -16 dB (for 3469 Hz to 4469 Hz) and -26 dB (for 4469 Hz to 6156 Hz), the reconstructed high frequency signal added to the original speech signal is 16 dB and 26 dB lower than what is found in the reference speech band, respectively.

The amount of reconstructed high frequency speech can be adjusted depending on the level of the acoustic background noise. Adjusting the **Noise Target Power Ratios** defines how much of the reconstructed speech signal is added based on what the NDVC has set the value of **Vol. Step** to.

To tune the High Frequency Emphasis Portion of AEQ:

1. Initiate a Handsfree call.
2. Disable the High Frequency Emphasis by selecting the **Bypass High Frequency Emphasis** option.

3. The near-end user listens to the original receive speech in a low noise environment (as confirmed by the AEQ noise level statistic).
4. After listening to approximately thirty (30) seconds of receive speech, de-select the **Bypass High Frequency Emphasis** option and listen to the receive speech (again, for approximately 30 seconds).
5. Adjust the spectral shape of the emphasis band by raising/lowering the **Low Noise Target Power Ratio** parameters under the appropriate **emp. band** column.
6. The Emphasis Band can be disabled for low noise by selecting the **Low Noise Disable EBs** option.
7. Disable the High Frequency Emphasis by selecting the **Bypass High Frequency Emphasis** option.
8. The near-end user listens to the original receive speech in a medium noise environment (as confirmed by the AEQ noise level statistic).
9. After listening to approximately thirty (30) seconds of receive speech, de-select the **Bypass High Frequency Emphasis** option and listen to the emphasis receive speech (again, for approximately 30 seconds).
10. Adjust the spectral shape of the emphasis band by raising/lowering the **Mid Noise Target Power Ratio** parameters under the appropriate **emp. band** column.
11. Disable the High Frequency Emphasis by selecting the **Bypass High Frequency Emphasis** option.
12. The near-end user listens to the original receive speech in a high noise environment (as confirmed by the AEQ noise level statistic).
13. After listening to about thirty (30) seconds of receive speech, de-select the **Bypass High Frequency Emphasis** option and listen to the emphasis receive speech (again, for approximately 30 seconds).
14. Adjust the spectral shape of the emphasis band by raising/lowering the **High Noise Target Power Ratio** parameters under the appropriate **emp. band** column.

To tune the Frequency Expansion Portion of AEQ:

1. Initiate a Handsfree call.
2. Disable the Frequency Expansion by selecting the **Bypass Expanded Bands** option.
3. The near-end user listens to the original receive speech in a low noise environment (as confirmed by the AEQ noise level statistic).
4. After listening to approximately thirty (30) seconds of receive speech, de-select the **Bypass Expanded Bands** option and listen to the receive speech (again, for approximately 30 seconds).
5. Adjust the spectral shape of the expanded bands by raising/lowering the **Low Noise Target Power Ratio** parameters under the appropriate **expanded bands** column(s).
6. The expanded bands can be disabled for low noise by selecting the **Low Noise Disable EBs** option.
7. Disable the Frequency Expansion by selecting the **Bypass Expanded Bands** option.
8. The near-end user listens to the original receive speech in a medium noise environment (as confirmed by the AEQ noise level statistic).
9. After listening to about thirty (30) seconds of receive speech, de-select the **Bypass Expanded Bands** option and listen to the expanded receive speech (again, for approximately 30 seconds).

10. Adjust the spectral shape of the expanded bands by raising/lowering the Mid Noise Target Power Ratio parameters under the appropriate **expanded bands** column(s).
11. Disable the Frequency Expansion by selecting the **Bypass Expanded Bands** option.
12. The near-end user listens to the original receive speech in a high noise environment (as confirmed by the AEQ noise level statistic).
13. After listening to about thirty (30) seconds of receive speech, de-select the **Bypass Expanded Bands** option and listen to the expanded receive speech (again, for approximately 30 seconds).
14. Adjust the spectral shape of the expanded bands by raising/lowering the **High Noise Target Power Ratio** parameters under the appropriate **expanded bands** column(s).

Document references

Document	Reference
BCSW-CVC-HF-4-8-2 1M-HF Parameter Manger User Guide	80-CT421-1 /CS-30009820-UG

Terms and definitions

ADC	Analogue to Digital Converter
ADK	Audio or Application Development Kit
AEC	Acoustic Echo Cancellation
AEQ	Adaptive Equalizer
AGC	Automatic Gain Control
AT	Attention (modem command prefix)
B&K	Bruel & Kjaer
BCSW	BlueCore Software
BlueCore®	Group term for QTIL's range of Bluetooth wireless technology chips.
Bluetooth	Set of technologies providing audio and data transfer over short-range radio connections.
CDMA	Code Division Multiple Access
CODEC	Coder Decoder
cVc	Clear Voice Capture DSP audio processing software
CVSD	Continuous Variable Slope Delta Modulation
DAC	Digital to Analogue Converter
DMSS	Dual Microphone Signal Separation
DSP	Digital Signal Processor
DUT	Device Under Test
e.g.	*exempli gratia*, for example
IC	Integrated Circuit
IEC	International Electrotechnical Commission
ITU	International Telecommunication Union
ITU-T	International Telecommunication Union-Telecommunication
MIC	Microphone
MIPS	Million Instructions Per Second
mSBC	Modified Sub Band Coding
NB	Narrow Band
NDVC	Noise Dependent Volume Control
NS	Noise Suppression
PEQ	Parametric Equalization

PLC	Packet Loss Concealment
PM	Parameter Manager
PS Key	Persistent Store Key
QTIL	Qualcomm Technologies International, Ltd.
RMS	Root Mean Square
SCO	Synchronous Connection-Oriented Link
SDK	Software Development Kit
SNR	Signal to Noise Ratio
SPI	Serial Peripheral Interface
SPKR	Loudspeaker
SPL	Sound Pressure Level
SSR	Simple Speech Recognition
ST	Side Tone
STMR	Side Tone Masking Rating
THD+N	Total Harmonic Distortion + Noise
UFE	Universal Front End
VM	Virtual Machine
WB	Wide Band
WNR	Wind Noise Reduction