Qualcomm Technologies International, Ltd.

# BCSW-CVC-HS-4-9-2 1M-HS

## Tuning Guide

80-CT419-1 Rev. AJ

November 7, 2017




Qualcomm Technologies International, Ltd. (formerly known as Cambridge Silicon Radio Limited) is a company registered in England and Wales with a registered office at: Churchill House, Cambridge Business Park, Cowley Road, Cambridge, CB4 0WZ, United Kingdom.
Registered Number: 3665875 | VAT number: GB787433096

# Revision history

| Revision | Date | Description |
|---|---|---|
| 1 | December 2013 | Original publication of this document. Alternate document number: CS-00309814-UG |
| 2 | December 2013 | Minor editorial corrections |
| 3 | March 2014 | Edit to the content into the current template |
| 4 | May 2014 | Editorial updates |
| 5 | July 2014 | Updated for ADK 3.5 |
| 6 | May 2015 | Updated Part Number Matrix |
| 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 |

# Contents

# Tables

# Figures

# 1 Introduction

This document is for the headset manufacturers' audio developers. It describes how to tune Clear Voice Capture (cVc) 1-mic Headset (HS) audio processing software running on Qualcomm® Technologies International, Ltd (QTIL) multimedia Integrated Circuits (IC).

Parameter Manager is a Windows PC-based configuration tool that communicates with QTIL's Qualcomm® BlueCore™ IC to simplify the tuning process. The Parameter Manager tool monitors audio signal statistics and adjusts cVc HS audio processing block parameters to achieve optimal audio performance.

## 1.1 cVc overview

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

You can perform the cVc tuning process at several stages during the headset system's development cycle. Typically, a developer fabricates a prototype headset system that packages audio, power, communication and processing components. You can begin tuning when a prototype system is available. Complete final tuning to verify optimal performance when the final production components and packaging are available.

The cVc HS application includes default settings for its audio processing. These defaults may require gain adjustment to compensate for variations in the hardware design, such as microphones and speakers.

## 1.2 Supported software versions

This document describes the audio adjustments of the CVC BCSW-CVC-HS-4-9-2 algorithm. QTIL recommends the same audio tuning procedure 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 (8 k) | B (16 K) | cVc License Key Part Number |
|---|---|---|---|---|---|
| CSR8670 (Flash) | BCSW-CVC-HS-4-9-2 | 0xB012 | Yes | Yes | BCSW-CVC-HS-1M-Fx |
| CSR8675 (Flash) | BCSW-CVC-HS-4-9-2 | 0xB012 | Yes | Yes | BCSW-CVC-HS-1M-Fx |

**NOTE** CSR8670,CSR8675 flash ICs supports narrow band (NB) using CVSD and includes wide band (16 kHz sample rate) using modified sub band coding (mSBC).
To download the CSR86xx UFE installer, go to www.csrsupport.com.

## 1.3 8th Generation new features

This new features and improvements in this release that improve performance and/or affect the tuning process include:

- cVc Generation 8 features support
- Simple Speech Recognition (SSR) functionality
- Updated defaults in the Acoustic Echo Cancellation (AEC) block
- Added parameter for noise Shape in the Comfort Noise (CNG) block
- Updated defaults in the Auxiliary Stream Mix

## 1.4 Before you begin

QTIL recommends that you become familiar with the principles of acoustic performance and the tuneable parameters supported by the Parameter Manager tool before you tune your devices. Figure 2-7 shows the tuning process.

# 2 Prerequisites

This section describes the prerequisites for cVc Headset tuning.

## 2.1 SPI communication protocol drivers

The QTIL Application Development Kit (ADK) software includes the Serial Peripheral Interface (SPI) drivers required by the cVc HS Parameter Manager tool. 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.

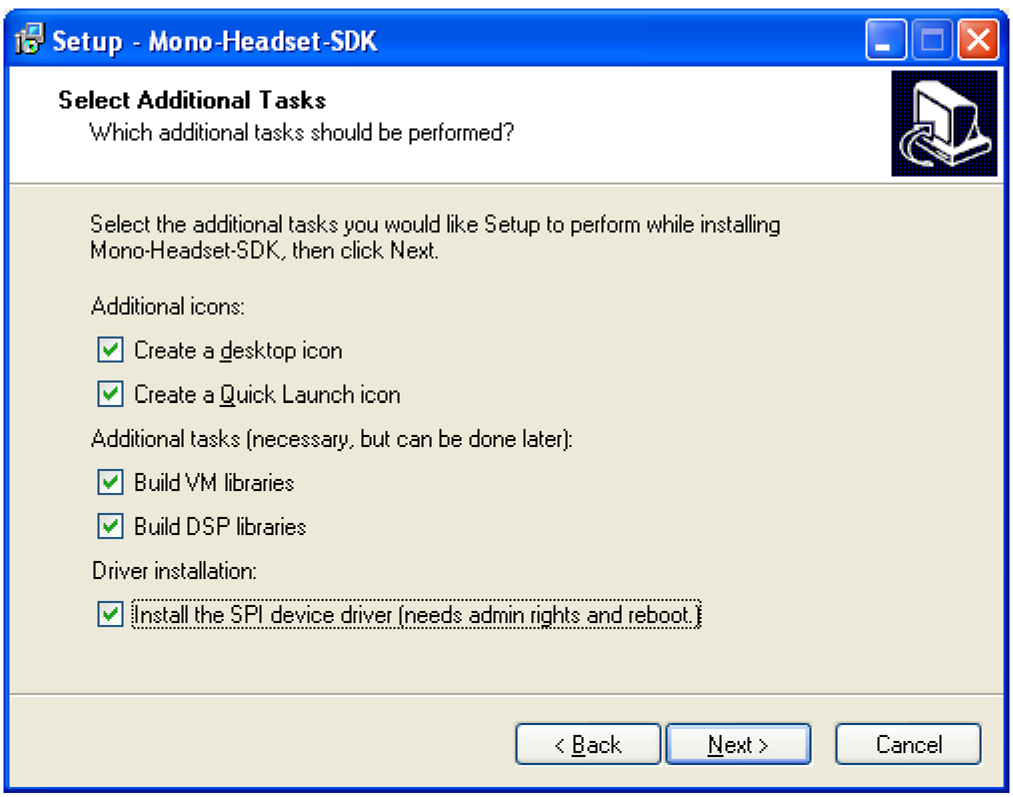



**Figure 2-1 QTIL ADK software setup window with driver installation option selected**

## 2.2 Hardware interfaces

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

## 2.3 Parameter manager tool and user guide

To assist in the tuning process, the Parameter Manager:

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

The *BCSW-cVc-HS-4-9-2 Parameter Manager User Guide* describes how to use the tool and explains the cVc parameters, their configuration, valid parameter ranges and their number formats.

Access Parameter Manage through the Universal Front End (UFE) Application. QTIL provides a Windows Installer application for ROM ICs, which is included with the ADK.

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:

**`Start -> All Programs -> <ADK Name> -> Tools -> UniversalFrontEnd`**

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

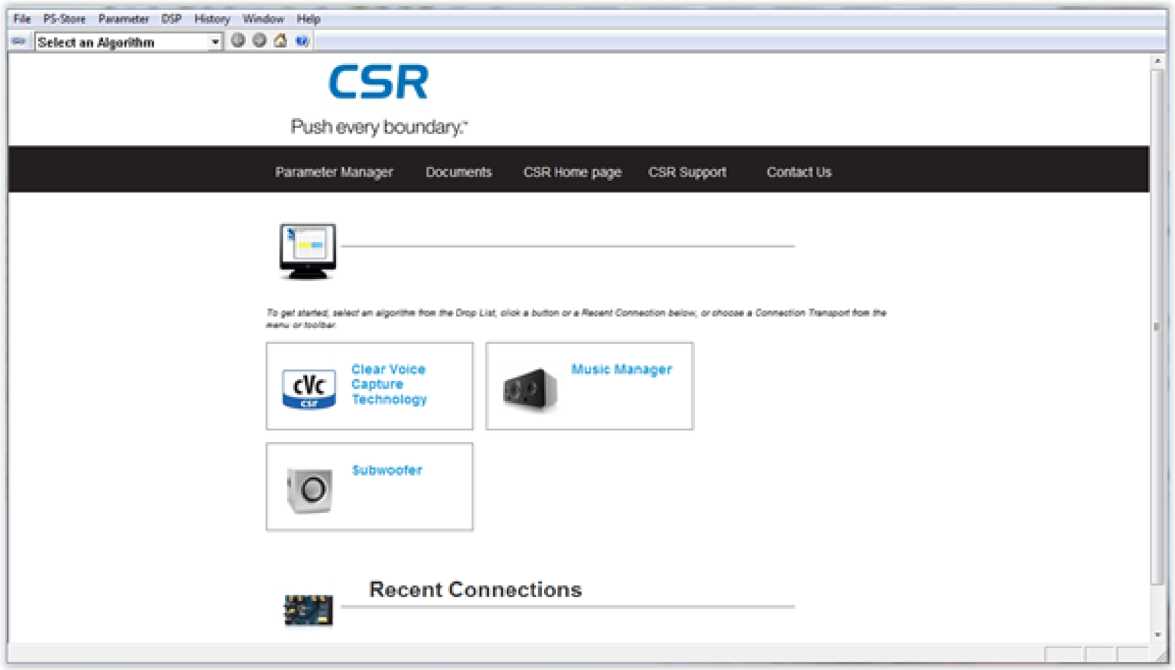



**Figure 2-2 Accessing Headset Parameter Manager from UFE**

Figure 2-3 shows the 1-mic CVC-HS-4-9-2 Headset Parameter Manager window that displays when the application is started. This signifies that the application is connected and in the Static mode. The application is in a connected mode for a narrow band (NB) 8 Khz sample rate system.

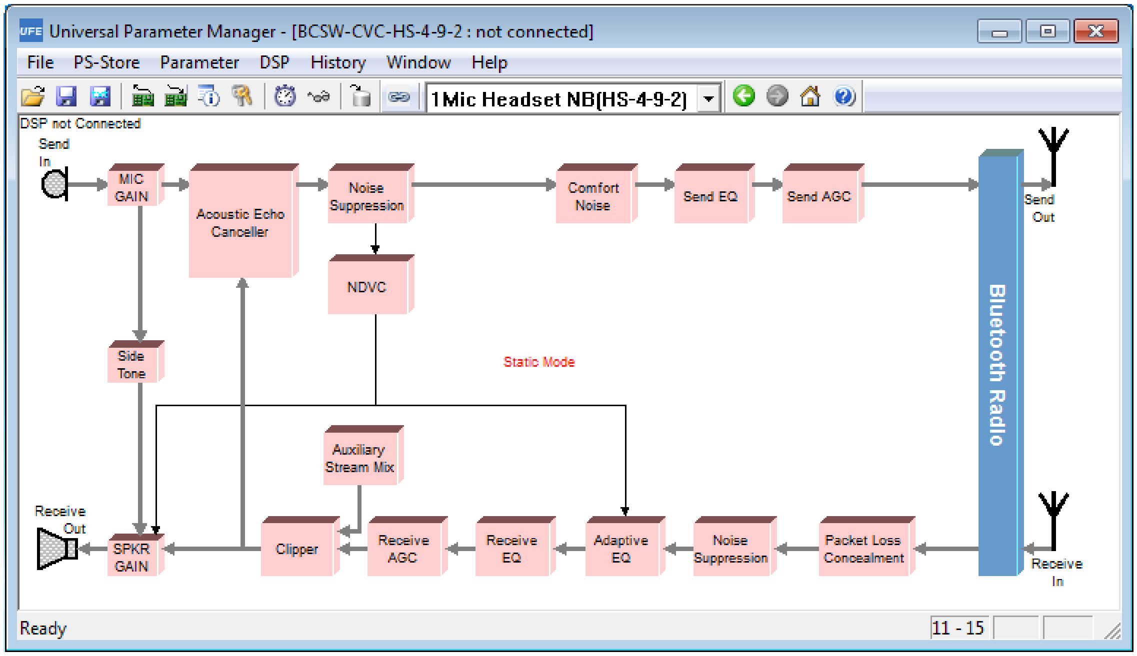



**Figure 2-3 Parameter Manager window, Static mode (narrow band example)**

Figure 2-4 shows an example screenshot in Monitoring mode. This mode provides live feedback and statistics while the algorithm is running during an active call.

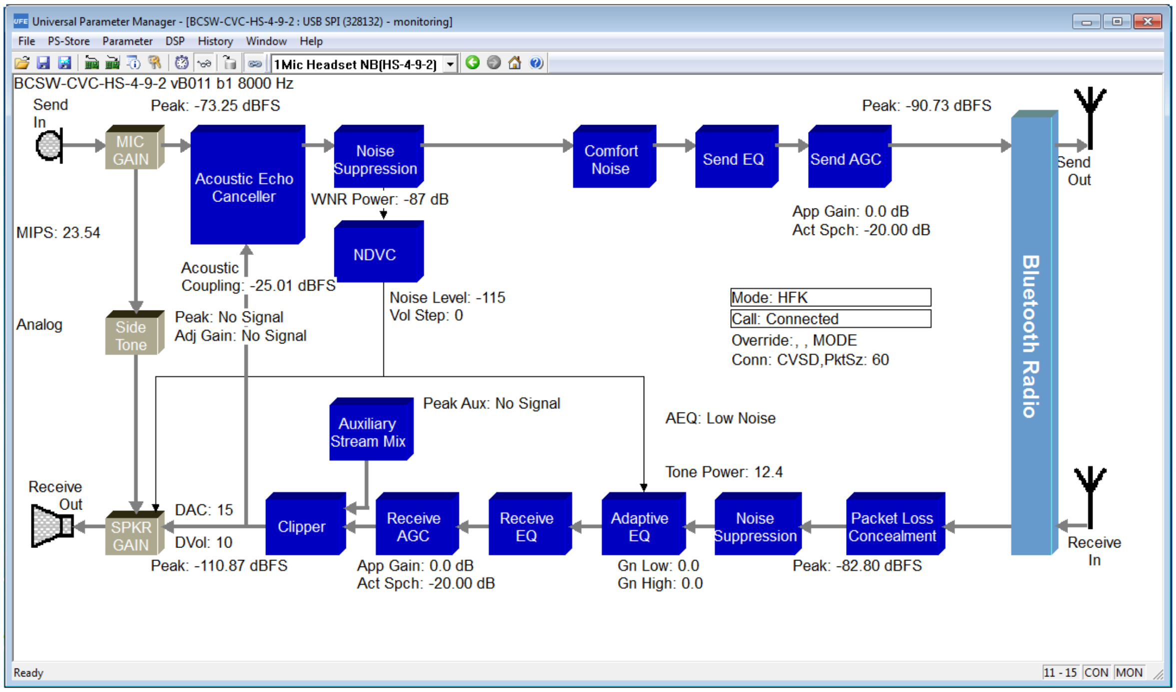



**Figure 2-4 Tuning overview**

This section describes the process for tuning the cVc HS audio processing subsystem, and suggests a tuning process sequence for cVc headset applications.

The tuning process involves adjusting each of these major processing blocks and setting the gains at each cVc interface point.

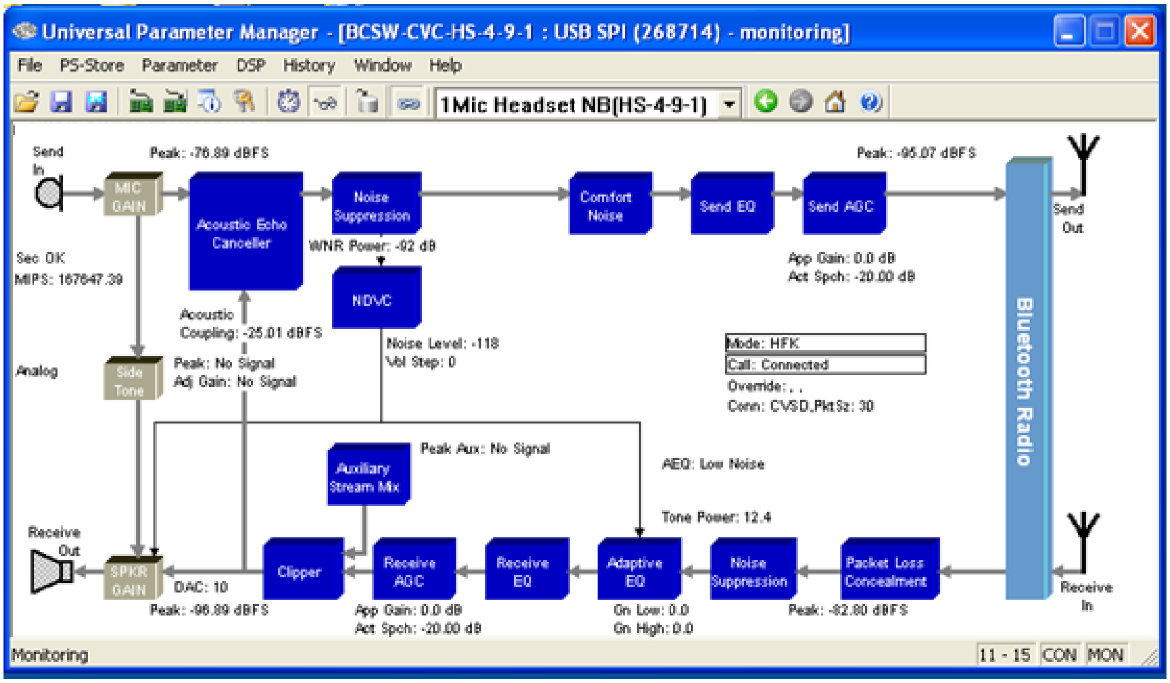



**Figure 2-5 Processing blocks on the Parameter Manager window, monitoring mode**

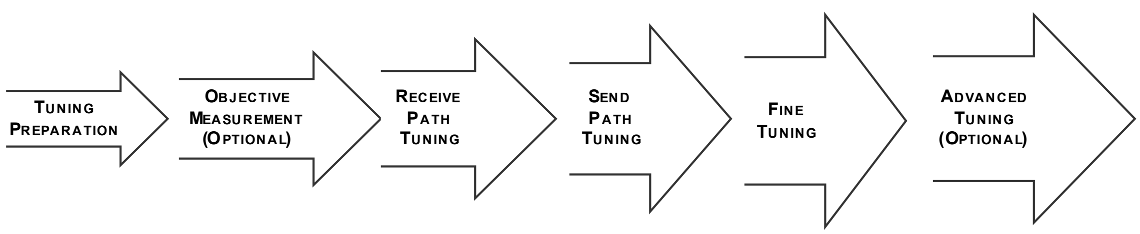



**Figure 2-6 Six cVc HS tuning stages**

**NOTE** Some stages may be unnecessary, if a design does not use a specific cVc feature block.

## 2.4 Tuning stages

The tuning stages are:

1. **Tuning Preparation:** Preparing the test environment and setting up the test equipment.
2. **Objective Measurement (optional):** Using a head and torso simulator (HATS) system to characterize frequency response, loudness rating and distortion characteristics of the device under test (DUT).

**Receive Path Tuning:** Focusing on tuning the Receive Path processing blocks of the cVc HS algorithm (PLC, Receive AGC, Speaker Gain and Noise Suppression).

**Send Path Tuning:** Focusing on tuning the Send path processing blocks of the cVc HS algorithm (Microphone, Send AGC, Noise Suppression, Acoustic Echo Canceller and Comfort Noise).

1. **Fine-Tuning:** Adjusting the processing blocks, as required (Receive AGC, Receive EQ, Clipper, Auxiliary Stream Mix, Send EQ, Send AGC and Send Noise Suppression).
2. **Advanced Tuning (optional):** Adding or tuning the advanced feature processing blocks.

To enhance the audio performance, enable and tune:

- Adaptive EQ with Frequency Expansion
- Noise dependent volume control (NDVC)
- Side Tone

**NOTE** Enabling these features slightly increases current consumption.
Use the *BCSW-cVc-HS-4-9-2 1-mic HS Parameter Manager User Guide* as a reference during the tuning process.
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.

## 2.5 Tuning flowchart

Figure 2-7 shows a suggested procedure for the tuning process. See Quick start guide and Advanced tuning for details about these steps.

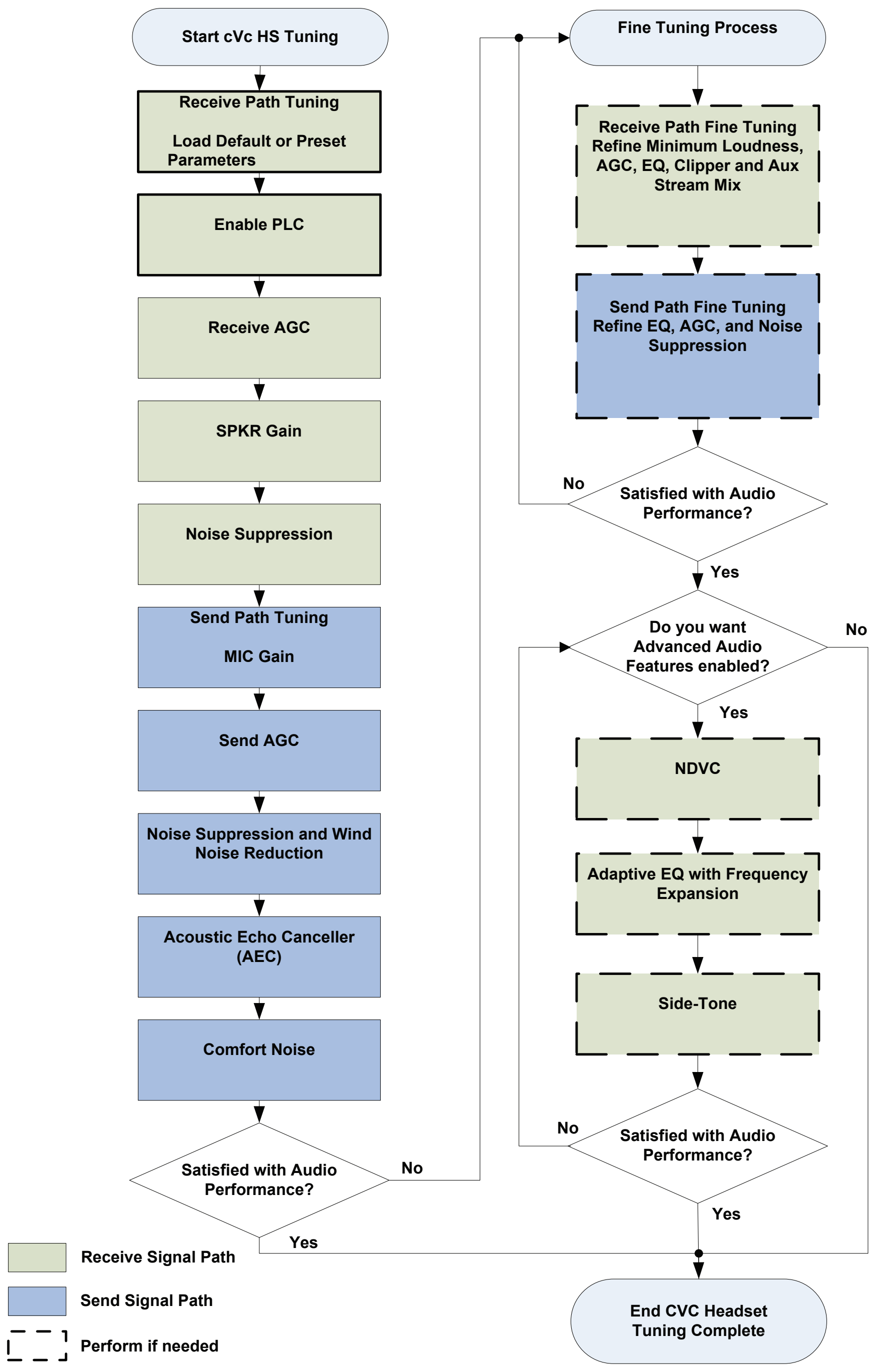



**Figure 2-7 Tuning flowchart**

# 3 Tuning preparation

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

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

## 3.1 Before tuning

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

## 3.2 Headsets

Before tuning headsets confirm that:

1. The headset's audio components and plastics are production-level or production-intent.
2. Any microphone wind noise reduction filters are in place.
3. The device includes SPI connectivity to a PC running the Parameter Manager software tool.
4. The SPI breakout cabling does not obstruct or interfere with the headset's microphone and receiver.
5. Before testing, the headset's battery is fully charged or has an external power source.
6. The headset is paired to a mobile telephone that supports the Bluetooth Headset profiles.
7. During testing, headset is positioned in its intended location.

You have made a short call with the default cVc HS parameters to check the performance before making any adjustments.

## 3.3 Phones models and network types

Understanding the general performance of the phones that the headset product will support is important because:

- Phone models and local networks vary and affect the sound quality, and also affect the headset 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 Synchronous Connection-Oriented Link (SCO) Audio level, QTIL recommends performing initial tuning using the mobile phone and network that has the lowest Receive SCO Audio level, and 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. You may need to make further fine adjustments 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 can influence telephone sound quality.

Use a Global System of Mobile Connections (GSM) mobile telephone as the primary tuning phone. After completing tuning, check the headset sound quality with a Code Division Multiple Access (CDMA) mobile phone for similar results. In CDMA-dominant countries, you can tune using only a CDMA phone.

While on a headset 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 headset microphone(s) and speakers.

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

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

## 3.4 Tuning environment

The tuning environment must be a controlled space that imitates or simulates the operating environment of the final product. For headsets, the tuning environment is a quiet environment with low reverberation, such as an anechoic chamber, and a noisy environment, such as a public space or car cabin.

During the tuning process, the near-end (headset) and far-end (landside) subjects must be acoustically isolated. Avoid the far-end subject's direct speech at the near-end microphone to mitigate unwanted echo and acoustic feedback.

**NOTE** To help minimize sound exposure that may be potentially damaging, use earplugs when testing in moderate to high sound pressure levels.

## 3.5 Level speech phrase

To obtain a proper signal for measuring speech levels in the processor, use 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** More precise tuning requires a laboratory-based test. QTIL can conduct 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 3-1 shows a recorded waveform of the near-end subject’s voice while speaking the test phrase “one two three four five” twice.

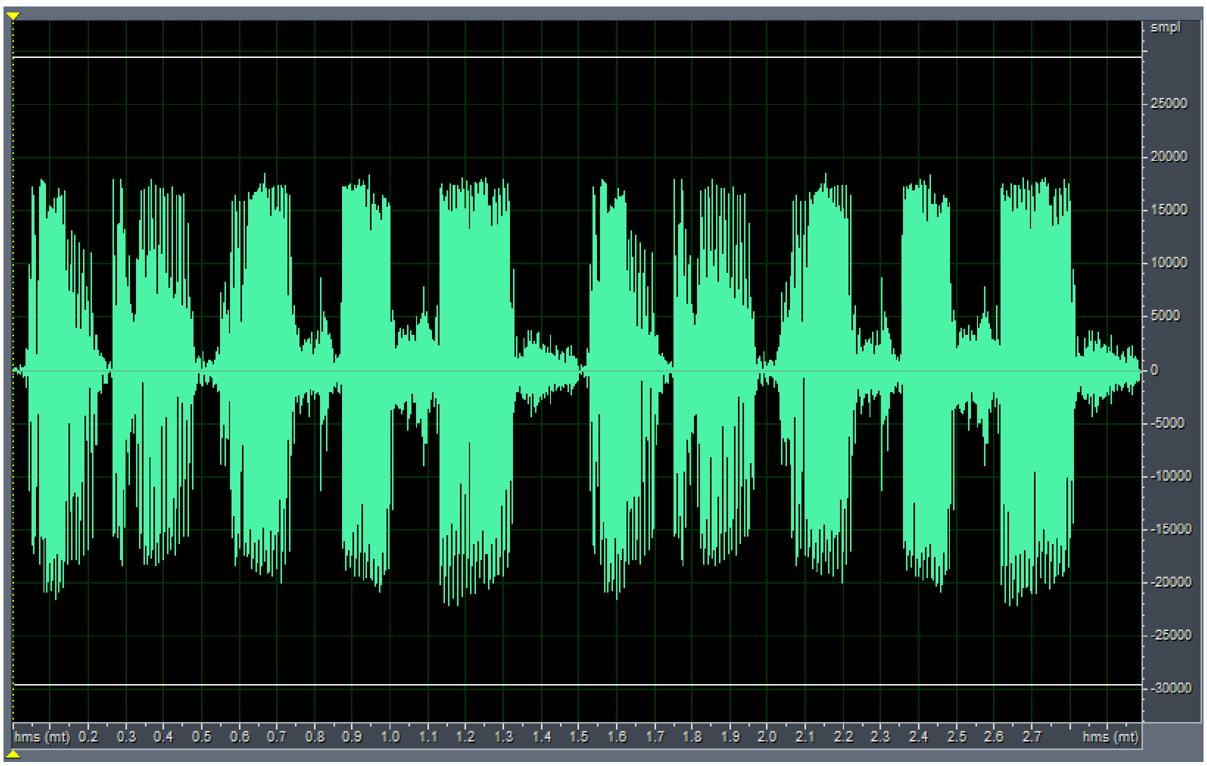



**Figure 3-1 Recorded waveform of the English phrase “one two three four five”**

# 4 Instrumentation

## 4.1 SPL meter

To measure Sound Pressure Level (SPL), use a high-quality digital sound meter, such as a Bruel & Kjaer (B&K) Type 2239 Sound Level Meter or equivalent.

**Table 4-1 Recommended meter settings**

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

## 4.2 HATS system

Use a HATS system capable of electro-acoustic measurements on telephones, per International Telecommunications Union-Telecommunication (ITU-T) recommendations. The HATS system must test send/receive frequency response, send/receive loudness ratings and receive-path Total Harmonic Distortion + Noise (THD+N).

# 5 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 headset microphone and receiver | ❑ |
| ■ Fully charged headset 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 section SPL meter | ❑ |
| ■ Calibrate the Sound Pressure Level Meter to a 1 kHz 9 4 dB re 20 μPa sine tone | ❑ |
| ■ Ensure all cables and power supplies are in proper working order | ❑ |
| ■ Parameter Manager tool is connected to the Headset system | ❑ |
| **Accessories** | |
| ■ Ear Plugs | ❑ |
| **Personal** | |
| ■ Far-end (landside) subject | ❑ |
| ■ Near-end (headset) subject | ❑ |
| **Documentation** | |
| ■ *cVc 8th Generation 1-mic Headset Parameter Manager User Guide* | ❑ |

# 6 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, establish an active SCO connection, with SPI communication, using the UFE in Monitoring Mode.

## 6.1 Set Receive Gain path

Listen to the receiver volume to obtain the maximum loudness, and adjust SPKR Gain to the required maximum receiver volume that the headset can support. As an alternative, measure RLR (Receive Loudness Rating).

The SPKR Gain highest level should not cause distortion. See SPKR Gain for details.

## 6.2 Set Send Gain path

Adjust the microphone gain (Left and Right) so speech (near-end) adjusts signal is approximately -15 dBfs as seen in the SND IN statistics of the front microphone

Set the headset receiver volume to the maximum by increasing the gain of the loudspeaker through manual volume controls if there is echo, decrease the MIC Gain level.

See MIC Gain for details on tuning the Mic Gain.

## 6.3 Acoustic Echo Canceller performance

If, at maximum volume, echo coupling is high and you hear some echo turn on the reuse primary filter. If you do not hear an echo, continue and set up the low volume mode.

For increased double talk, increase the Double Talk Aggressiveness to hear more double-talk. For less echo, decrease the Double Talk Aggressiveness to hear less echo.

See Acoustic Echo Canceller with half-duplex option for details on tuning the acoustic Echo Canceller.

**NOTE** It might take several attempts to find the best ratio to achieve optimal doubletalk and echo performance.

# 7 Tuning procedures

This section describes different tuning procedures.

## 7.1 Objective measurement

After completing the tuning preparation process, the headset device can be acoustically characterized using an objective telephone sound quality test system. This system typically 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): Tunes Receive EQ and Send EQ
- Send/Receive loudness rating (ITU-T P.79, ITU-T P.50): Tunes SPKR Gain, MIC Gain and Send Gain
- Send/Receive Distortion (THD/THD+N): Tunes the MIC Gain to a level limiting distortion and sets the SPKR Gain to a level-limiting distortion

## 7.2 Receive Path tuning

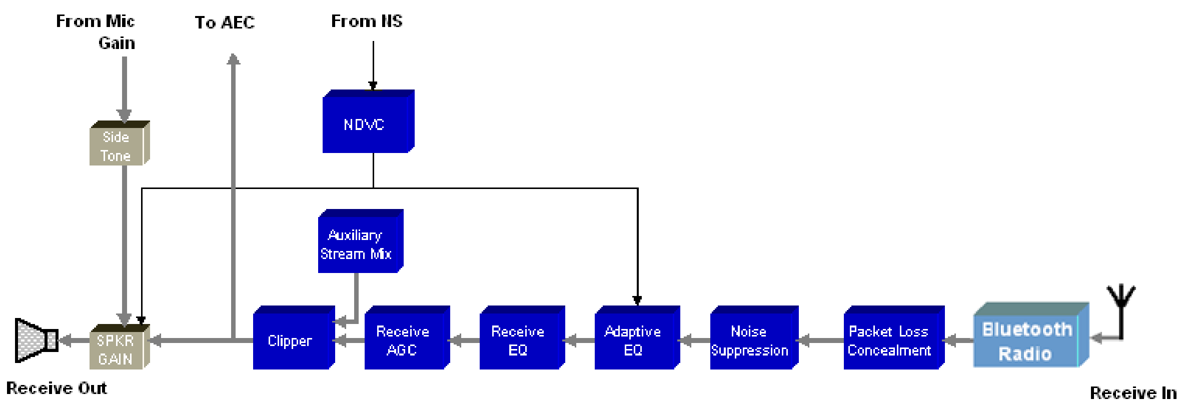



**Figure 7-1 Receive Path processing blocks**

**NOTE** Repeat all tuning measurements at least twice.

### 7.2.1 Load Preset parameters

When tuning the headset for the first time, begin tuning with the default parameters that QTIL provides with the cVc HS release. To load these defaults using the Parameter Manager application, select Use Default Parameters on the Parameters menu. When the defaults have been loaded, bypass the Adaptive EQ, Clipper and NDVC advanced processing blocks to simplify tuning.

If you have previously tuned the headset and saved the parameters, preload the saved parameters and continue with the tuning process. See the Parameter Manager integrated documentation for instructions on how to load saved parameters. Click the Documentation link on Parameter Manager home page and read the Managing Parameter Settings and PS Key section.

### 7.2.2 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. It uses 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, check Bypass Packet Loss Concealment.

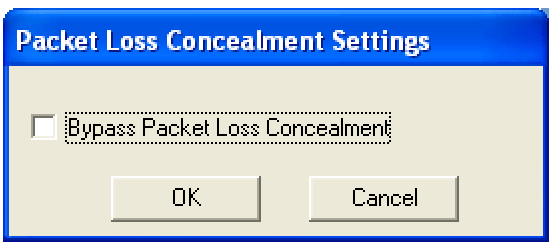



**Figure 7-2 Packet Loss Concealment Settings window**

### 7.2.3 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 (RMS) and does not require additional tuning. If you require additional fine tuning, see Receive AGC.

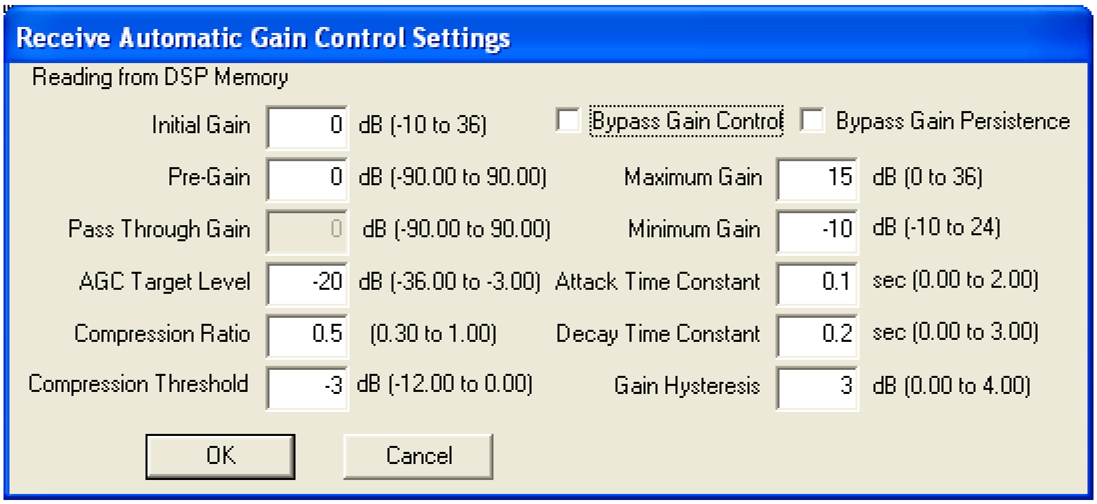



**Figure 7-3 Packet Loss Concealment Settings window**

**NOTE** You can bypass the Receive AGC by selecting Bypass Receive AGC. QTIL does not recommend bypassing the Receive AGC.

### 7.2.4 SPKR Gain

Tuning the speaker involves determining the maximum receiver volume that the headset supports. As the headset's 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 to help reduce echo and distortion.

To tune the SPKR GAIN:

1. Initiate a headset 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.
    a. Click the SPKR GAIN block.
    b. Select Temporarily override DAC.
    c. Adjust the SPKR Gain to the highest level that does not cause distortion and passes objective and subjective loudness judgment.

5. If there is distortion:
   a. The near-end subject lowers the SPKR Gain or enables and adjusts the Clipper.
   b. Follow the Clipper setup described in Clipper.
   c. Optional: The near-end subject adjusts the Boost located in the clipper so that the required receiver loudness is maintained.
6. In the CSR86xx Configuration Tool, select the volume control tab to configure the required number of volume steps and set the maximum volume.

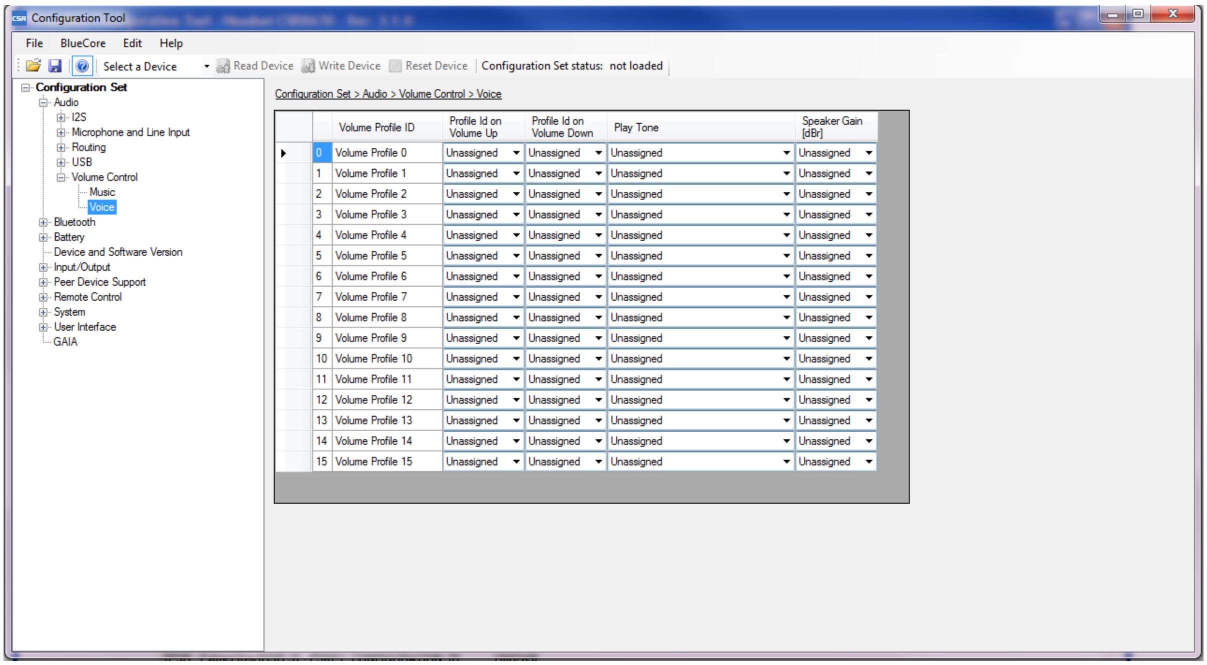



**Figure 7-4 Headset Configuration tool, audio gains**

### 7.2.5 Receive Noise Suppression

The HFK Aggressiveness parameter is a primary tuning parameter that 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. The receive signal is processed by the cellular network and transmitted over Bluetooth. To avoid over-processing the voice, set the aggressiveness conservatively.

To tune for noise suppression:

1. Initiate a headset 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. Customize HFK Agressiveness:
   a. Increase the HFK Aggressiveness for more noise suppression. This may affect voice quality.
   b. Decrease the HFK Aggressiveness for less noise suppression.

**NOTE** The far-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 checked (enabled) by default. This does not affect the noise suppression but provides improved speech quality. If voice quality is not critical, un-check this option to reduce the processing load. However, because high quality mode only consumes approximately 1 MIPS, QTIL recommends that you leave it enabled.

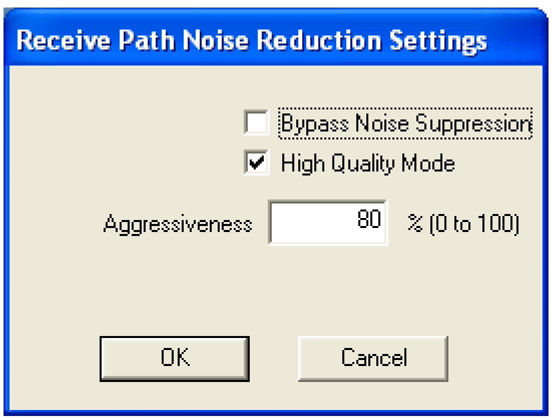



**Figure 7-5 Receive Path Noise Suppression Settings window**

## 7.3 Send Path tuning

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

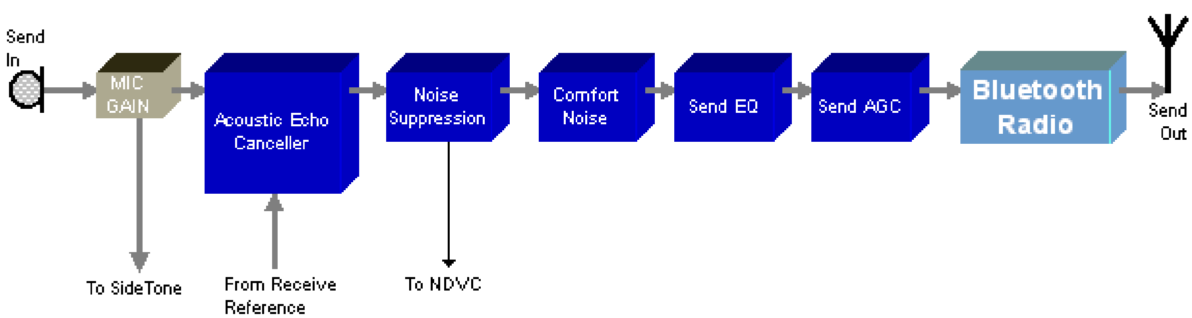



**Figure 7-6 Send Path Processing block diagram**

### 7.3.1 MIC Gain

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 headset call.
2. The near-end subject adjusts the headset receiver volume to the maximum, by increasing the gain of the loudspeaker through manual volume controls.
3. The far-end subject speaks the speech level phrase while the near-end subject monitors the MIC GAIN Peak statistics.
4. 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. 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 8.
6. The far-end subject speaks the speech level phrase while the near-end subject monitors the MIC GAIN Peak statistics.
7. 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.
8. The near-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 -15 dBFS. End MIC GAIN tuning for near-end speech is larger than far-end speech (echo).
10. The far-end subject speaks short bursts of speech (for example, “one”, “two”, “hello”, “ok” or “check”) and checks for echo at the far-end.
11. The near-end adjusts the MIC Gain so that the Peak Mic statistic reads no more than -15 dBFS.
12. If there is echo present at the far-end, the near-end subject decreases the MIC Gain level.
13. Re-check for far-end echo.

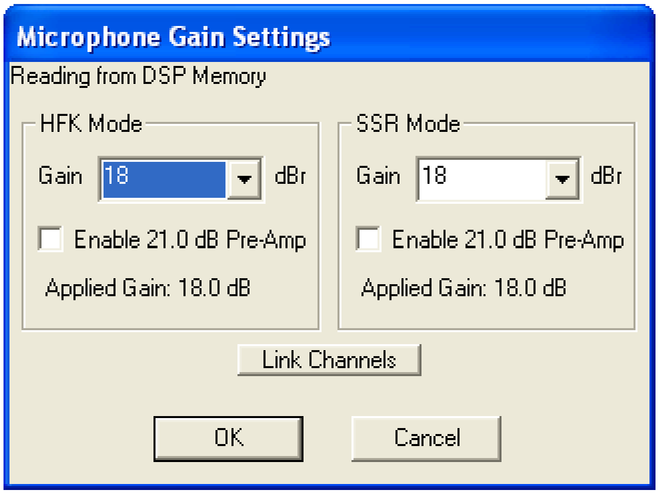



**Figure 7-7 Microphone Gain Settings window**

### 7.3.2 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 -20 dB (RMS) and needs no additional tuning. See Figure 7-8.

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

To Tune the AGC Target Level (for headsets without a close-coupled microphone and speaker):

1. Initiate a headset call.
2. The near-end subject places the primary headset microphone at the closest specified operating distance from the near-end subject.
3. 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.
4. The near-end subject adjusts the AGC Target Level to achieve the required listening level at the far-end.
5. Monitor the **Send Out Peak** statistic to ensure the speech is not clipping, to avoid saturation. Typically, the Send AGC should not be raised above -10 dB target full scale (to allow for overshoot), processing in the event of saturation or clipping.
6. Limit the Maximum Gain that can be applied to the signal, if necessary.
7. Specify the Compression Ratio to suit the needs of the application, if necessary.
8. Ensure that the far-end subject never hears clipped or saturated speech.

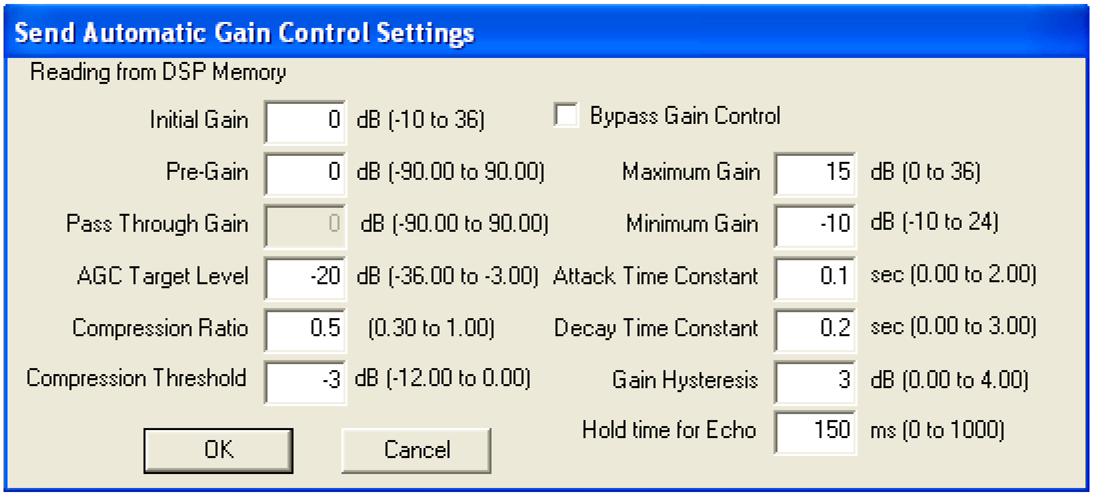



**Figure 7-8 Send Automatic Gain Control Settings window**

## 7.3.3 Send Noise Suppression (includes wind noise reduction)

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

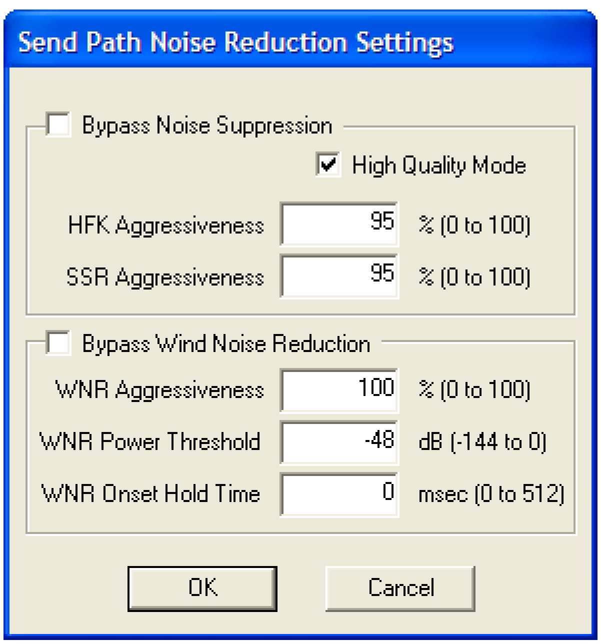



**Figure 7-9 Send Path Noise Suppression Settings window**

### 7.3.3.1 Tuning for Noise Suppression

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

To tune for noise suppression:

1. Initiate a headset 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. Customize HFK Aggressiveness.
   a. Increase the HFK Aggressiveness for more noise suppression (at the cost of voice quality).
   b. 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 can better judge speech intelligibility in the presence of noise.

#### 7.3.3.2 Tuning for Wind Noise Reduction

The WNR Aggressiveness is the primary tuning parameter and 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. The WNR Onset Hold Time parameter determines how long to wait once wind is detected before WNR is applied.

To tune for wind noise reduction:

1. Ensure that the near-end subject stops and the background noise is quiet.
2. Set the WNR Aggressiveness to 100%.
3. Set both WNR Power Threshold and WNR Onset Hold Time to 0.
4. The near-end subject adds wind using a controlled source (such as a fan), attempting to maintain a constant moderate wind speed of approximately 4 mph blowing directly onto the front face of the microphone.
5. 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 WNR is on.
6. Stopping the wind, the near-end subject speaks a test phrase or a normal conversational phrase, continually.
7. Record or subjectively evaluate the speech quality. If the speech quality (mainly the onset 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.
8. The near-end subject combines both the moderate wind noise and speech phrases.
9. Decrease the WNR Aggressiveness until the required level of WNR is achieved, or leave at maximum (100%).

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

### 7.3.4 Acoustic Echo Canceller with half-duplex option

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

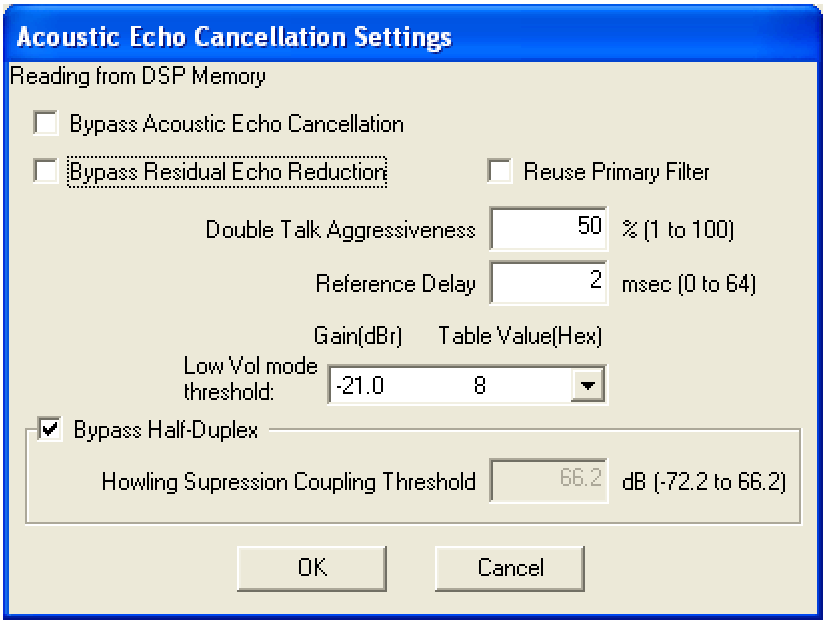



**Figure 7-10 Acoustic Echo Cancellation Settings window**

#### 7.3.4.1 Acoustic Echo Cancellation settings fields description

The Acoustic Echo Cancellation settings fields are:

- Bypass Acoustic Echo Cancellation: Check this option when echo cancellation is not required. If bypassed, see Half-Duplex settings fields description.
- Bypass Residual Echo Reduction: The Bypass Residual Echo Reduction option enables or disables additional echo cancellation built into the AEC. The Residual Echo Reduction reduces the subtle non-linearities that may exist after the primary adaptive filter. It is enabled by default. For best echo cancellation performance, enable the Residual Echo Reduction feature.
- Reuse Primary Filter: Under normal conditions, the filter converges to the required level so that echo is reduced. Select this option when the AEC filter convergence must be as close as possible to the optimal convergence value. Check this option when the echo coupling is high. Selecting this option increases the DSP processing load.
- Reference Delay: This feature 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.

- Double Talk Aggressiveness: This feature adjusts the amount of double-talk signal that is heard at the far-end.
  - Decrease the Double Talk Aggressiveness to hear less double-talk signal at the far-end or increase the amount of echo cancellation.
  - Increase the Double Talk Aggressiveness to hear more double-talk signal or decrease the amount of echo cancellation.
- Low Volume Mode Threshold: This feature saves headset battery power by turning off the Acoustic Echo Canceller and Comfort Noise generator when the Speaker Gain is set below a certain threshold. Determine the threshold by lowering the volume until no perceived echo or an acceptable echo is heard.

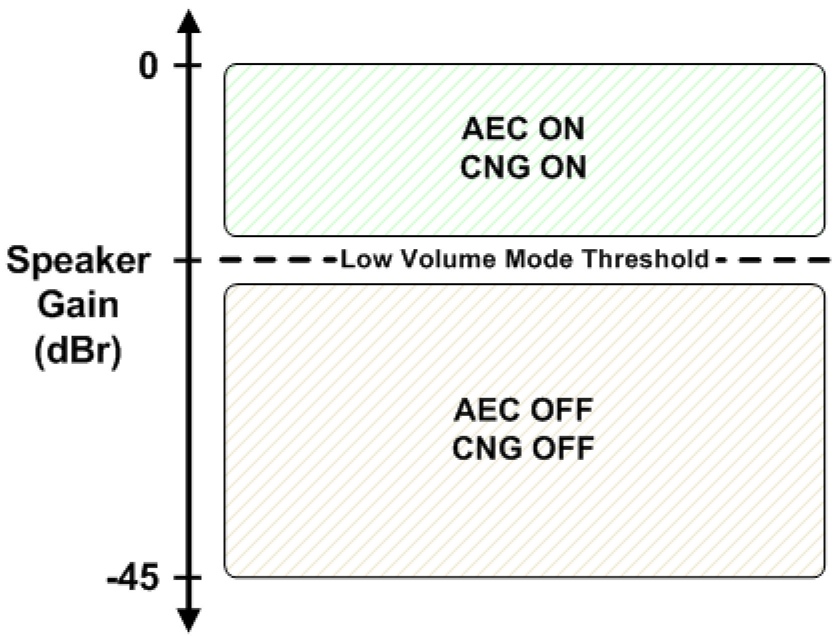



**Figure 7-11 Low volume mode threshold operation**

### 7.3.4.2 Half-Duplex settings fields description

Half-duplex is required if the primary AEC cannot reduce the echo to an acceptable level. This condition can result if the acoustic coupling is excessive or high loudspeaker distortion, such as a car kit. Using the half-duplex feature may result in reduced double talk performance.

Check the Bypass Half-Duplex option when half-duplex is not required.

The Howling Suppression Coupling Threshold sets the threshold to invoke full band attenuation in the presence of far end speech in an effort to eliminate echo. The threshold is based on the acoustic coupling statistic displayed during monitoring mode.

To tune the Acoustic Echo Canceller:

1. Initiate a headset call and set the loudspeaker volume to maximum.
2. The near-end subject sets the Low Vol Mode Threshold to minimum (-45 dBr, Table Value 0) and uncheck the Reuse Primary Filter.

3. The far-end subject speaks short bursts of speech (for example, “one”, “two”, “hello”, “ok”, or “check”) and checks for echo at the far-end.
4. If echo is heard, check the Reuse Primary Filter and repeat step 3. If no echo is heard, continue and setup the low volume mode.

To fine-tune the Acoustic Echo Canceller EC set the low volume mode threshold:

1. Initiate a headset call.
2. The near-end subject selects Bypass Acoustic Echo Cancellation.
3. The far-end subject speaks short bursts of speech (for example, “one”, “two”, “hello”, “ok”, or “check”) and checks for echo at the far-end.
4. Starting at the maximum receiver volume (gain), adjust the near-end receiver volume downward. At each volume adjustment, repeat step 3. Continue until the echo at the far-end user cannot be heard or the echo is low enough not to require the echo canceller software. Record this volume setting.
5. The near-end subject sets the Low Vol mode threshold to the speaker volume (gain) level determined in step 4.
6. The near-end subject removes the checkmark in the Bypass Acoustic Echo Cancellation check box.
7. At each speaker gain level, the far-end subject checks for echo at the far-end.

**NOTE** To adjust the volume use the Speaker Gain Setting block (check Temporary override DAC) and choose a volume from the Gain Select drop list. When complete, uncheck the Temporary override DAC option.
The Bypass Residual Echo Reduction feature enables or disables additional echo cancellation built into the AEC. The echo reduction feature is enabled by default. QTIL recommends that this feature remain enabled.

### 7.3.5 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.

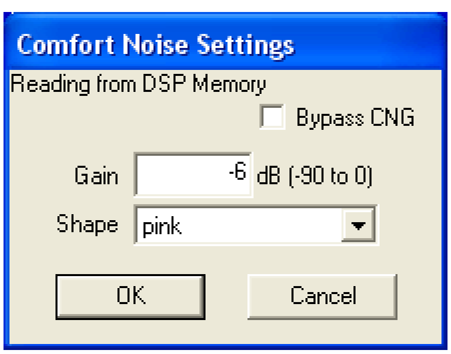



**Figure 7-12 Comfort Noise Settings window**

To tune the Comfort Noise Generator:

1. Initiate a headset 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”, or “C”) with a one-second pause between each number 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. Use Shape to select the weighting of the comfort noise spectrum

# 8 Fine-tuning

When the cVc headset receive and send paths are tuned, you may need to make minor parameter changes to reach a good performance level. Some products have unique acoustic designs or require special headset sound quality requirements for the product. This section describes these additional tuning instructions.

## 8.1 Receive Path fine-tuning

This section describes fine adjustments that you can make to the cVc HS receive path.

### 8.1.1 Setting minimum Speaker Gain loudness

SPKR Gain describes how to set the speaker gain tuning to obtain the maximum loudness. Setting the Minimum Loudness for the loudspeaker gain is similar to setting the Maximum Loudness.

To fine-tune the Minimum Loudness Level:

1. Initiate a headset 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 Mono Headset Configuration Tool, and choose the Audio Gains tab to configure the required number and minimum number of volume steps.

### 8.1.2 Receive AGC

To tune the Receive AGC:

1. Adjust the AGC Target Level to the required value. The default is -20 dB RMS, 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 will not fall below the Minimum Gain.

3. Adjust the Maximum Gain, which sets the high threshold level for the gain factor. The gain factor will not exceed 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 Compression Threshold. The gain factor follows the compression curve above the Compression Threshold, while the slope of gain curve below the Compression Threshold is unity.

### 8.1.3 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 126 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 the *Headset BCSW-cVc-HS-4*-9-*2 Parameter Manager User Guide* for details.

### 8.1.4 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 limit this distortion. Select a Clip Point 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 headset call.
2. Adjust the headset volume to the maximum.
3. The far-end subject speaks the level speech phrase.
4. The near-end subject listens for distortion in the hands-free 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.

Optional Steps:

1. If the Clipper is enabled, the near-end subject adjusts 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 adjusts the Boost Clip Limit enforcing the maximum digital limit allowed in the path prior to the DAC.

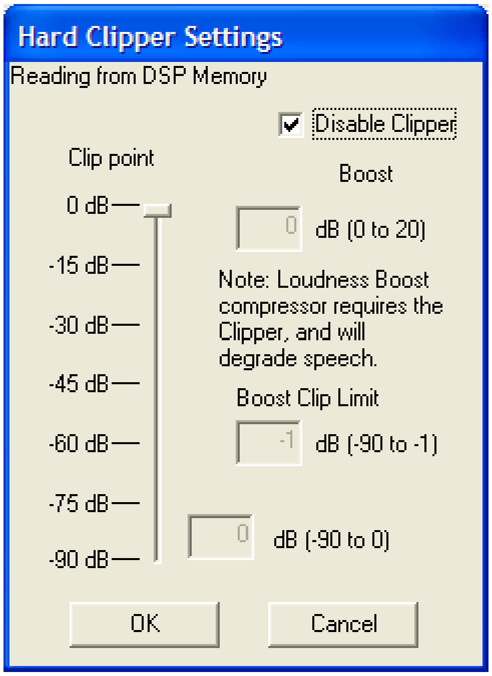



**Figure 8-1 Hard Clipper Settings window**

### 8.1.5 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.

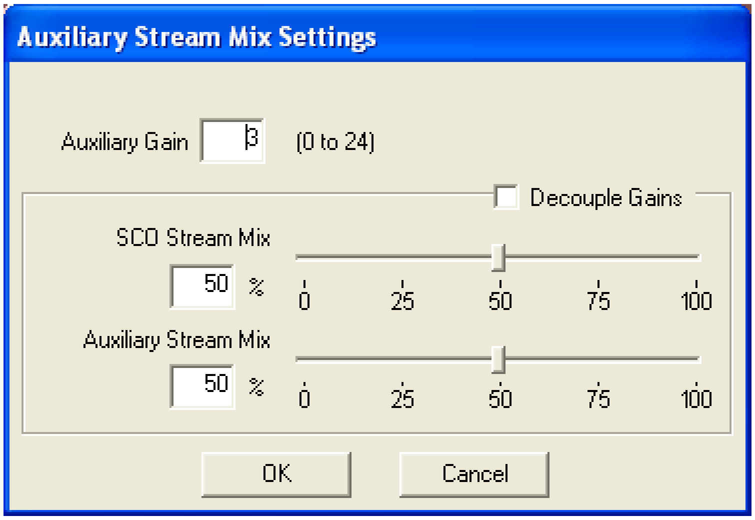



**Figure 8-2 Auxiliary stream mix Settings window**

## 8.2 Send Path fine-tuning

This section describes how to make fine adjustments to the cVc Headset send path.

### 8.2.1 Send EQ

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

If required, the Send EQ performs some frequency shaping to fit a required response curve. The graphical user interface enables the Send EQ parameters to be graphically selected. See the *Headset BCSW-cVc-HS-4-8-1 Parameter Manager User Guide* for details.

**NOTE** To avoid saturation and distortion, minimize Gain (unity) through the EQ.

### 8.2.2 Send AGC

To fine-tune the Send AGC:

1. Adjust the AGC Target Level to the required value. The default is -20 dB RMS, 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 will not fall below the Minimum Gain.

3. Adjust the Maximum Gain, which sets the high threshold level for the gain factor. The gain factor will 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 Compression Threshold. The gain factor follows the compression curve above the target level. The slope of gain curve below the target level is unity.

# 9 Advanced tuning

This section describes advanced tuning features.

## 9.1 Advanced tuning features

For headset products that offer adequate fidelity, enable these advanced cVc features:

- Noise Dependent Volume Control (NDVC)
- Adaptive EQ (AEQ)
- Side-tone

### 9.1.1 NDVC

Tune NDVC after the MIC Gain. See MIC Gain for instructions on tuning 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 determined in SPKR Gain.

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 headset 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 Minimum 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 55 mph).
6. Monitor the Noise Level statistic and place this value in the Maximum Noise Level field.
7. 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.

8. Set the Hysteresis to a value between zero and one. Higher values reduce the NDVC sensitivity when reacting to changes in the background noise. Lower values increase the sensitivity. The default value is 0.75.
9. 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. Lower values cause a quicker reaction.

**NOTE** 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. See SPKR Gain for details.
The Total SPKR Gain is 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.

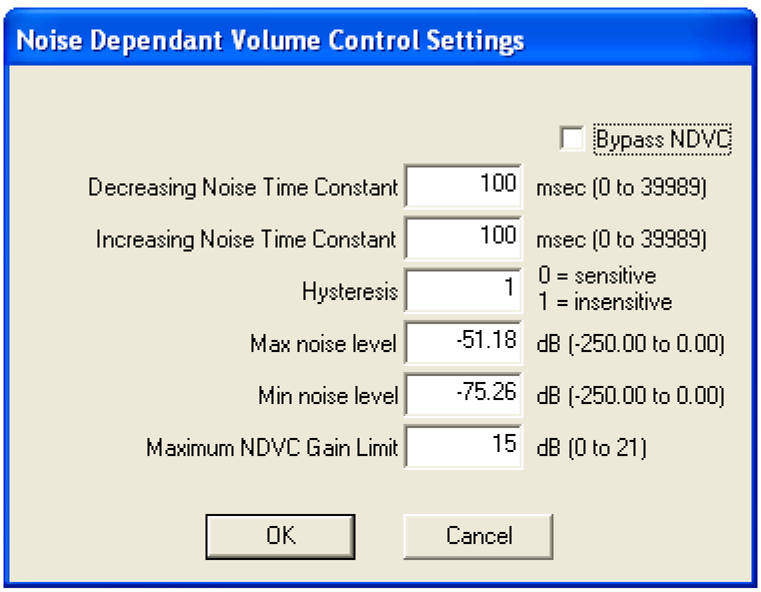



**Figure 9-1 Noise Dependent Volume Control Settings window**

**NOTE** Figure 9-1 shows the default settings. Adjust Max noise level and Min noise level for your specific headset.

### 9.1.2 AEQ

The three systems available for the CSR86 are:

- 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).

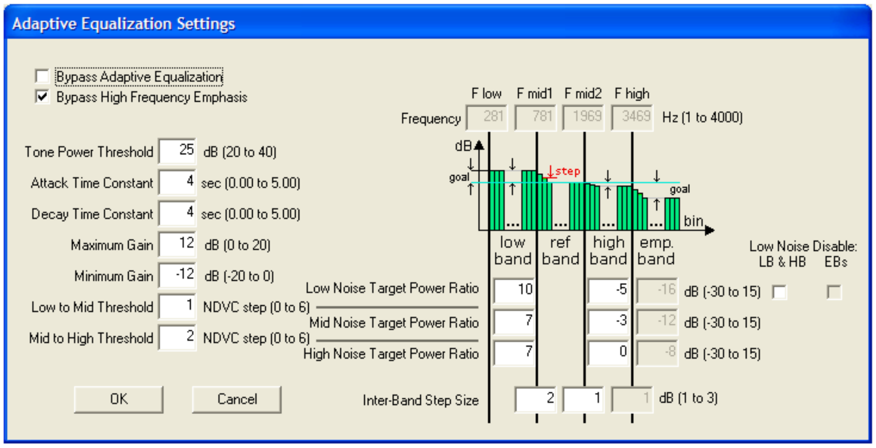
When Adaptive Equalization block is enabled, it alters the spectral shape of the receive path signal while maintaining the overall power level. This improves the intelligibility of the receive path voice signal in the presence of near end noise.

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. Increasing the frequency components that contribute to the consonants while in the presence of noise can improve the intelligibility.

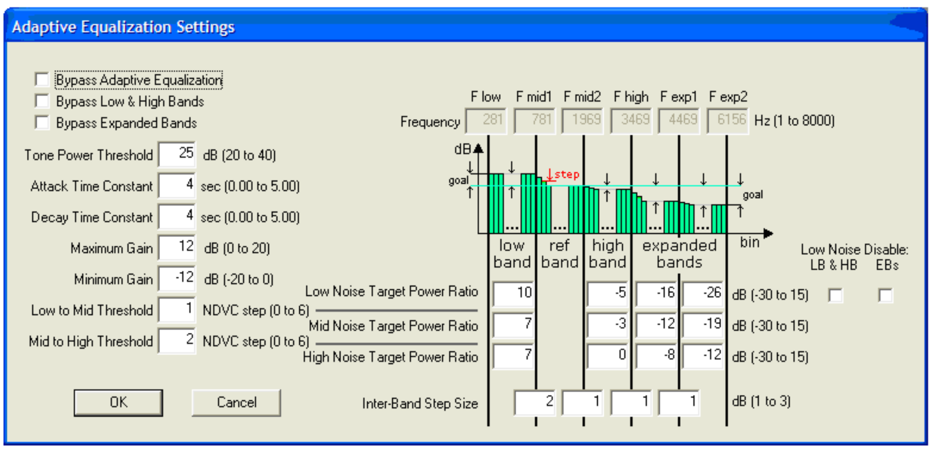
To maintain a consistent amplitude level, the Adaptive Equalization block adaptively increases the high frequencies relative to the middle frequencies while reducing low frequencies accordingly.

The adaptive equalizer also has the capability to compensate for variations in voice transmission channels, which include far-end devices and telecommunication channels.

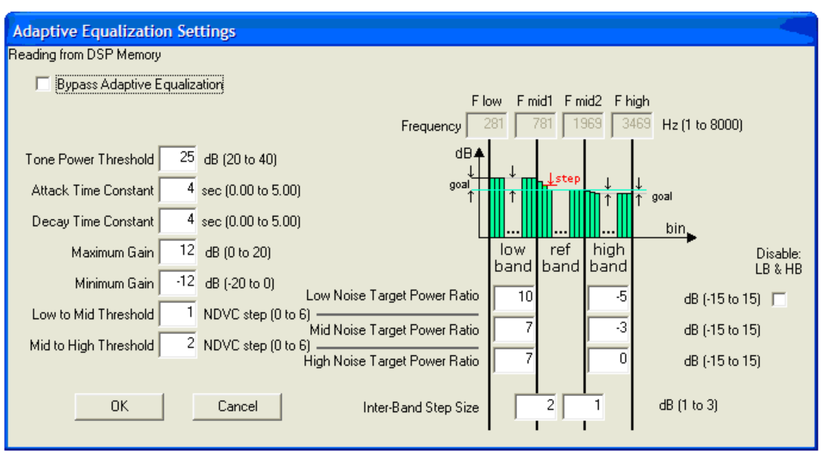
**NOTE** For the headset to benefit from this feature, the loudspeaker must deliver adequate fidelity to the user's ear. Good examples are headsets fitted with a gel ear bud that seals the ear canal. An open air, hard plastic speaker headset is not a good design for use with the AEQ.




**Narrow Band AEQ plus High Frequency Enhancement**




**Narrow Band AEQ plus Frequency Expansion**




**Wide Band AEQ**

**Figure 9-2 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. When the Mid to High Threshold is crossed, the High Noise Target Power Ratio curve is applied.

**NOTE** To bypass application of the Adaptive EQ in quiet situations, check 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. Place the step transitions evenly across the range of NDVC steps available (in this case 0 to 5).

The NDVC at Vol Step: 2 triggers the Low to Mid Threshold switching from the Low to Mid Noise Target Power Ratio curve.

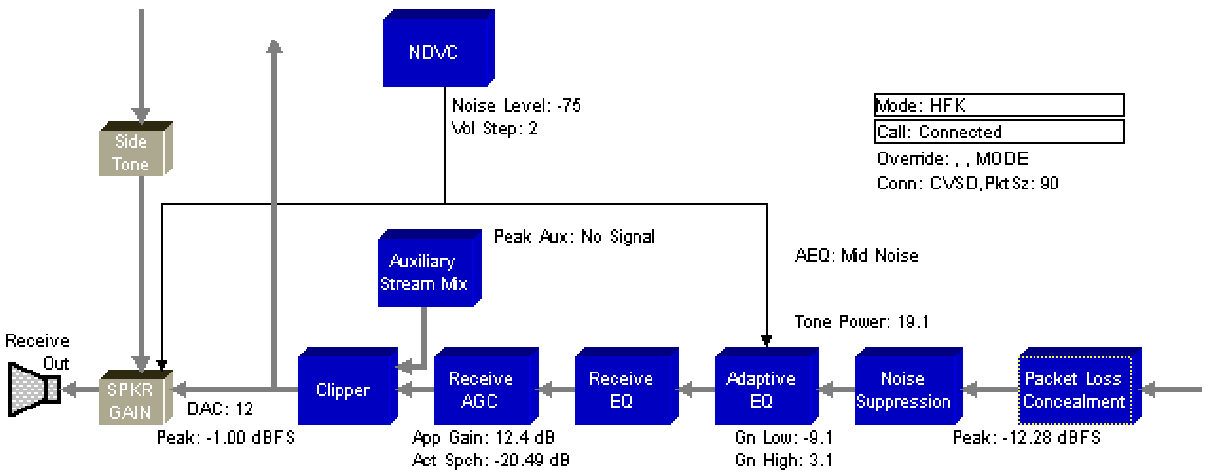



**Figure 9-3 Adaptive Equalization switching to mid noise tier**

The NDVC at Vol Step: 3 triggers the Mid to High Threshold switching from the Medium to the High Noise Target Power Ratio curve.

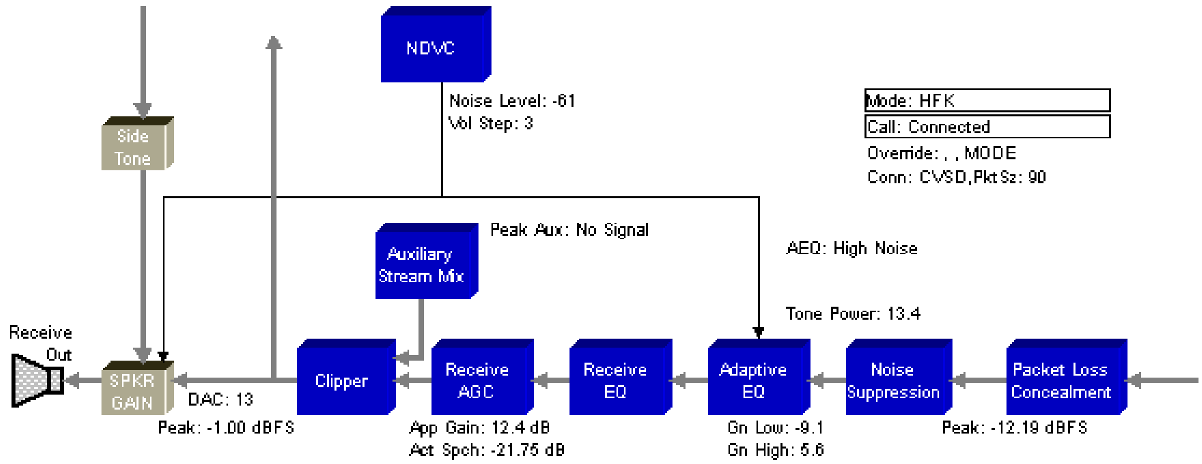



**Figure 9-4 Adaptive Equalization switching to high noise tier**

To tune the AEQ:

1. Initiate a headset call.
2. To isolate the Adaptive Equalization, disable any high frequency expansion or enhancement by checking the Bypass Expanded Bands or Bypass High Frequency Emphasis options (bypassed throughout the Adaptive Equalization tuning).
3. Disable the Adaptive Equalization by checking the Bypass Adaptive Equalization option.
4. The near-end user, while wearing the headset, listens to the original receive speech in a low noise environment (as confirmed by the AEQ Noise Level statistic).
5. After listening to approximately 30 seconds of receive speech, uncheck the Bypass Adaptive Equalization box and listen to the receive speech (again, for approximately 30 seconds).
6. Adjust the spectral shape of the low and high bands by raising or lowering the Low Noise Target Power Ratio parameters under the appropriate low or high band column(s).
7. Disable the low and high bands for low noise (if necessary) by checking the Low Noise Disable LB & HB box.
8. Disable the Adaptive Equalization by checking the Bypass Adaptive Equalization option.
9. The near-end user, while wearing the headset, listens to the original receive speech in a medium noise environment (as confirmed by the AEQ Noise Level statistic).
10. After listening to approximately 30 seconds of receive speech, uncheck 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 or lowering the Mid Noise Target Power Ratio parameters under the appropriate low or high band column(s).
12. Disable the Adaptive Equalization by checking the Bypass Adaptive Equalization option.
13. The near-end user, while wearing the headset, listens to the original receive speech in a high noise environment (as confirmed by the AEQ Noise Level statistic).

14. After listening to approximately 30 seconds of receive speech, uncheck 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 or lowering the High Noise Target Power Ratio parameters under the appropriate low or high band column(s).

### 9.1.3 AEQ with high frequency emphasis or expansion

#### 9.1.3.1 Narrow band plus high frequency emphasis

To turn on High Frequency Emphasis, un-check 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 in the column emp. band 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.

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

#### 9.1.3.2 Narrow band plus frequency expansion

When running a system including Frequency Expansion, the AEQ component operates as described in AEQ.

To turn on Frequency Expansion, un-check 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. Controls 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 headset call.
2. Disable the High Frequency Emphasis by checking the Bypass High Frequency Emphasis option.
3. The near-end user, while wearing the headset, listens to the original receive speech in a low noise environment (as confirmed by the AEQ Noise Level statistic).

4. After listening to approximately 30 seconds of receive speech, uncheck 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 or lowering the Low Noise Target Power Ratio parameters under the appropriate emp. band column.
6. Disable the Emphasis Band for low noise (if necessary) by checking the Low Noise Disable EBs option.
7. Disable the High Frequency Emphasis by checking the Bypass High Frequency Emphasis option.
8. The near-end user, while wearing the headset, listens to the original receive speech in a medium noise environment (as confirmed by the AEQ Noise Level statistic).
9. After listening to approximately 30 seconds of receive speech, uncheck 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 or lowering the Mid Noise Target Power Ratio parameters under the appropriate emp. band column.
11. Disable the High Frequency Emphasis by checking the Bypass High Frequency Emphasis option.
12. The near-end user, while wearing the headset, listens to the original receive speech in a high noise environment (as confirmed by the AEQ Noise Level statistic).
13. After listening to approximately 30 seconds of receive speech, uncheck 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 or 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 headset call.
2. Disable the Frequency Expansion by checking the Bypass Expanded Bands option.
3. The near-end user, while wearing the headset, listens to the original receive speech in a low noise environment (as confirmed by the AEQ Noise Level statistic).
4. After listening to approximately 30 seconds of receive speech, uncheck 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 or lowering the Low Noise Target Power Ratio parameters under the appropriate expanded bands column(s).
6. Disable the expanded bands for low noise (if necessary) by checking the Low Noise Disable EBs option.
7. Disable the Frequency Expansion by checking the Bypass Expanded Bands option.
8. The near-end user, while wearing the headset, listens to the original receive speech in a medium noise environment (as confirmed by the AEQ Noise Level statistic).
9. After listening to approximately 30 seconds of receive speech, uncheck 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 or lowering the Mid Noise Target Power Ratio parameters under the appropriate expanded bands column(s).
11. Disable the Frequency Expansion by checking the Bypass Expanded Bands option.

12. The near-end user, while wearing the headset, listens to the original receive speech in a high noise environment (as confirmed by the AEQ Noise Level statistic).
13. After listening to approximately 30 seconds of receive speech, uncheck 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 or lowering the High Noise Target Power Ratio parameters under the appropriate expanded bands column(s).

### 9.1.4 Side Tone

Side Tone is the signal picked up by the headset's microphone and reproduced at the headset's receiver.

The cVc HS Side Tone block has a Sidetone Gain setting that increases or decreases the amount of side tone heard at the receiver. cVc maintains the same side tone level as the headset user adjusts the volume up or down setting by the Gain Adjustment Limit. The Sidetone Clip can be set to limit side tone gain that could cause DAC saturation. A high pass filter can be controlled by the Filter State Off / On radio control, and response set by the Filter Corner Frequency and Filter Q fields.

To tune the Side Tone:

1. Initiate a headset call.
2. Set the mobile phone volume to the maximum.
3. Set the headset's volume to the maximum.
4. Uncheck the Bypass Sidetone and set the Filter State to Off.
5. The near-end subject speaks a test phrase or a normal conversational phrase, continually.
6. The near-end subject listens to the side tone signal during single-talk and double-talk conditions at the far-end.
7. The near-end subject adjusts the Sidetone Gain parameter to achieve the required amount of side tone gain.
8. Would the near-end like to have the side tone increase with changes? If no, skip this field. If yes, adjust the Gain Adjustment Limit parameter to achieve the required amount of side tone gain based on volume changes.
9. Do you want to filter the side tone (for example, to reduce road noise)? If no, skip this field. If yes, set the Filter State to On.

The filter comprises three cascaded bi-quad stages, with each stage user configurable. The GUI enables the side tone filter EQ parameters to be graphically selected. See the *Headset ADK cVc BCSW-cVc-HS-4-9-2 Parameter Manager Users Guide* for more details about the EQ.

1. Figure 9-5 shows a recommended band pass configuration, where low frequency (road noises) and high frequency (hisses) are rolled-off. The filter helps the side tone from saturating by reducing unwanted noise while maintaining adequate speech.
2. Is the RCV out Peak statistic saturating with near-end speech at maximum volume? If no, skip this field. Otherwise adjust the Sidetone Clip until the saturation is avoided.

**NOTE** Too little side tone may cause the headset to appear to work improperly. Too much side tone can cause discomfort to the headset user, including a howling or feedback condition.

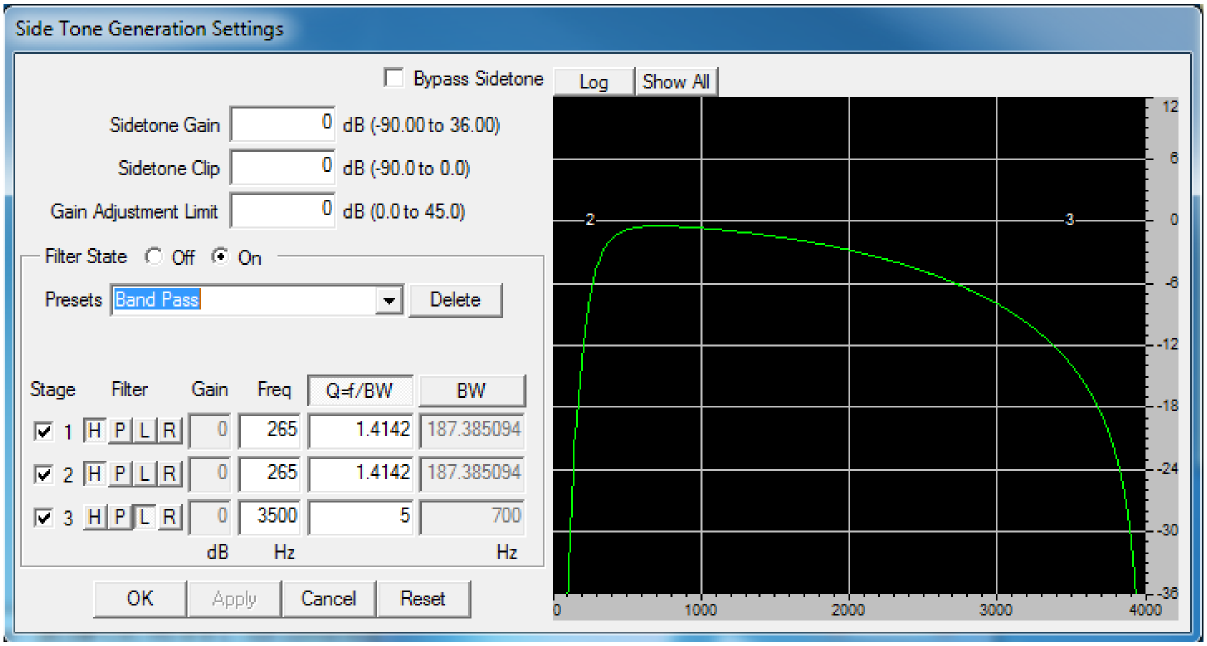



**Figure 9-5 Side Tone Generation Settings window**

## 9.2 Simple Speech Recognition microphone gain tuning

Simple Speech recognition (SSR) is a small vocabulary, “Yes” and “No” speech recognition system that answers or rejects an incoming call on a Bluetooth headset. The optimal target level of speech for the SSR engine is no higher than -25 dBFS. The SSR MIC Gain tuning is done while in 1MIC Headset HFK mode, using the HFK Mode Microphone Gain. The MIC Gain settings are then placed in the SSR Mode Microphone Gain settings and downloaded. Figure 9-6 shows the Microphone Gain Settings window.

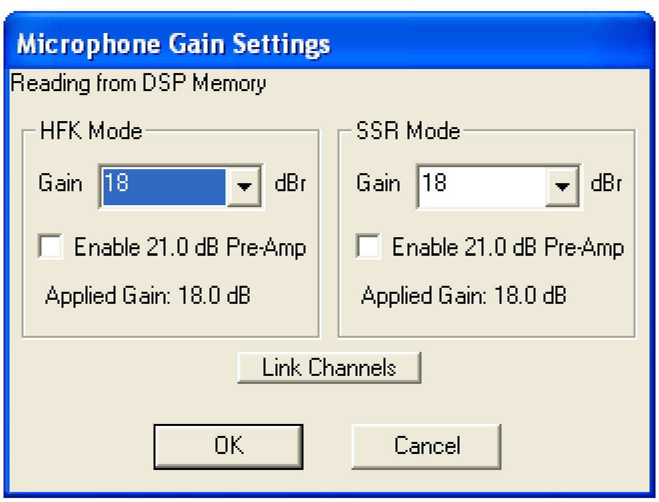



**Figure 9-6 SSR mode Microphone Gain Settings window**

To tune the MIC Gain for SSR:

1. Initiate a headset call.
2. 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.
3. The near-end adjusts the HFK Mode MIC Gain so that the Peak MIC statistic reads no more than -25 dBFS.
4. Once proper gain settings are determined for SSR, this gain value is placed in the SSR Mode MIC Gain settings and the HFK Mode MIC gain settings return to the original tuned value for HFK mode.
5. Download Parameters.

**NOTE** The applied MIC Gain does not change when switching from an HFK value to a SSR value during an Override Mode change using the UFE during an active Headset call.

# Document references

| Document | Reference |
|---|---|
| *BCSW-CVC-HS-4-9-2 1M-HS Parameter Manager User Guide* | 80-CT420-1/CS-00309816-UG |

# Terms and definitions

| | |
|---|---|
| ADC | Analog 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 |
| BlueCore | Group term for QTIL's range of Bluetooth wireless technology chips. |
| Qualcomm® BlueLab™ | QTIL's development toolset for building applications to run in the firmware's VM. |
| Bluetooth | Set of technologies providing audio and data transfer over short-range radio connections. |
| BCSW | BlueCore Software |
| CDMA | Code Division Multiple Access |
| CNG | Comfort Noise |
| 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 |
| EQ | Equalizer |
| ERLE | Echo Return Loss Enhancement |
| GSM | Global System of Mobile Communications |
| GUI | Graphical User Interface |
| HATS | Head and Torso Simulator |
| HFK | Headset Mode |
| HS | Headset |
| HSP | Headset Profile |
| HTML | HyperText Markup Language |
| 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 |
| PC | Personal Computer |
| PCM | Pulse Code Modulation |
| RCV | Receive |
| PEQ | Parametric Equalization |
| PLC | Packet Loss Concealment |
| PM | Parameter Manager |
| ROM | Read Only Memory |
| PS Key | Persistent Store Key |
| 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 |