Qualcomm Technologies International, Ltd.

BCSW-CVC-HS-6-0-2 2M-HS

Tuning Guide

80-CT441-1 Rev. AH

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Revision history

Revision	Date	Change reason
1	December 2013	Original publication. Alternate document number: CS-00309817-UG
2	December 13	Editorial updates
3	March 14	Editorial updates
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5	July 14	Updated for ADK 3.5
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7	September 16	Updated for ADK 4.1
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1 cVc overview

Clear Voice Capture (cVc) is 2-mic headset (HS) audio processing software running on Qualcomm BlueCore™ Multimedia Integrated Circuit (IC).

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

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 is completed by 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 headsetsound quality.

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

1.1 Supported software versions

This Tuning Guide describes the audio adjustments of the cVc BCSW-CVC-HS-6-0-2 algorithm. Use the same audio tuning procedure when the algorithm is used on these ICs.

Table 1-1 Part number matrix

IC Supported	cVc Product Code	Version SysID	NB (8 k)	WB (16 k)	cVc License Key Part Number
CSR8670 (Flash)	BCSW-CVC-HS-6-0-2	BI0E	Yes	Yes	BCSW-CVC-HS-6-0-2
CSR8675 (Flash)	BCSW-CVC-HS-6-0-2	BI0E	Yes	Yes	BCSW-CVC-HS-6-0-2

NOTE CSR8670/CSR8675 flash IDs 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.

1.2 cVc 8th Generation new features

New or improved features since the previous release (BCSW-CVC-5-6-1, A008) that improve performance or affect the tuning process include:

- All cVc Generation 8 feature support
- **Simple Speech Recognition** support
- Parameter for **Front Mic Bias** and updated defaults in the dual Mic signal Separation (DMSS) block added
- Automatic Gain Control (AGC) Module default blocks updated
- Accoustic Echo Cancelation (AEC) block updated
- Auxiliary Stream Mix blocks updated
- Microphone Configuration tool support
- Reduced tuning compatibility for WNR
- Reduced tuning compatibility for Noise Suppression
- Half duplex mode support

2 cVc tuning prerequisites

Before starting to tune their devices, product developers should become familiar with the principles of acoustic performance and the tunable parameters supported by the Parameter Manager tool

2.1 SPI communication protocol drivers

The cVc HS Parameter Manager tool requires Serial Peripheral Interface (SPI) drivers. The appropriate drivers are included in the QTIL ADKs, if needed.

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

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

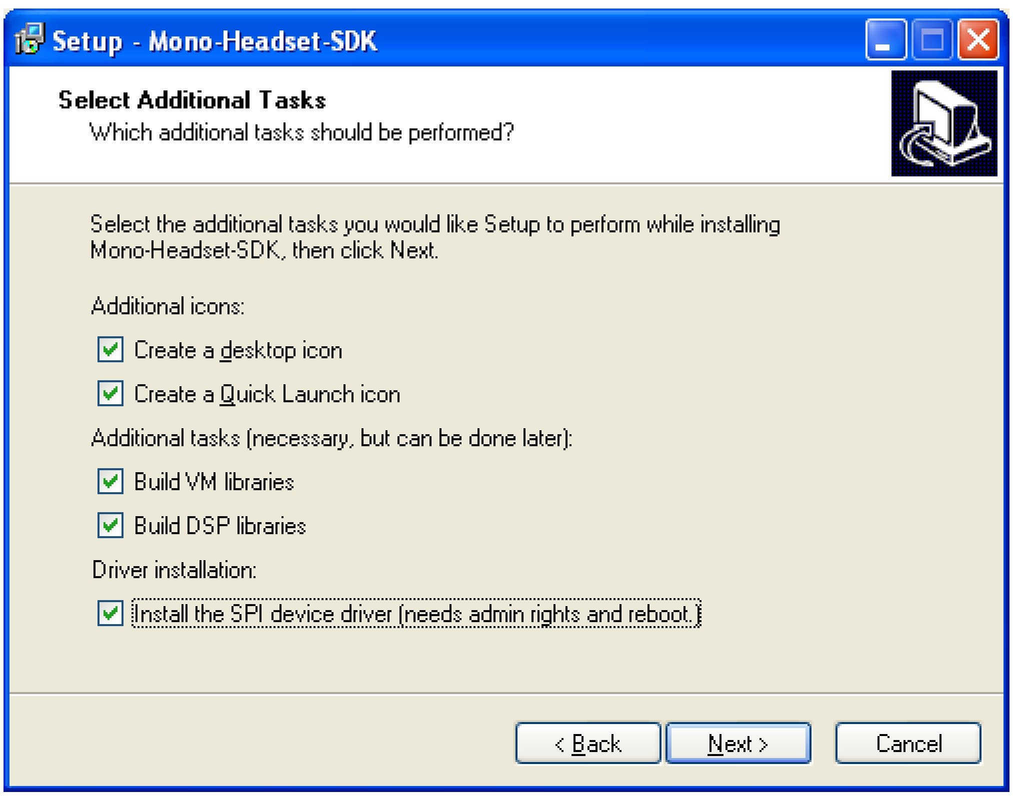


Figure 2-1 SDK software setup window with driver installation option selected

2.2 Hardware interfaces

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

2.3 Parameter Manager tool

The Parameter Manager tool 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-HS-6-0-2 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 the Parameter Manager through the Universal Front End (UFE) application.

2.3.1 Installation for ROM ICs

By default the ADK installation creates a subdirectory on the program files of the PC: **C:\Program Files\CSR™\<InstallerName>**

A corresponding **Start** menu link is created during the installation process: **Start -> All Programs -> <ADK Name> -> Tools -> UniversalFrontEnd**

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

2.3.2 Installation 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**

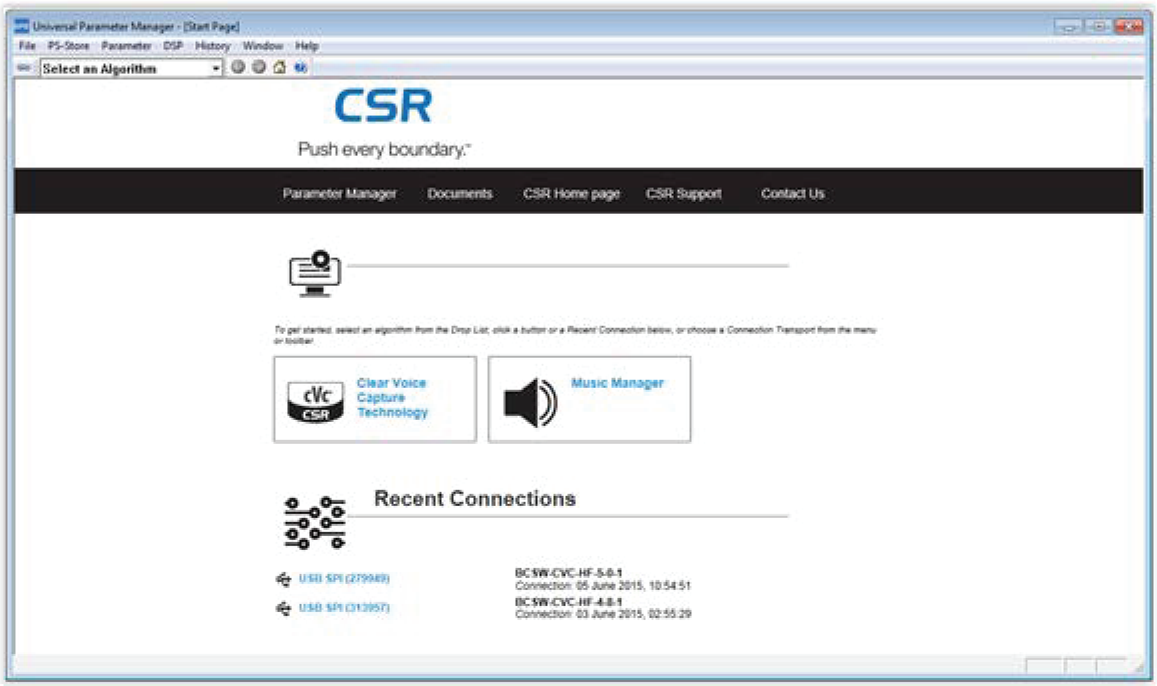


Figure 2-2 Accessing Parameter Manager from UFE

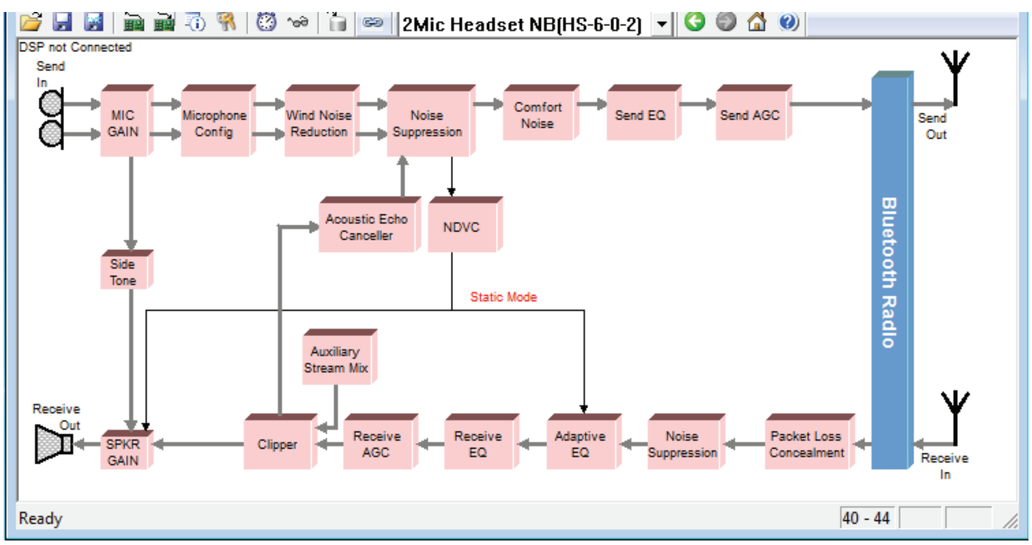


Figure 2-3 Parameter Manager window, Static mode (Narrow Band example)

The Parameter Manager window displays when the application is started (in this example it show that it is connected and in the Static mode).

In Monitoring mode, the Parameter Manager window provides live feedback and statistics, while the algorithm is running during an active call.

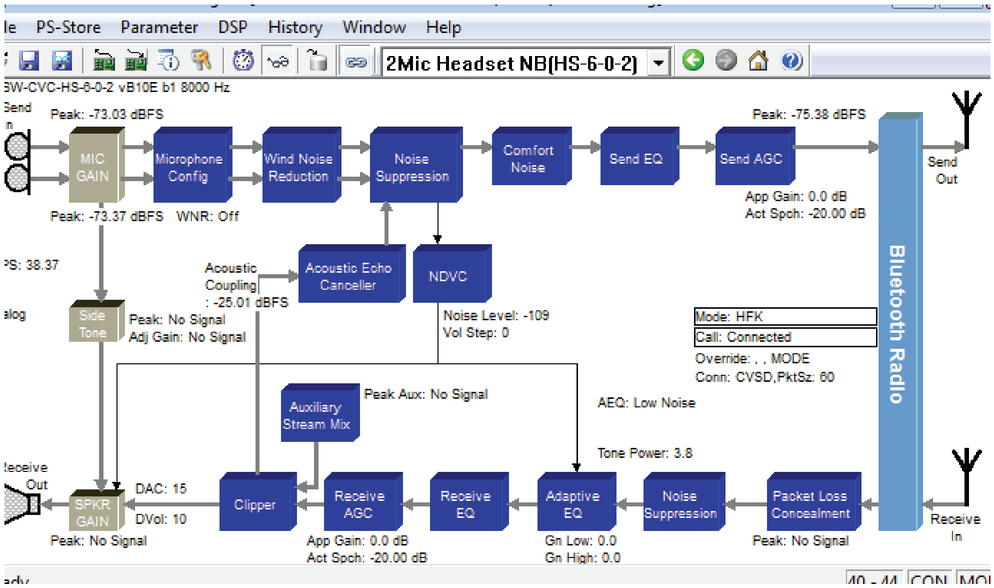


Figure 2-4 Parameter Manager window in Monitoring mode

3 cVc tuning overview

Tuning cVc involves adjusting the major processing blocks on the Parameter Manader window, and setting the gains at each cVc interface point.

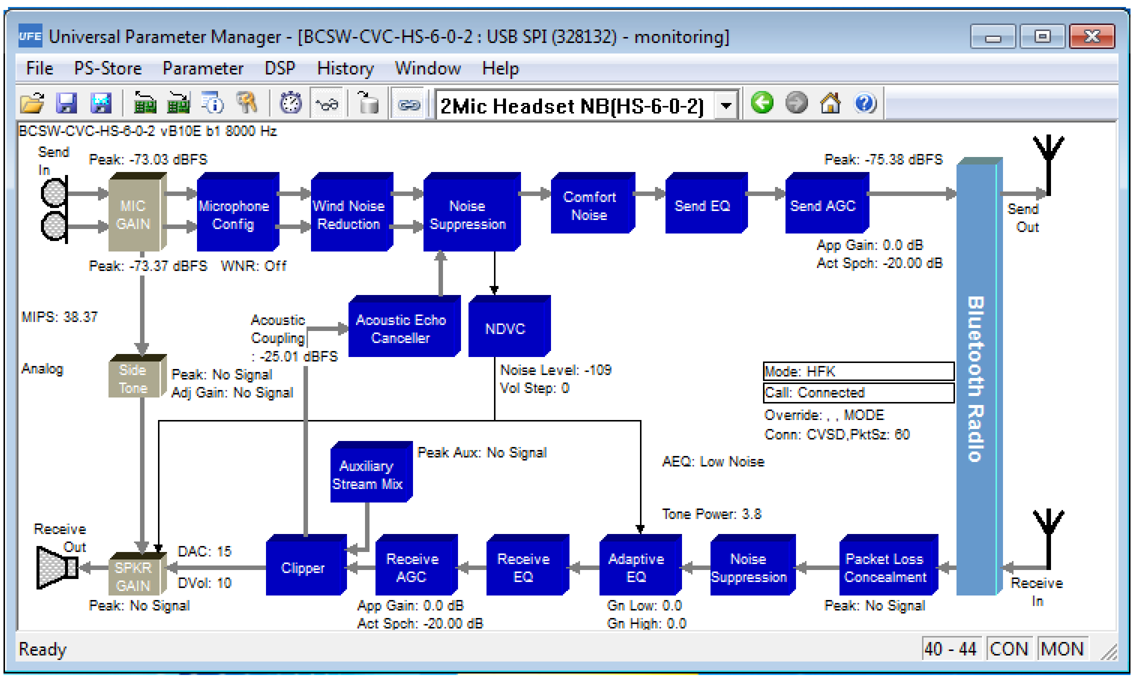


Figure 3-1 Processing blocks on the Parameter Manager window, Monitoring mode

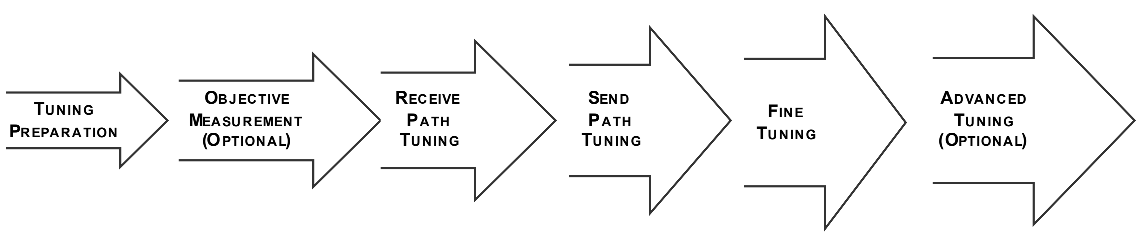


Figure 3-2 Six cVc HS tuning stages

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

3.1 cVc tuning stages

1. **Tuning Preparation**: The process of preparing the test environment and setting up the test equipment.
2. **Objective Measurement** (optional): Uses a HATS system to characterize frequency response, loudness rating, and distortion characteristics of the DUT.
3. **Receive Path Tuning**: Tuning of the Receive Path processing blocks of the cVc HS algorithm (PLC, Receive AGC, Speaker Gain, and Noise Suppression).
4. **Send Path Tuning**: Tuning of the Send path processing blocks of the cVc HS algorithm (Microphone Gain, Send AGC, Noise Suppression, Acoustic Echo Canceller and Comfort Noise).
5. **Fine Tuning**: Minor adjustments are made to the processing blocks as necessary (Receive AGC, Receive EQ, Clipper, Auxiliary Stream Mix, Send EQ, Send AGC and Send Noise Suppression)
6. **Advanced Tuning** (optional): Adding/tuning the advanced feature processing blocks.

The following features may be enabled and tuned to enhance the audio performance.

Enabling and tuning the following processing blocks, as necessary, to enhance audio performance::

- Adaptive EQ with Frequency Expansion
- NDVC
- Auxiliary Stream Mixer
- Side Tone

NOTE Current consumption slightly increases as a result.

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 necessary..

3.2 cVc tuning flowchart

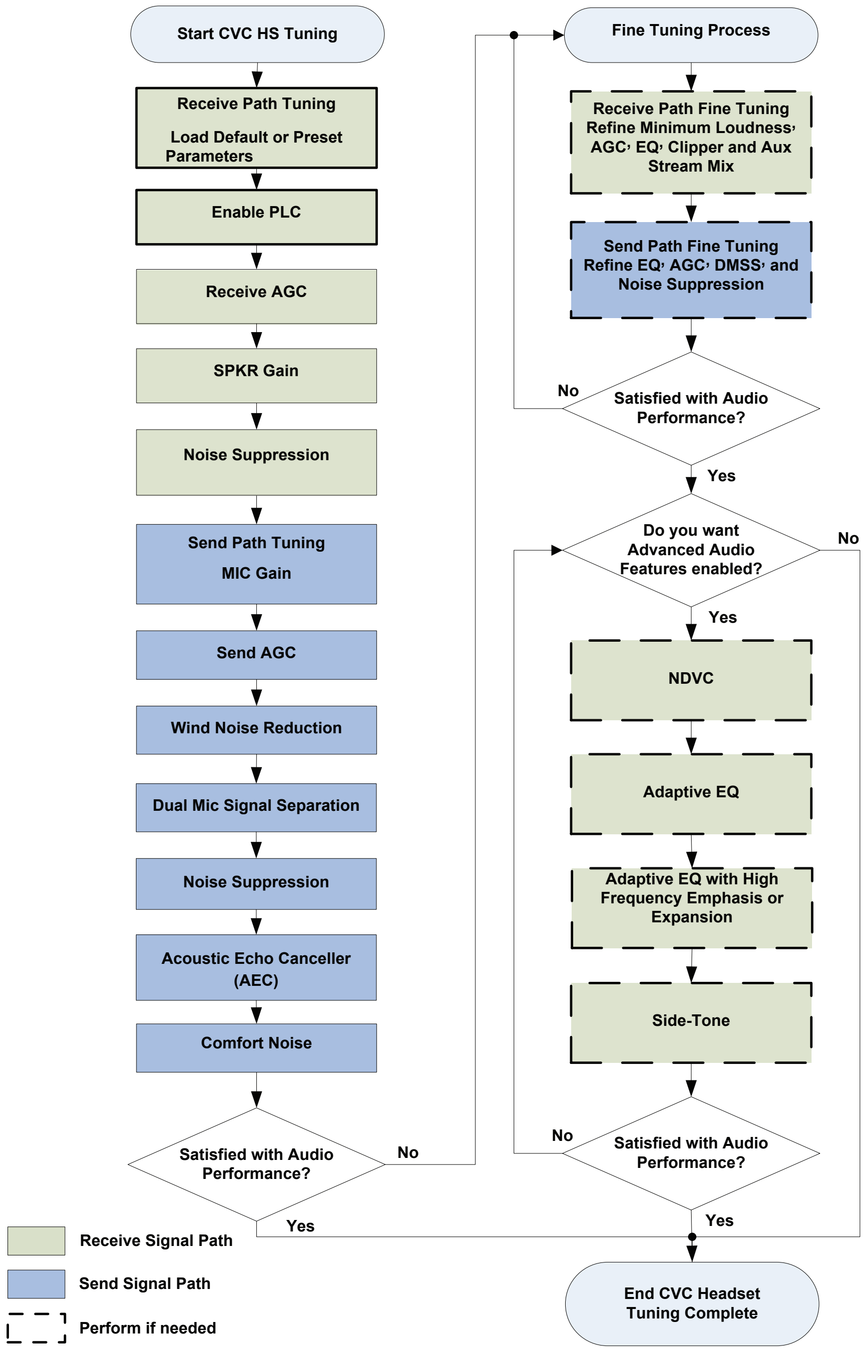


Figure 3-3 cVc tuning flowchart

4 cVc tuning preparation

This version of cVc software is designed for use with headset devices that have a two microphone input channels.

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.

4.1 Headsets design guidance and SPI access

Before tuning headsets check that the headset has been manufactured to meet the QTIL 2-Mic headset design guidelines described in *2-Mic Headset Design Guidelines*.

A partial list of recommendations is:

- The audio components and plastics are production-level or production-intended.
- The microphone arrangement and industrial design is consistent and within the cVc 2-mic design limitations, as in the QTIL DEV-SYS-MONOHS-1A extension type headset, with microphones in-line with the mouth and properly separated. Use a microphone separation distance of 30 mm, with a tolerance of +170 mm /-10 mm.
- Microphones are oriented orthogonal to each other. The microphone furthest from the mouth is oriented to face outwards while the microphone nearest the mouth is oriented to face towards the direction of the mouth. The device has SPI connectivity to a PC running the Parameter Manager software available.
- The SPI breakout cabling does not obstruct or interfere with the headset's microphone and receiver.
- Before testing, the headset's battery is fully charged or has an external power source.
- The headset is paired to a mobile telephone supporting the Bluetooth Headset profiles.
- During testing, the headset is worn in the way it was designed to be. For example, if the headset is an inner-ear type, it should be worn in this fashion.
- Make a short call with the default cVc HS parameters to test the operation before making any adjustments.

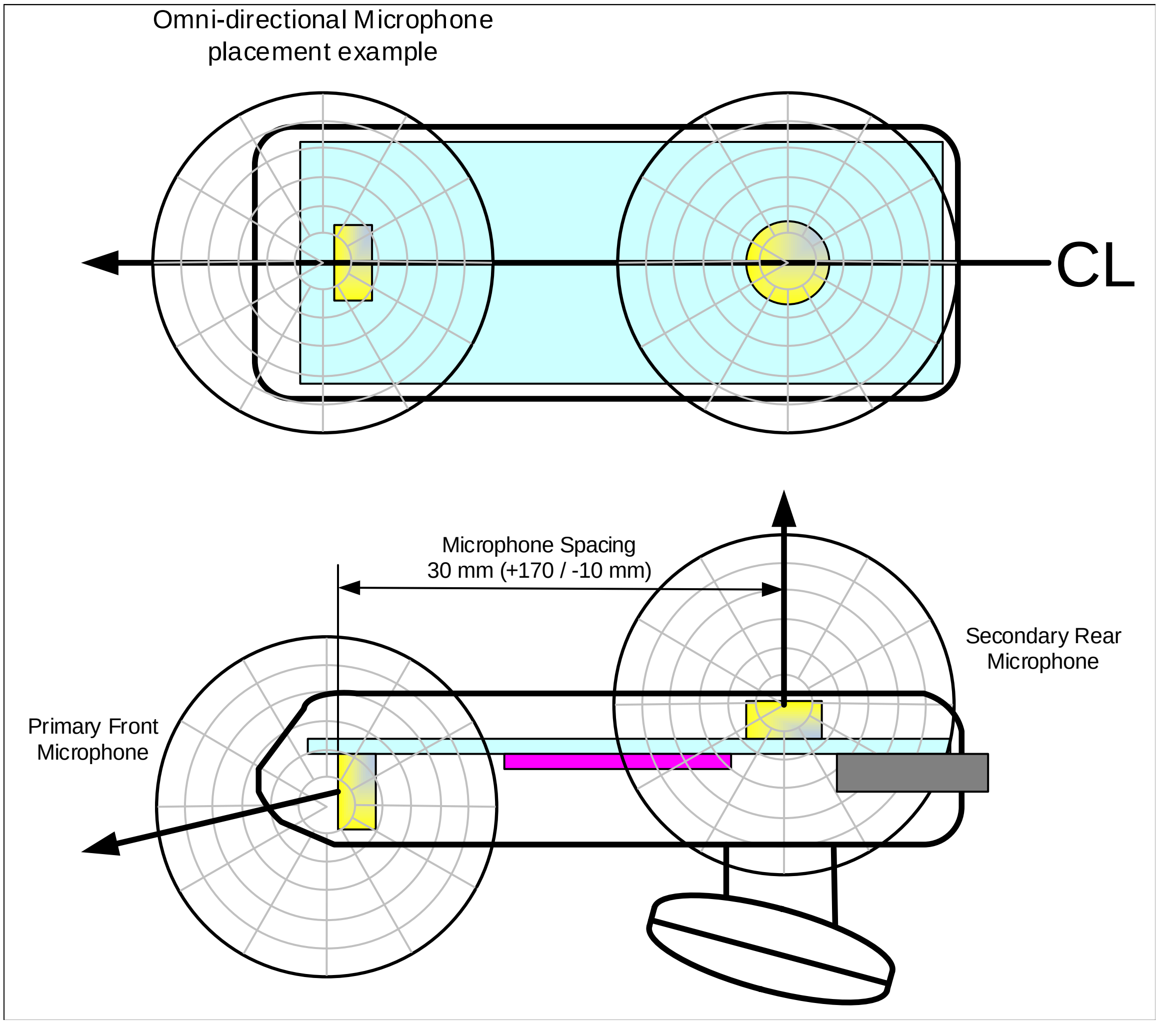


Figure 4-1 DEV-SYS-MONOHS-1A Extension Headset

4.2 Phones models and network types

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

- Phone models and local networks vary and affect the sound quality, while also affecting 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 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 necessary, 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.

Use a GSM mobile telephone as the primary tuning phone. After completing tuning, check the headset sound quality with a CDMA mobile phone for similar results. In CDMA-dominant countries, it is adequate to 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 the headset product, use the Bluetooth headset Profile (HSP) with an implementation of AT commands for turning off the mobile phone's noise reduction and echo cancellation processing.

4.3 Tuning environment

The tuning process requires that the tuning environment is a controlled space that imitates, or can simulate, 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.

4.4 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.

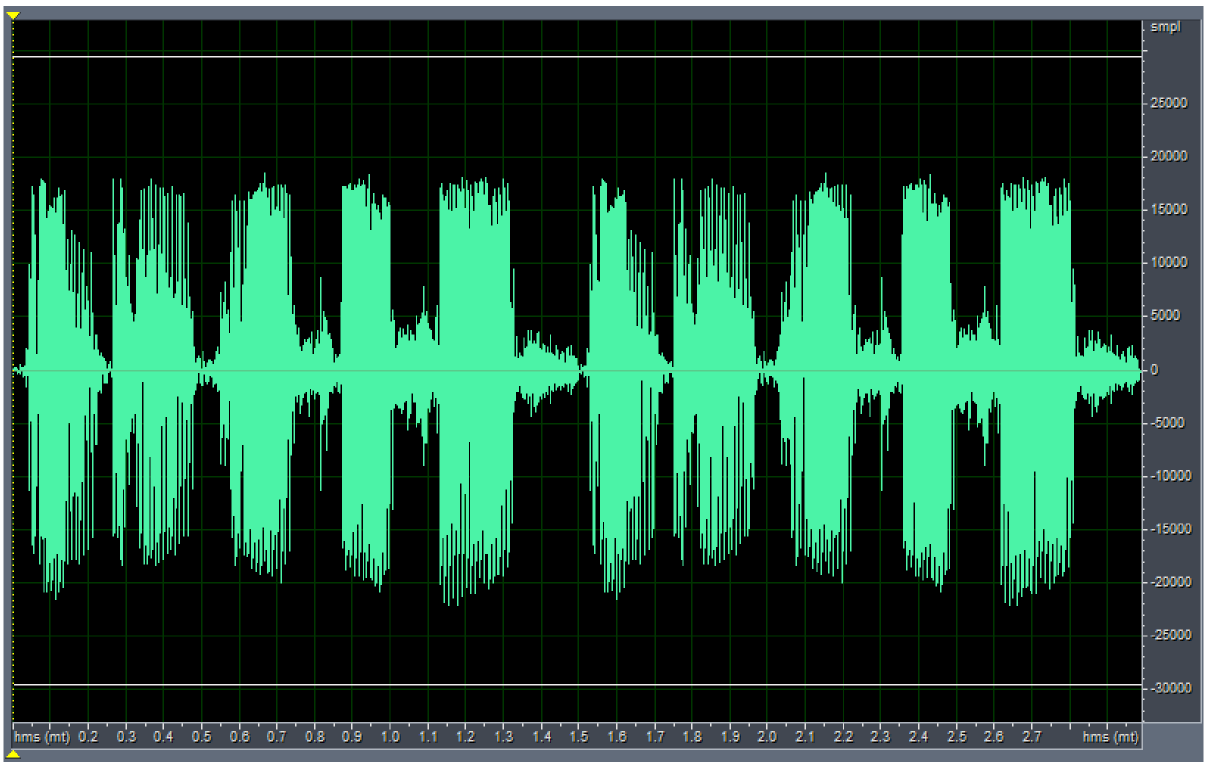


Figure 4-2 Recorded waveform of the English phrase “one two three four five”

5 Instrumentation

5.1 Sound Pressure Level meter

For measuring 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 5-1 Recommended SPL 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

Use a Head and Torso Simulator (HATS) that can perform electro-acoustic measurements on telephones per ITU-T recommendations.

Ensure that the HATS system can test:

- Send/receive frequency response
- Send/receive loudness ratings
- Receive-path THD+N

6 cVc Tuning preparation checklist

Tuning consideration	Check
Production or production-intended audio components and plastics	
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 Sound Pressure Level meter	❑
■ Calibrate the Sound Pressure Level Meter to a 1 kHz 94 dB re 20 μPa sine tone	❑
■ Ensure all cables and power supplies are in proper working order	❑
■ Parameter Manager tool connected to the headset system	❑
Accesories	
■ Ear Plugs	❑
Personal	
■ Far-end (landside) subject	❑
■ Near-end (headset) subject	❑
Documentation	
■ *BCSW-CVC-HS-6-0-2 Parameter Manager User Guide*	❑
■ *cVc 2-mic Headset Design Guidelines*	❑

7 cVc quick start guide

To perform simple tuning, set receive/send gain path and tune the cVc algorithm for acceptable echo cancellation and doubletalk performance. Additional tuning may be necessary to achieve acceptable performance, depending on the hardware and design of the product

This method requires an active SCO connection, with SPI communication, using the UFE in Monitoring Mode.

To establish SPI communication, ensure that the UFE is in Monitoring Mode.

7.1 Set Receive Gain path

1. Adjust **SPKR Gain** to the required maximum volume of the HFK.
2. Measure the echo signal at the microphone with a SPL meter or by measuring the Receive Loudness Rating (RLR).

NOTE To prevent microphone clipping, maximum volume should not exceed 115 dB peak measured at the microphone.

7.2 Set Send Gain path

1. Select Link Channels.
2. Adjust the microphone gains so that the speech signal is approximately -18 dBfs.

7.3 Set Microphone configuration

Measure and populate the **Mic Separation Distance** parameter as measured from the center point of the two microphones.

7.4 Accoustic echo canceller

1. With SPPKR Gain at predetermined Max volume, have a person on the far end check for echo. If echo is heard, select **Reuse Primary Filter**.
2. If echo is still present after selecting Reuse **Primary Filter,** see the details for tuning the Acoustic Echo Canceller.

8 cVc tuning procedures

8.1 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 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

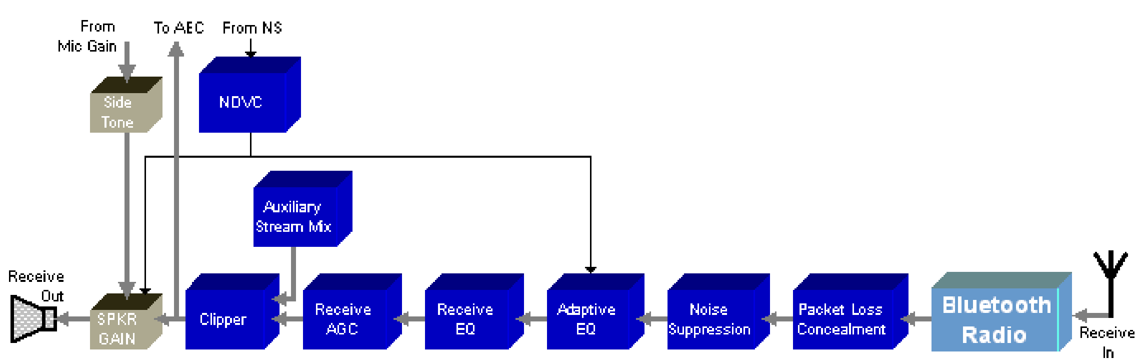


Figure 8-1 Receive Path processing blocks

NOTE Repeat all tuning measurements at least twice.

8.2.1 Load preset parameters

If the headset device is being tuned for the first time, begin tuning with the default parameters provided with the cVc HS release.

To load the defaults, use 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 headset 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 **Documentation** on the Parameter Manager opening window 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 echniques such as pitch based waveform substitution. Leave the Packet Loss Concealment enabled to achieve the best audio quality.

To disable Packet Loss Concealment, check **Bypass Packet Loss Concealment**.

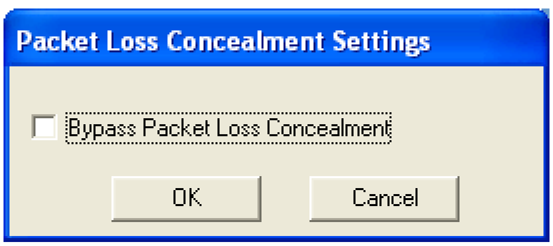


Figure 8-2 Packet Loss Concealment settings window

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.

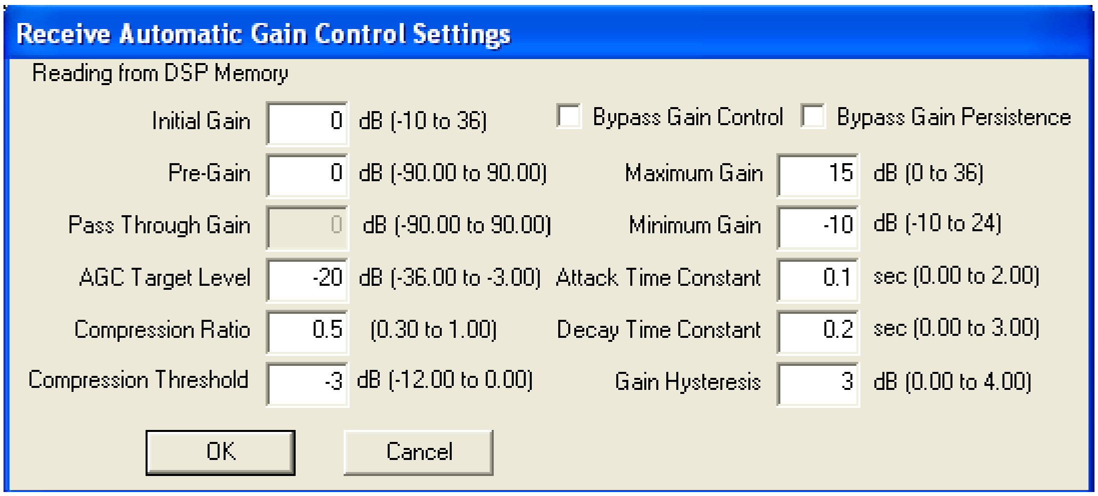


Figure 8-3 Receive Automatic Gain Control default settings

NOTE QTIL does not recommend bypassing the Receive AGC.

8.2.4 SPKR gain

Tuning the speaker involves determining the maximum receiver volume that the headset device supports. As the headset 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.

Distortion cannot be eliminated, but using good quality loudspeakers, amplifiers, microphones, leak-tolerant packaging or improved acoustic separation of the microphone and receiver helps to reduce echo and distortion.

The amount of distortion in the microphone and speaker determines the maximum volume. Limit the maximum receiver volume to the microphone's saturation level (approximately 110 dB SPL Average Fast Peek at the headset microphone position).

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 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.

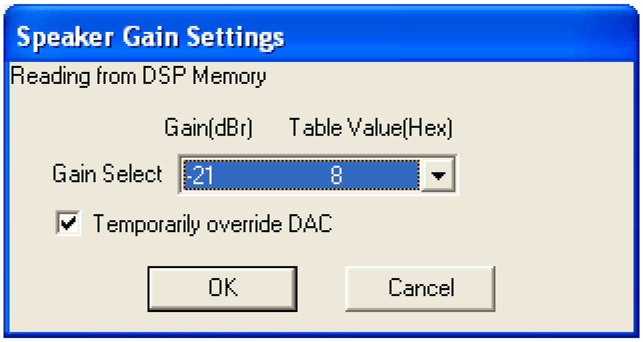


Figure 8-4 Adjusting the speaker gain

If there is distortion:

1. The near-end subject lowers the **SPKR Gain** or enables and adjusts the Clipper:
2. Using the Headset Configuration Tool, select the **Volume Control** tab to configure the required number of volume steps and set the maximum volume.

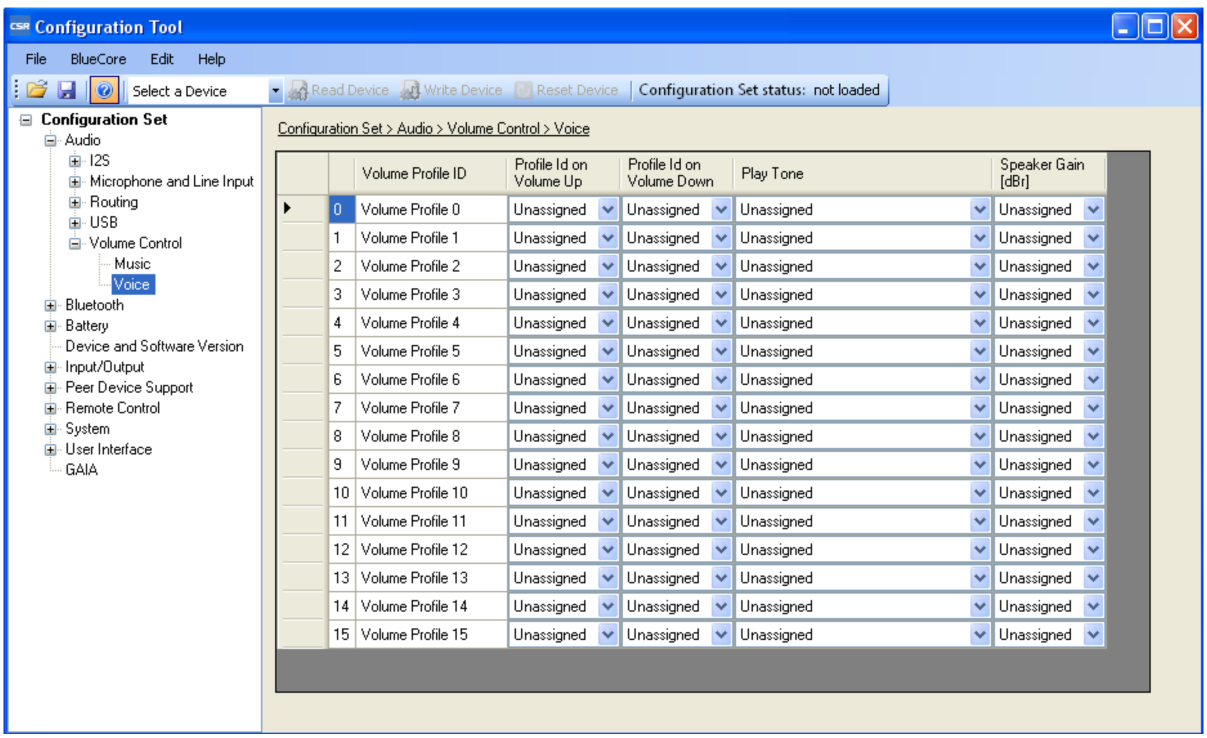


Figure 8-5 Headset configuration tool, audio gains

8.2.5 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 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. Increase the **HFK Aggressiveness** for more noise suppression (at the cost of voice quality). Decrease the **HFK 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 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-checked to reduce the processing load. However, because high quality mode only consumes about 1 MIPS, QTIL recommends leaving it enabled.

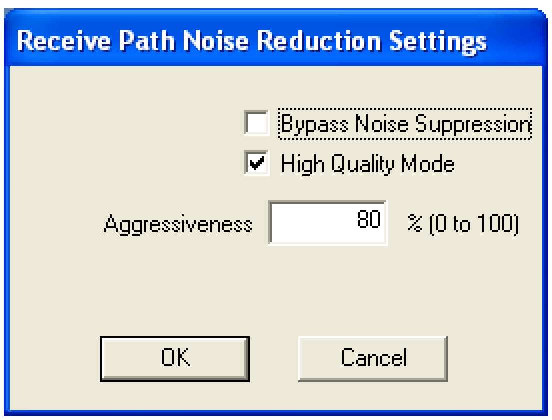


Figure 8-6 Receive Path Noise Suppression settings window

8.3 Send Path tuning

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

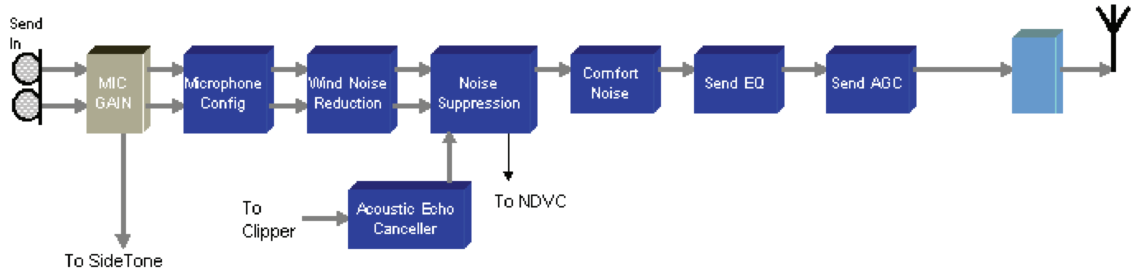


Figure 8-7 Send Path processing block diagram

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 headset call link primary and secondary microphone to have matching gain levels on the microphone.
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 and monitors the MIC GAIN Peak statistics.
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.

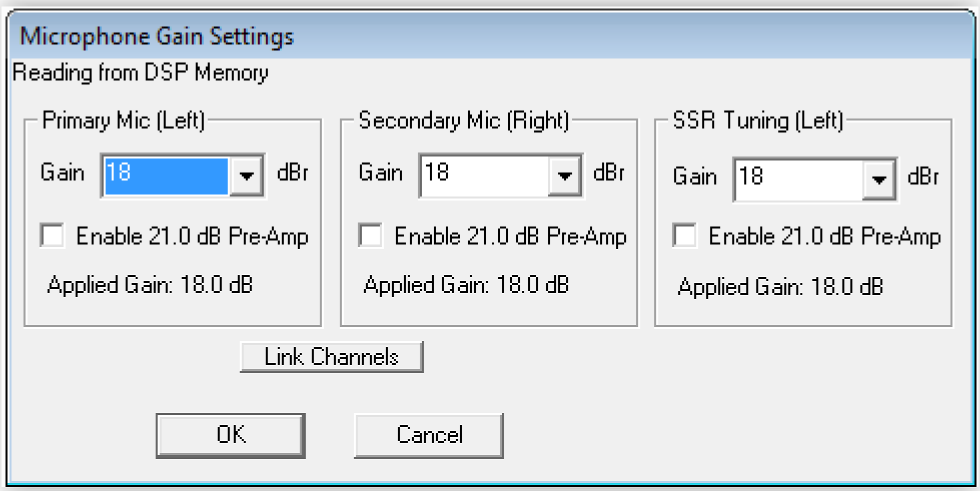


Figure 8-8 Microphone Gain settings window

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 -20 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 headsets without a close-coupled microphone and speakers:

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. Next, the near-end subject speaks the level speech phrase at approximately 90 dB SPL (C) Weighted Fast, measured 25 mm from the speaker’s mouth.
4. The near-end subject adjusts the **AGC Target Level** to achieve the necessary 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 -3 dB target scale to allow for overshoot, processing in the event of saturation or clipping.
5. The **Maximum Gain** that can be applied to the signal can also be limited.

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

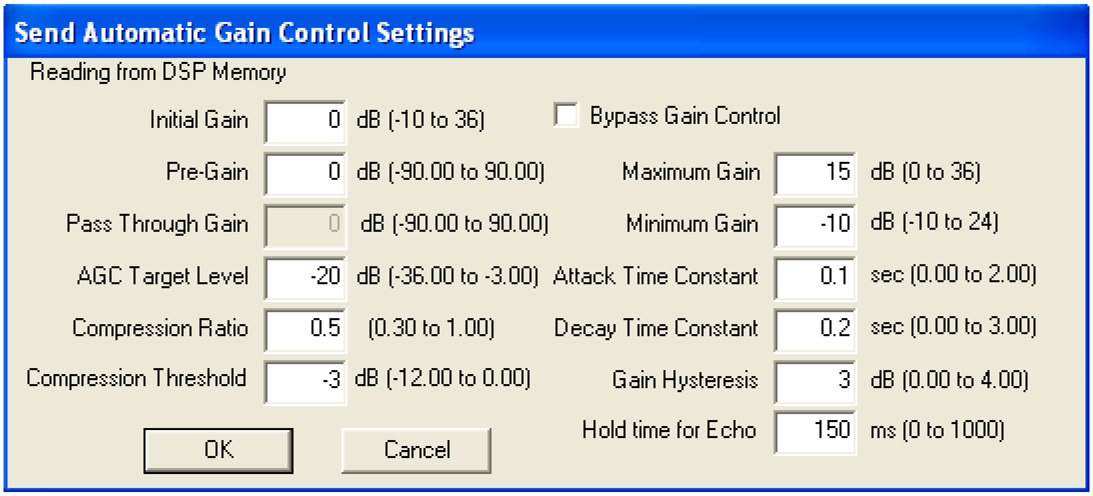


Figure 8-9 Send Automatic Gain Control settings window

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.

- **Applied 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.
- **Active Speech Level:** 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 Wind Noise Reduction

To tune for Wind Noise Reduction (WNR):

1. Set the **Wind Noise Gain Aggressiveness** to 100% to begin tuning WNR and **Bypass Gain Control** on the Send AGC.
2. Place Headset in a “quiet” environment, such as an office-type area.
3. Monitor **Send In** statistic, for ~20 seconds and record average value (confirm this value agrees with **Mic Internal Noise Level** setting found in the Microphone Configuration Settings .
4. Determine at what dB level you want Wind Noise to be detected. Add that number to the **Send In** statistic average (in a quite environment) and set **Wind Noise Silence Threshold** to that dB value . Any signal below the **Wind Noise Silence Threshold** is treated as “non-wind silence”.
5. The near-end subject slowly adds wind using a controlled source (i.e. fan), attempting to maintain a constant wind speed. Only add enough wind to determine when the WNR should start cleaning the audio. To listen to the audio without the influence of the WNR, you can temporarily check the **Bypass WNR** to determine the lowest wind speed. Once set, uncheck the **Bypass WNR**.
6. Maintaining a constant wind speed, the NEAR END subject should add speech occasionally and evaluate speech quality.

7. Adjust the **Wind Noise Gain Aggressiveness** and the **Wind Noise Silence Threshold** until you obtain the required performance.
8. Re-enable (un-check the **Bypass Gain Control**) the Send AGC.

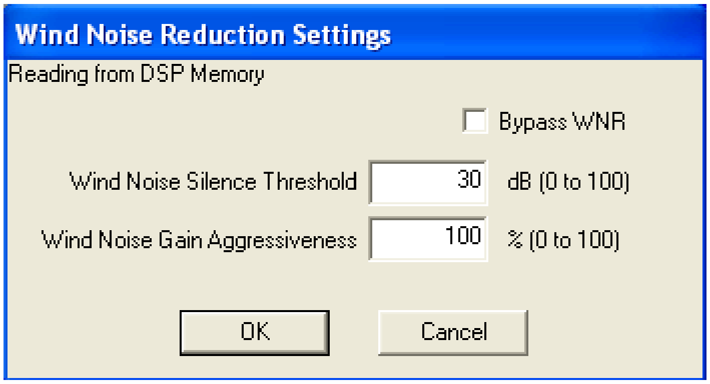


Figure 8-10 Wind Noise Reduction Settings window

8.3.4 Microphone Configuration

This section describes areas to fine tune on the Microphone Configuration Settings window.

To tune for Microphone Configuration:

1. Initiate a headset call.
2. Based on the headset to be tuned, adjust **Mic Separation Distance, Mic Internal Noise Level and Target Speech Degradation Factor** (if necessary).
3. The near-end subject introduces background noise into the testing environment.
4. The near-end subject speaks a test phrase or a normal conversational phrase, continually.
5. Under different noise conditions, the far-end subjectively evaluates the noise level and its quality with and without near-end speech.

8.3.5 Send 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. Setting this parameter to 100% suppresses up to 20 dB of noise.

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. 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.

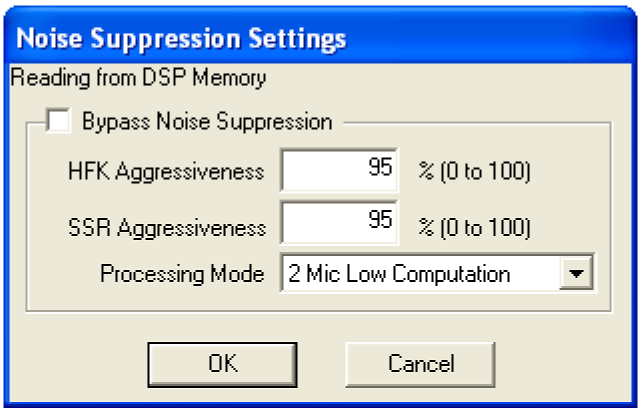


Figure 8-11 Send Path Noise Suppression settings window

8.3.6 Acoustic Echo Canceller

Table 8-1 Acoustic Echo Cancellation settings field description

Settings	Description
Bypass Acoustic Echo Cancellation	Check this option when echo cancellation is not required
Bypass Residual Echo Reduction	The **Bypass Residual Echo Reduction** checkbox 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. Enable the **Residual Echo Reduction** feature for the best echo cancellation performanc
Reuse Primary Filter	Under normal conditions, the filter converges to the required level so that echo is reduced. This option is selected when the AEC filter convergence is required to 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	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:	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.

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

Settings	Description
Low Volume Mode threshold	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. To determine the threshold, lower the volume until no perceived echo or an acceptable echo is heard.
Bypass Half Duplex	Enables or disables additional attenuation that is applied to the microphone signal during receive speech or double talk.
Howling Supression Coupling Threshold	When the Acoustic Coupling level has exceeded the Howling Suppression Coupling Threshold Half-Duplex is enabled until the Acoustic Coupling level goes below the Howling Suppression Coupling Threshold.

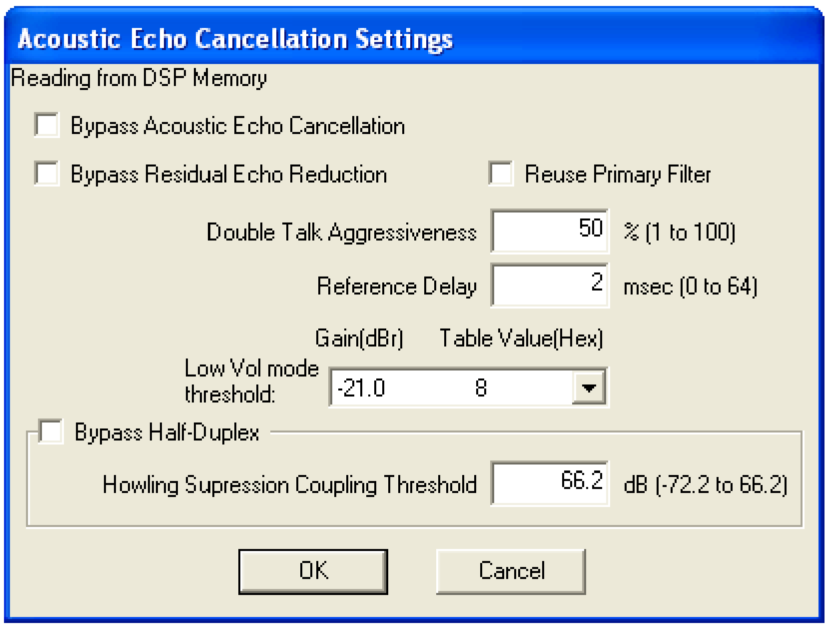


Figure 8-12 Acoustic Echo Cancellation settings window

To tune the Acoustic Echo Canceller

1. Initiate a headset call, 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”, “check”, “echo”) 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 fine tune set the AEC by setting the low volume mode threshold:
   a. Initiate a headset call.
   b. The near-end subject selects Bypass Acoustic Echo Cancellation.

c. The far-end subject speaks short bursts of speech (for example, “one”, “two”, “hello”, “ok”, “check”, “echo”) and checks for echo at the far-end.

d. 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.

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.

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 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 enabled.

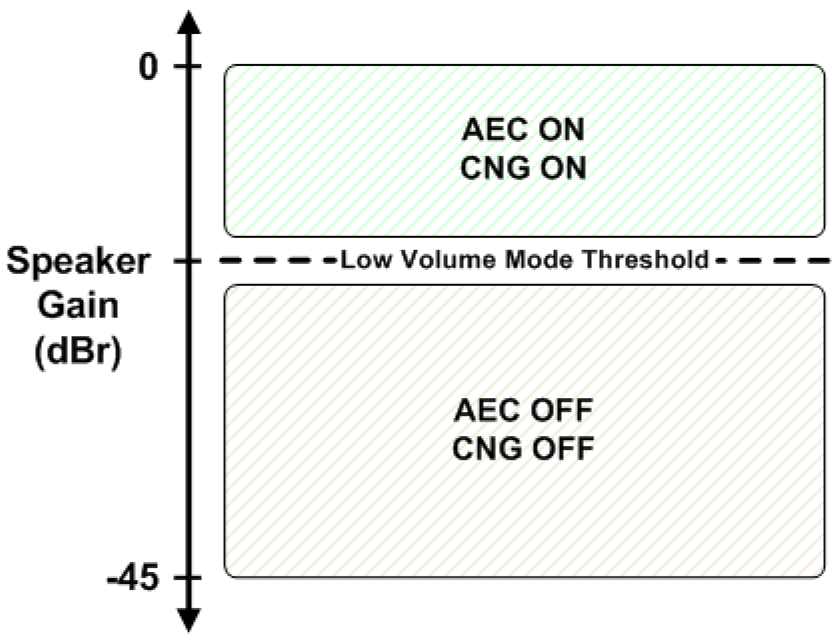


Figure 8-13 Low Volume Mode threshold

8.3.7 Howling Control

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

To fine-tune the Howling Control:

1. Initiate a headset call.

2. While the far-end subject checks for echo, the near-end subject increases the headset 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**.

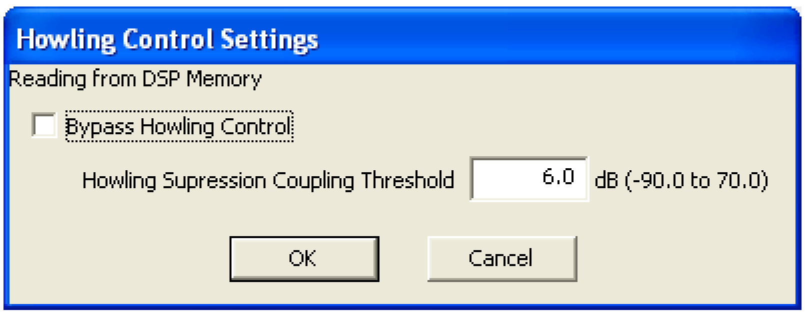


Figure 8-14   3: Howling Control settings window

8.3.8 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.

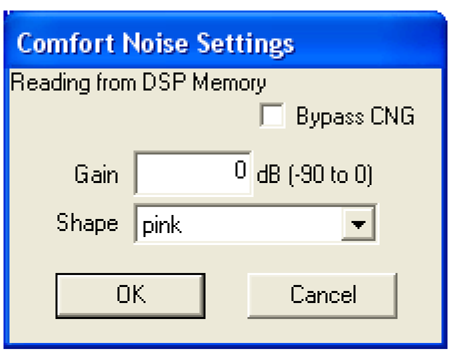


Figure 8-15   Comfort Noise settings window

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”, “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 headset receive and send paths are tuned, minor parameter changes may be necessary to reach a good performance level. Some products have unique acoustic designs or have special headset sound quality requirements for the product.

9.1 Receive Path fine-tuning

9.1.1 Setting minimum speaker gain 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 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 the required and minimum number of volume steps.

9.1.2 Receive AGC

To tune the Receive AGC:

1. Adjust the **AGC Target Level** to the 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 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** define 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 **Compression Thrreshold,** while the slope of gain curve below the **Compression Threshold** 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 necessary, the receive EQ can be used for frequency shaping to fit an appropriate response curve. The GUI allows the Receive EQ parameters to be graphically selected. See the *BCSW-CVC-HS-6-0-2 Parameter Manager User Guide* for 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 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 headset receiver.
5. The near-end subject adjusts the **Clip Point** until distortion is controlled by the Clipper (distortion is added).
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 will be 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 before the DAC.

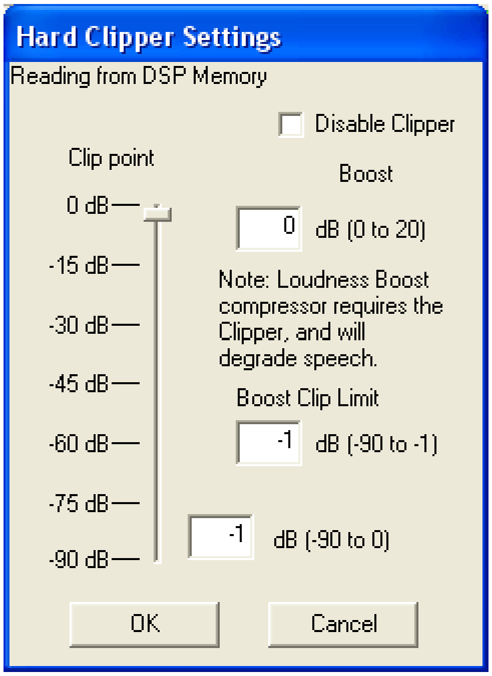


Figure 9-1 Hard Clipper setting window

9.1.5 Auxiliary Stream Mix

Auxiliary Stream Mix is always enabled. Stream mixing only occurs during a stream mix event (such as tone play).

It mixes an auxiliary signal with the SCO input signal. The auxiliary signal could be tones or another source such as voice prompts. The ratio of the mixture is controlled by using the slider controls to achieve the required balance on the receive out signal. Changing a stream mix using a slider will inversely control 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 but caution should be taken if the percentage sums >100% as saturation could occur. The auxiliary signal can be boosted by using the **Auxiliary Gain** parameter. The maximum limit on the auxiliary gain is 24 dB.

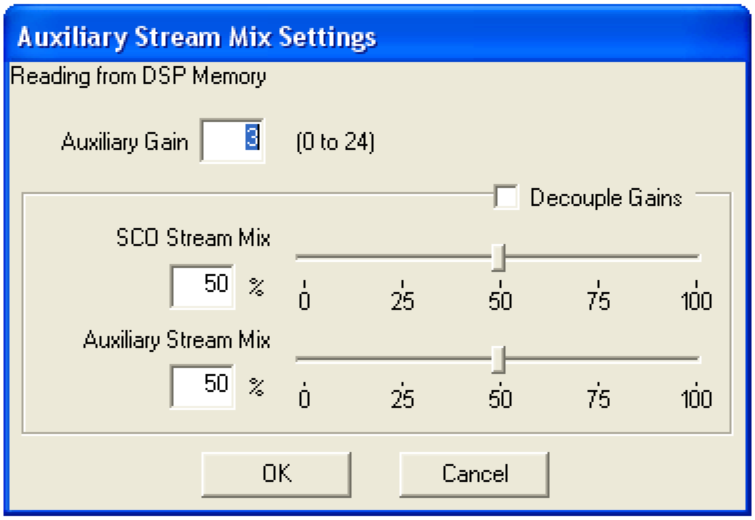


Figure 9-2 Auxiliary Stream Mix setting window

9.2 Send Path fine-tuning

9.2.1 Send EQ

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

If necessary, the Send EQ can perform frequency shaping to fit a required response curve. The graphical user interface allows the Send EQ parameters to be graphically selected.

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 -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 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 target level. The gain factor follows the compression curve above the **Compression Threshold,** while the slope of gain curve below the **Compression Threshold** is unity.

9.2.3 Simple Speech Recognition (SSR) Aggressiveness

The secondary tuning parameter in the Noise Suppression block, **SSR Aggressiveness**, controls the amount of noise suppression applied to the send signal during SSR mode. Setting this parameter to 100% suppresses up to 20 dB of noise. In the SSR mode, the cVc-enabled headset acts as a speech capture device used for voice recognition applications. Adjust the SSR mode gain to the specific demands of the voice recognition application.

Use 95% **SSR Aggressiveness** as this provides a good balance of having superior noise suppression with minimal voice distortion.

10 Advanced tuning

10.1 Noise-Dependent Volume Control

Tune the Noise-Dependent Volume Control (NDVC) 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 determined in SPKR gain.

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 **Min Noise L** > **evel** 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). Monitor the **Noise Level** statistic and place this value in the **Max noise level field**.
6. During the high 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.

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. The maximum loudness is limited to 0 dB under the highest noise condition.
- The Total **SPKR Gain** = **SPKR Gain** + Maximum NDVC Gain Limit
- The Total **SPKR Gain** is important when tuning the Send Path.
- 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 1 and should not be changed.
- Adjust the **Increasing Noise Attack Time Constant** and **Decreasing Noise Decay Time Constant** to a required level. A high time constant value causes the NDVC to react more slowly to changes in the background noise and lower values cause a quicker reaction.

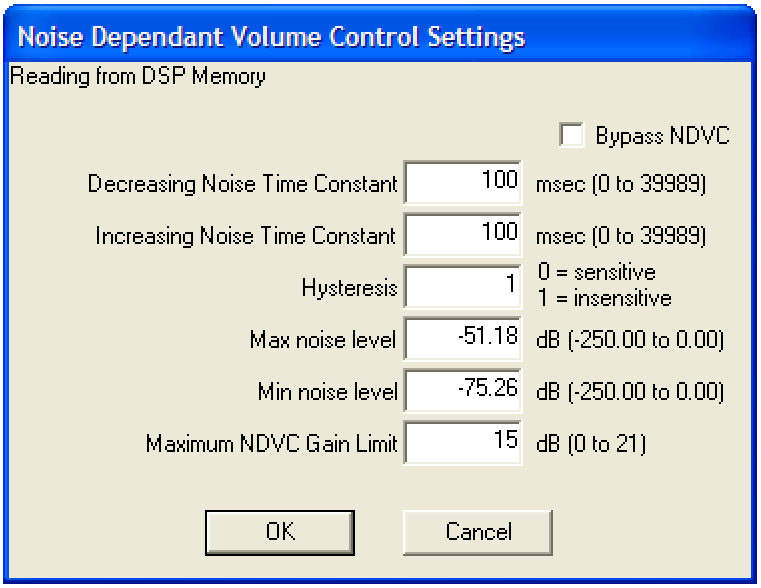


Figure 10-1 NDVC default settings

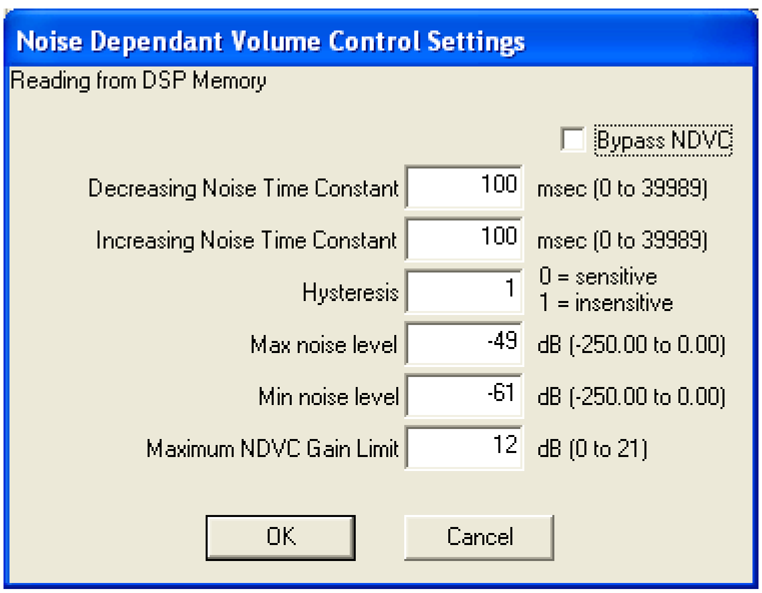


Figure 10-2 NDVC example alternate tuning

Noise- Dependent Volume Control settings

NOTE Adjust the **Max noise level** and **Min noise level** for your specific headset.

10.2 Adaptive EQ (AEQ)

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.

There are three systems available for the CSR86xx:

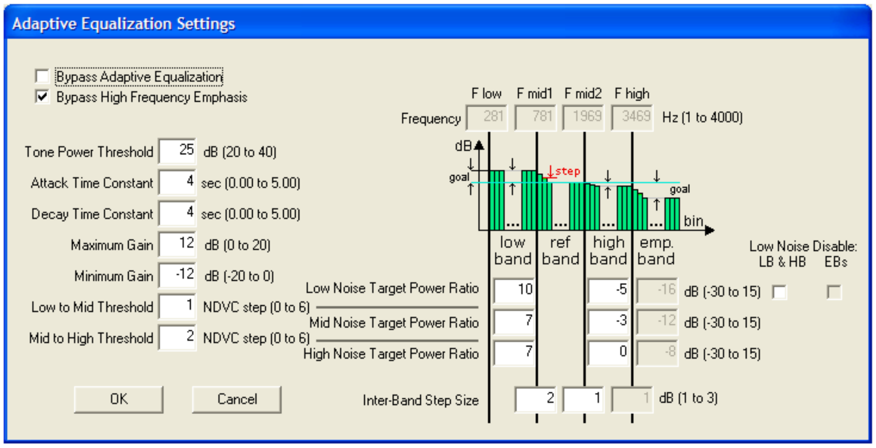
- Narrow Band plus High Frequency Emphasis: consists of AEQ (0 to ~3.5 kHz) plus the addition of an Emphasis Band (~3.5 to 4 kHz).
- Narrow Band plus Frequency Expansion: consists of an AEQ (0 to ~3.5 kHz) plus the addition of Expanded Bands (~3.5 to ~6.2 kHz).
- Wide Band: consists of an AEQ (0 to 8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

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. Therefore, by increasing the frequency components that contribute to the consonants while in the presence of noise, the intelligibility can be improved.

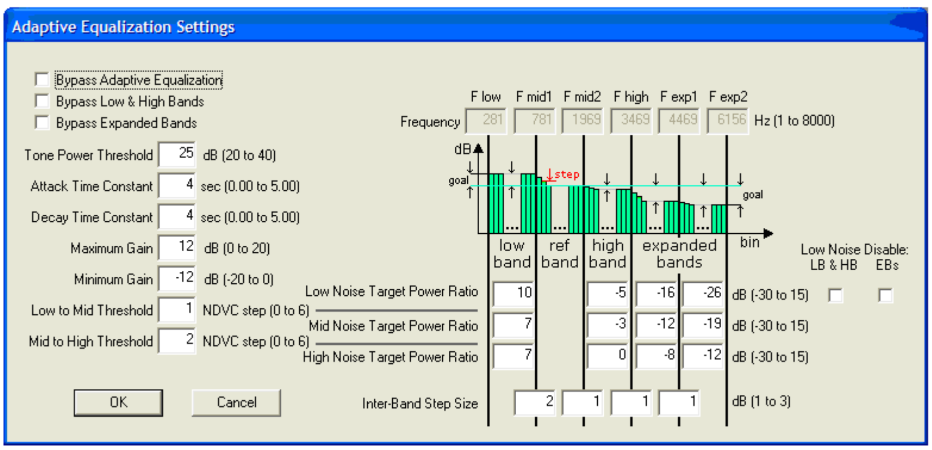
To maintain a consistent amplitude level, the Adaptive Equalization block adaptively increases the high frequencies relative to the middle frequencies while reducing low frequencies accordingly.

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

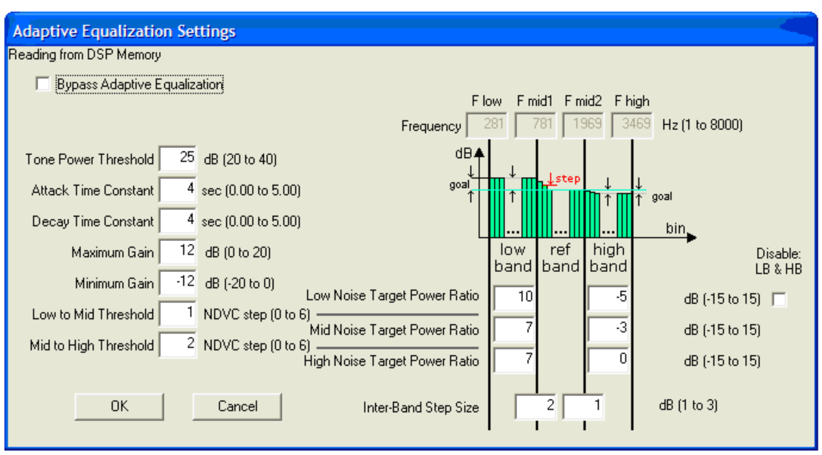
NOTE For the headset to benefit from this feature, the loudspeaker must provide adequate fidelity delivered to the ear of the user. Good examples are headsets fitted with a gel ear bud that seals the ear canal, conversely an open air, hard plastic speaker headset is not a good design for use with the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



Narrow Band AEQ plus High Frequency Enhancement



Narrow Band AEQ plus Frequency Expansion



Wide Band AEQ

Figure 10-3 Adaptive EQ 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.

To bypass application of the Adaptive EQ in quiet situations, check **Low Noise Disable LB & HB**. 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 example, 0 to 5).

Adaptive EQ (AEQ) 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.

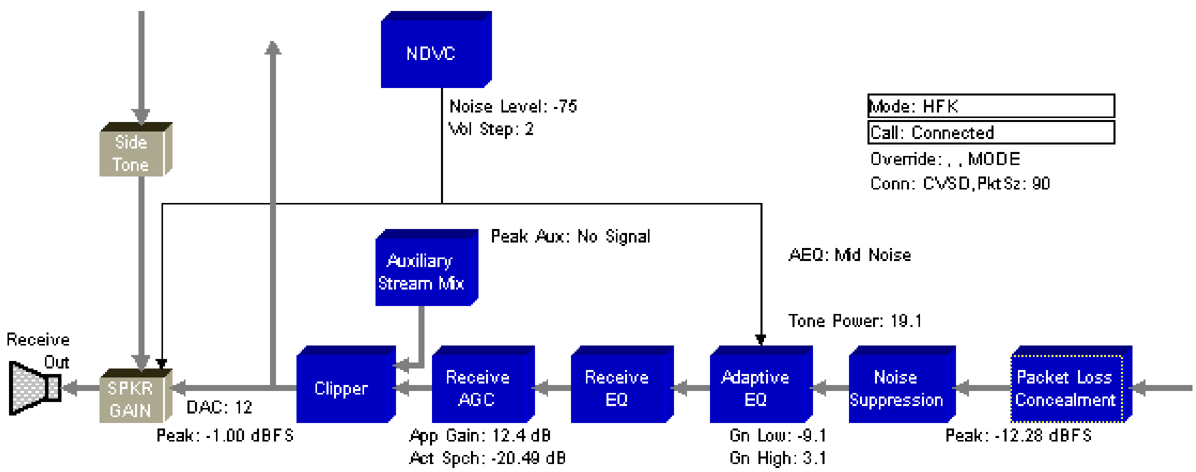


Figure 10-4 Adaptive Equalization switching to mid-noise tier

Adaptive EQ (AEQ) 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.

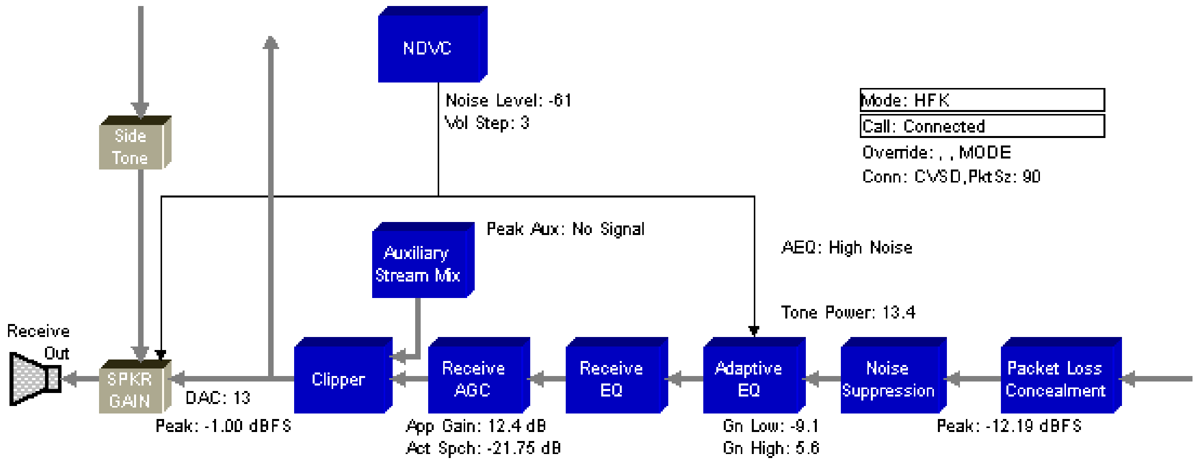


Figure 10-5 Adaptive Eqialization switching to high noise tier

To tune the AEQ:

1. Initiate a headset call.
2. To isolate the Adaptive Equalization, check either**Bypass Expanded Bands** or **Bypass High Frequency Emphasis** (bypassed throughout the Adaptive Equalization tuning) to disable any high frequency expansion/enhancement.
3. Check **Bypass Adaptive Equalization** to disable the Adaptive Equalization.
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 30 seconds of receive speech, uncheck the **Bypass Adaptive Equalization** box and listen to the receive speech (again, for about 30 seconds).
6. Raise/lower the **Low Noise Target Power Ratio** parameters under the appropriate low/high band column(s) to adjust the spectral shape of the low and high bands.
7. The low and high bands can be disabled for low noise by checking the **Low Noise Disable LB & HB** box.
8. Check **Bypass Adaptive Equalization** to disable the Adaptive Equalization.
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 ro seconds of receive speech, uncheck **Bypass Adaptive Equalization** and listen to the equalized receive speech (again, for about 30 seconds).
11. Raise/lower the **Mid Noise Target Power Ratio** parameters under the appropriate low/high band column(s) to ajust the spectral shape of the low and high bands.
12. Check **Bypass Adaptive Equalization**to disable the Adaptive Equalization.
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 30 seconds of receive speech, uncheck **Bypass Adaptive Equalization** and listen to the equalized receive speech (again, for about 30 seconds).
15. Raise/lower the **High Noise Target Power Ratio** parameters under the appropriate low/high band column(s) to adjust the spectral shape of the low and high bands.

10.3 Narrow band plus high frequency emphasis

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

High Frequency Emphasis repairs speech information (3469 Hz to 4000 Hz) that is lost because of 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.

10.4 Narrow band plus frequency expansion

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.

To enable Frequency Expansion, un-check **Bypass Expanded Bands**.

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 a value 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 lower than what is found in the reference speech band (781 Hz to 1969 Hz).

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 the value of **Vol. Step**.

To tune the High Frequency Emphasis Portion of AEQ:

1. Initiate a headset call.
2. Check **Bypass High Frequency Emphasis** to disable the High Frequency Emphasis.
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 about 30 seconds of receive speech, uncheck **Bypass High Frequency Emphasis** and listen to the receive speech (again, for about 30 seconds).

5. Raise/lower the **Low Noise Target Power Ratio** parameters under the appropriate **emp. band** column to adjust the spectral shape of the emphasis band.
6. To disable the Emphasis Band low noise, check **Low Noise Disable EBs**.
7. Check**Bypass High Frequency Emphasis** to disable the High Frequency Emphasis.
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 30 seconds of receive speech, uncheck **Bypass High Frequency Emphasis** and listen to the emphasis receive speech (again, for about 30 seconds).
10. Raise/lower the **Mid Noise Target Power Ratio** parameters under the appropriate **emp. band** column to adjust the spectral shape of the emphasis band.
11. Check **Bypass High Frequency Emphasis** to disable the High Frequency Emphasis.
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 30 seconds of receive speech, uncheck **Bypass High Frequency Emphasis** and listen to the emphasis receive speech (again, for about 30 seconds).
14. Raise/lower the **High Noise Target Power Ratio** parameters under the appropriate **emp. band** column to adjust the spectral shape of the emphasis band .

To tune the Frequency Expansion Portion of AEQ:

1. Initiate a headset call.
2. Check **Bypass Expanded Bands** to disable the Frequency Expansion.
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 about 30 seconds of receive speech, uncheck **Bypass Expanded Bands** and listen to the receive speech (again, for about 30 seconds).
5. Raise/lower the **Low Noise Target Power Ratio** parameters under the appropriate **expanded bands** column(s) to adjust the spectral shape of the expanded bands.
6. The expanded bands can be disabled for low noise by checking the **Low Noise Disable EBs** option.
7. Check **Bypass Expanded Bands** to disable the Frequency Expansion
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 30 seconds of receive speech, uncheck **Bypass Expanded Bands** and listen to the expanded receive speech (again, for about 30 seconds).
10. Raise/lower the **Mid Noise Target Power Ratio** parameters under the appropriate **expanded bands** column(s) to adjust the spectral shape of the expanded bands.
11. Check **Bypass Expanded Bands** to disable the Frequency Expansion..
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 30 seconds of receive speech, uncheck **Bypass Expanded Bands** and listen to the expanded receive speech (again, for about 30 seconds).
14. Raise/lower the **High Noise Target Power Ratio** parameters under the appropriate **expanded bands** column(s) to adjust the spectral shape of the expanded bands.

10.5 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 used to increase or decrease 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.

   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/feedback condition.

8. Would the near-end like to have the side tone increase with changes? If no, skip this field, otherwise 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, otherwise set the **Filter State** to On.
10. The filter comprises three cascaded bi-quad stages, with each stage user configurable. The GUI allows the side tone filter EQ parameters to be graphically selected.
11. Figure 10-6 shows a recommended band pass configuration, were 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.

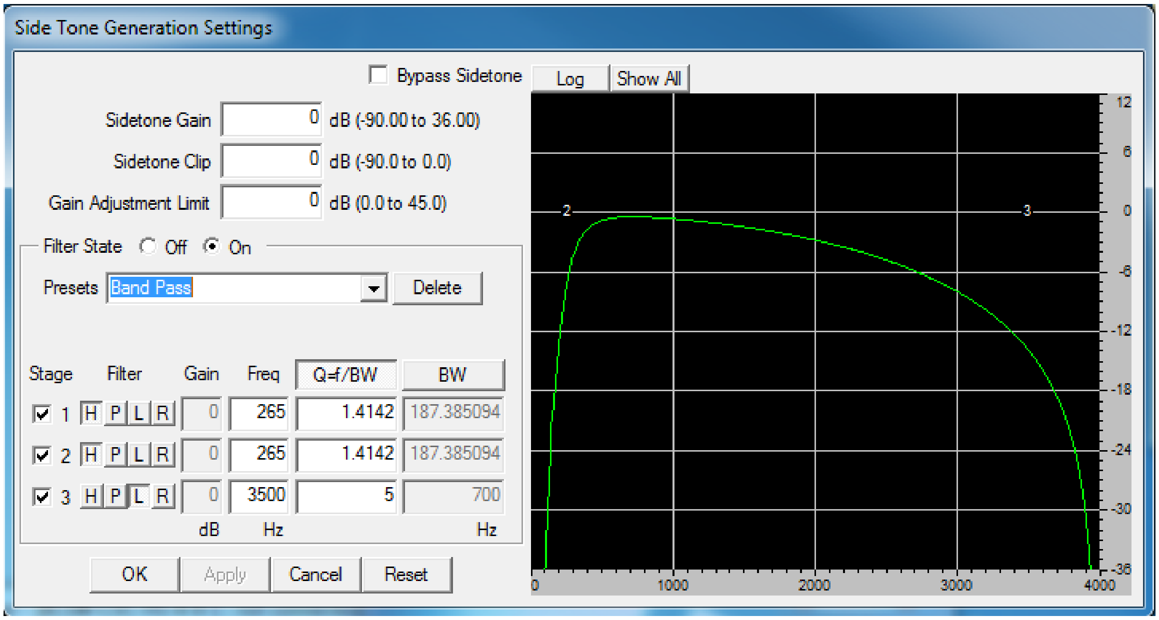


Figure 10-6 Side Tone Generation Settings window

12. 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.

Document references

Document	Reference
cVc 2-mic Headset Design Guidelines	80-CE530-1/CS-00218321-DC
BCSW-CVC-HS-6-0-2 Parameter Manager User Guide	80-CT442-1/CS-0030981-UG

Terms and definitions

ADC	Analogue to Digital Converter
ADK	Audio or Application Development Software
AEC	Acoustic Echo Cancellation
AEQ	Adaptive Equalizer
AGC	Automatic Gain Control
AT	Attention (modem command prefix)
B&K	Bruel & Kjaer
BlueCore	Group term for CSR's range of Bluetooth wireless technology chips.
Bluetooth	Set of technologies providing audio and data transfer over short-range radio connections.
BCSW	BlueCore Software
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
DOA	Direction Of Arrival
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	Hands Free Kit
HS	Headset
HSP	Headset Profile
HTML	HyperText Markup Language
i.e.	*Id est*, that is
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
NDVC	Noise Dependent Volume Control
NB	Narrow Band
NS	Noise Suppression
PC	Personal Computer
PCM	Pulse Code Modulation
RCV	Receive
PEQ	Parametric Equalization
PLC	Packet Loss Concealment
ROM	Read Only Memory
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
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