Qualcomm Technologies International, Ltd.

BCSW-CVC-HF-5-0-3 2M-HF

Tuning Guide

80-CT412-1 Rev. AH

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Revision history

Revision	Date	Description
1	December 2013	Original publication of this document Alternate document number: CS-00309822-UG
2	December 2013	Minor editorial corrections
3	March 2014	Minor editorial changes
4	May 2014	Editorial updates
5	July 2014	Updated for ADK 3.5
6	June 2015	Updated for ADK 4.0
7	September 2016	Updated for ADK 4.1
AH	November 2017	Updated for ADK 4.3, and to reflect new document numbering and revision scheme.

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1 cVc overview

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

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

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 handsfreesound quality.

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

The cVc HF application includes default settings that can be used for its audio processing. These defaults may require 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-HF-5-0-3 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-HF-5-0-3	A104	Yes	Yes	BCSW-CVC-HF-5-0-3
CSR8675 (Flash)	BCSW-CVC-HF-5-0-3	A104	Yes	Yes	BCSW-CVC-HF-5-0-3

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

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

1.2 cVc 8th Generation new features

New or improved features since the previous release (handsfreeA008) that improve performance or affect the tuning process include:

- All cVc Generation 7 feature support
- Automatic Gain Control (AGC) Module updated to improve tracking of target level
- Parametric Equalization (PEQ) Master Gain independent form bi-quad stages
- Tone volumes have been normalized between processing modes with and without cVc
- Microphone Configuration tool
- 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 HF 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**.

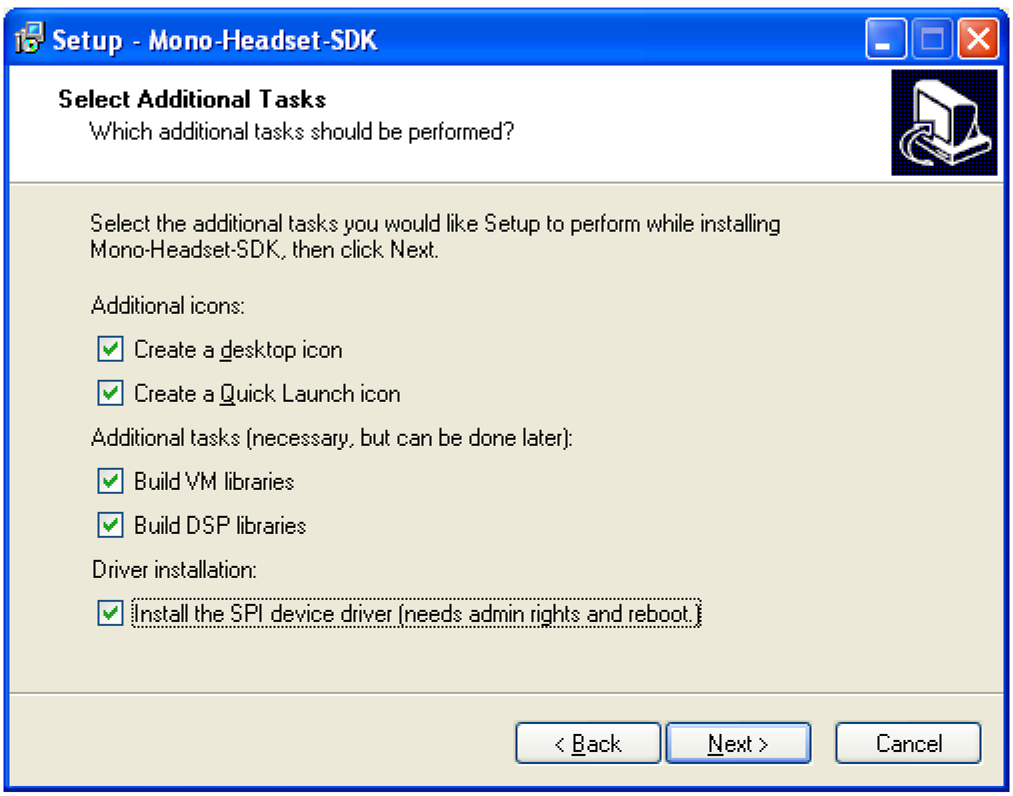


Figure 2-1 QTIL ADK software setup window with driver installation option selected

2.2 Hardware interfaces

The Parameter Manager and **PSTool** require an SPI connection to communicate with the target hardware. The handsfree 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-HF-5-0-3 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 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

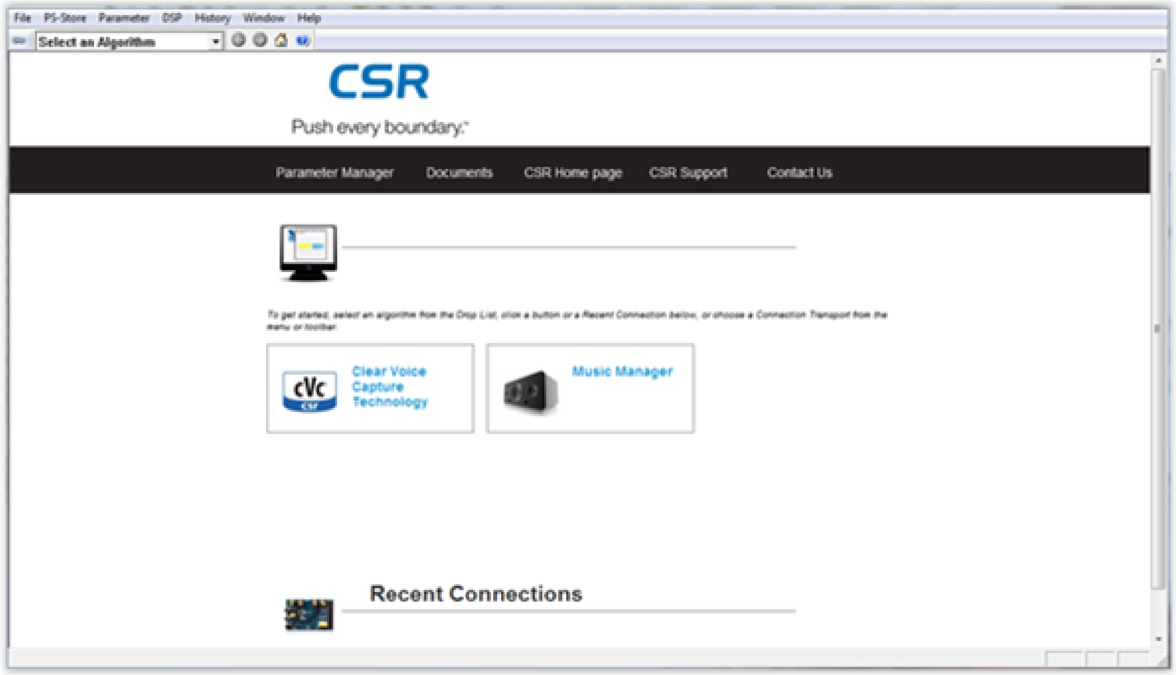


Figure 2-2 Accessing Handsfree Parameter Manager from UFE

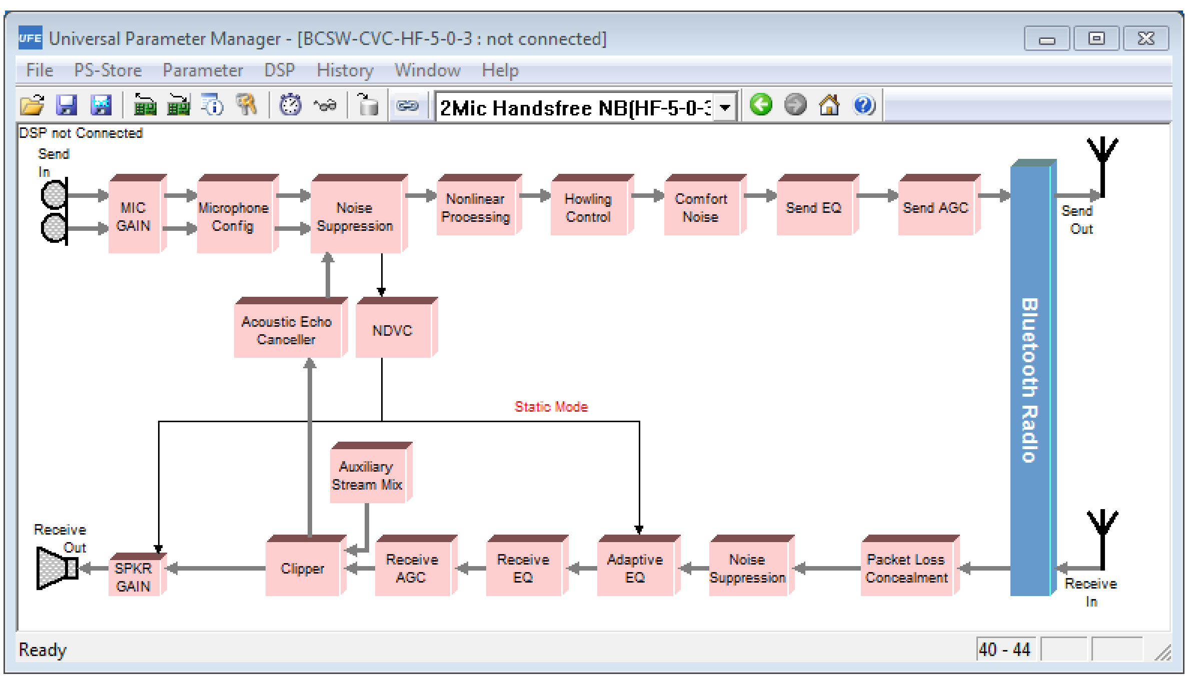


Figure 2-3 Parameter Manager window, Static mode (Narrow Band example)

The 2-mic CVC-HF-5-0-3 Handsfree Parameter Manager window displays when the application is started (in this example it show that it is connected and in the Static mode). The application is in a connected mode for a NB (narrow band, 8 KHz sample rate) system.

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.

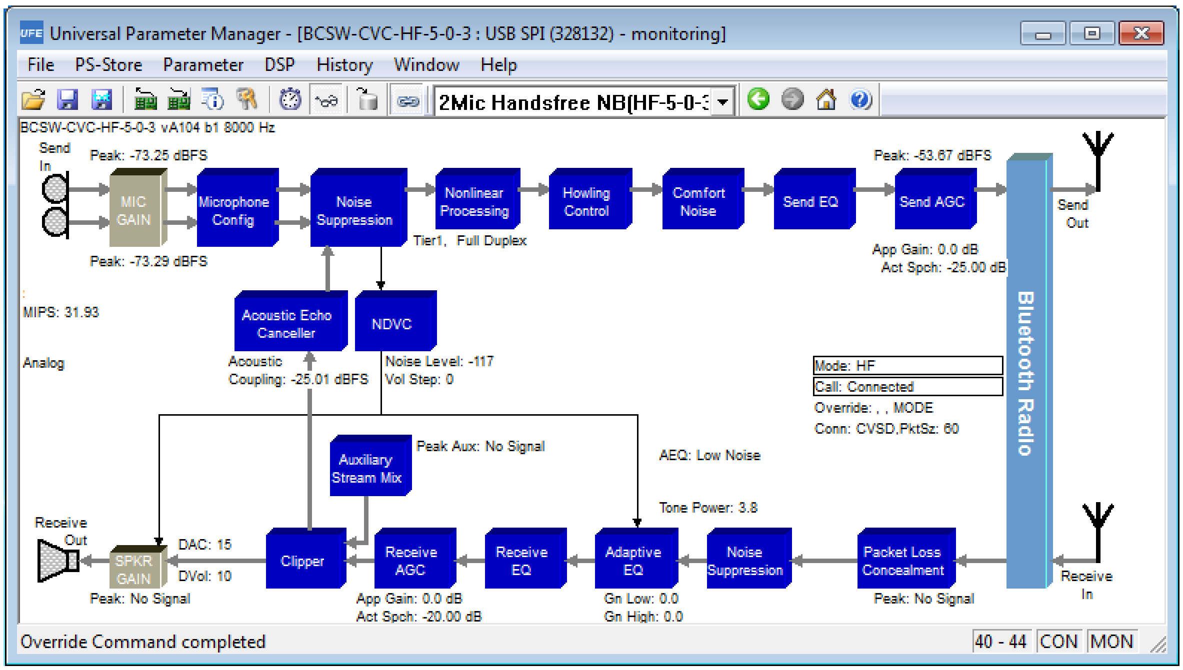


Figure 3-1 Processing blocks on the Parameter Manager window, Monitoring mode

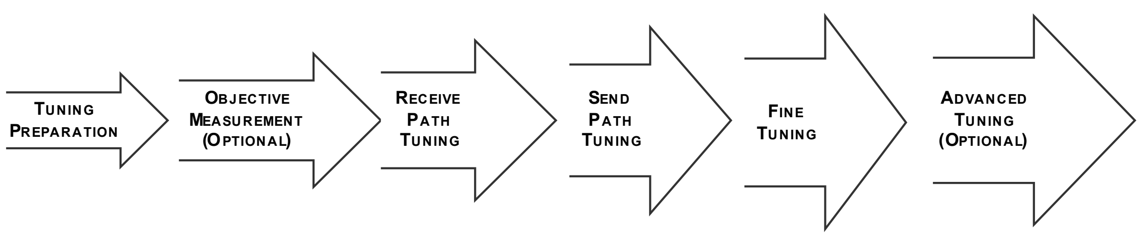


Figure 3-2 Six cVc HF tuning stages

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 HF algorithm (PLC, Receive AGC, Speaker Gain, and Noise Suppression).
4. **Send Path Tuning**: Tuning of the Send path processing blocks of the cVc HF 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.

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

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

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

- Adaptive EQ with Frequency Expansion
- NDVC

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

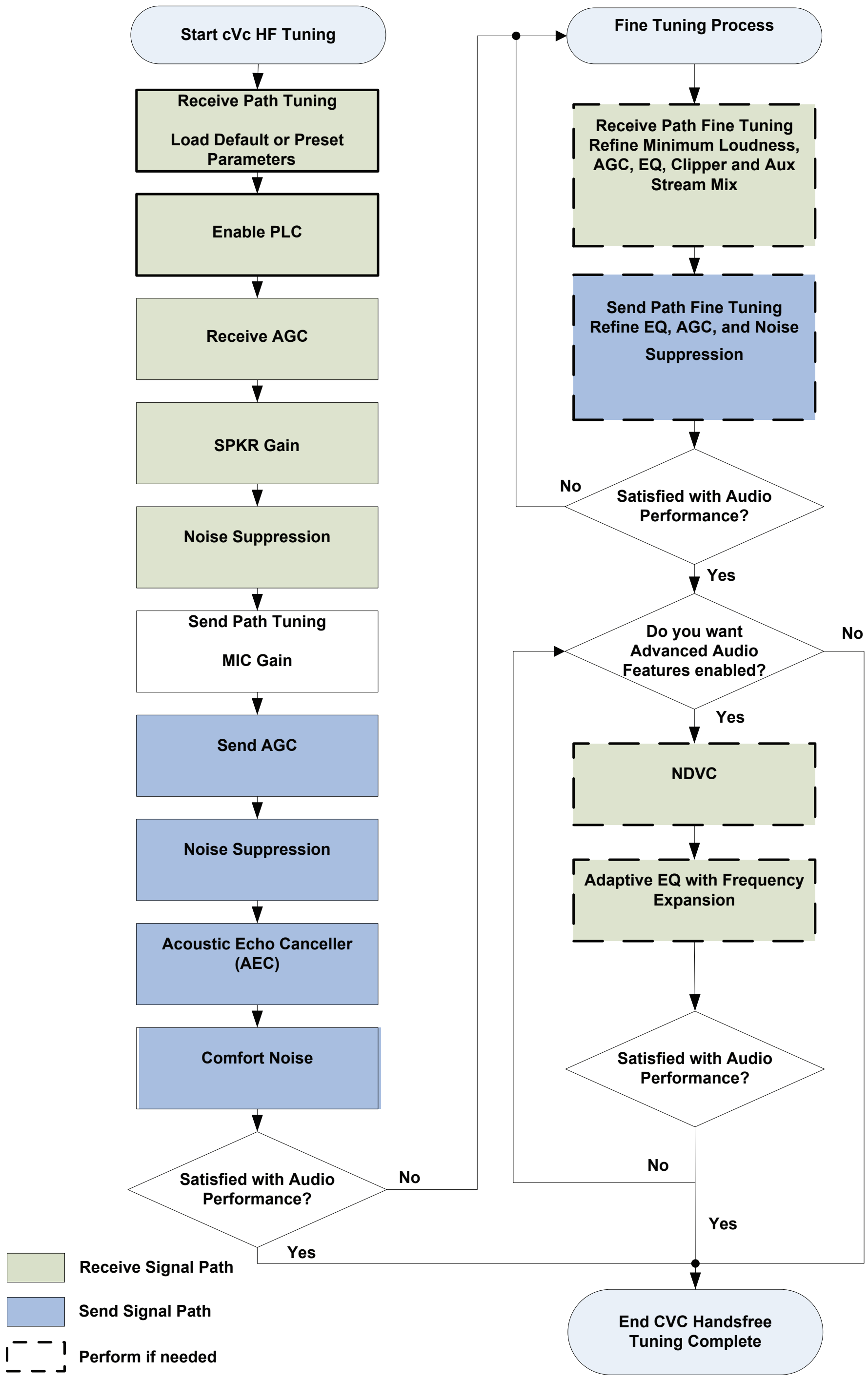


Figure 3-3 cVc tuning flowchart

4 cVc tuning preparation

This version of cVc software is designed for use with handsfree 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 In-vehicle handsfree system

Before tuning the handsfree system, ensure that:

1. The handsfree 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 handsfree microphone and receiver.
5. Before testing, the battery is fully charged or has an external power source.
6. The handsfree kit is paired to a mobile telephone supporting the Bluetooth handsfree profiles.
7. During testing, the handsfree kit is positioned in its intended location.

NOTE Before making any adjustments, make a short call with the default cVc HF parameters to check the performance.

4.2 Phones models and network types

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

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

Because network types and phone models vary in the Receive SCO Audio level, QTIL recommends performing initial tuning using the mobile phone and network that has the lowest Receive SCO Audio level and has no known or discovered audio issues. To check the lowest Receive SCO Audio level, monitor the **Peak SCO in** statistic. When the initial tuning is complete and validated, test other phones to verify the performance. Further fine adjustments may be 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 handsfree 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 handsfree call, it is normal to hear a buzz noise while tuning with a GSM mobile. You may need to move the GSM mobile to another location so that the buzz noise does not couple to the test hardware, especially the handsfree microphone(s) and speakers.

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

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

4.3 Using the in-vehicle audio system

When using an audio/navigation head-unit, ensure all in-vehicle audio system settings are nominal, for example:

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

In automotive OEM solutions, it may be necessary to tune one handsfree kit packaged with multiple types of audio systems. For example, a vehicle may offer a handsfree kit packaged with:

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

In these cases:

1. Tune the system that has the highest Sound Pressure Level (SPL) output at the handsfree microphone position.
2. Check the other audio systems for any problems in handsfree sound quality.

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

4.3.1 Vehicle HVAC settings

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

4.3.2 Automotive factors

Many automotive factors contribute to handsfree sound quality. When testing, consider the following conditions and any modifications made to the tests:

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

4.4 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.

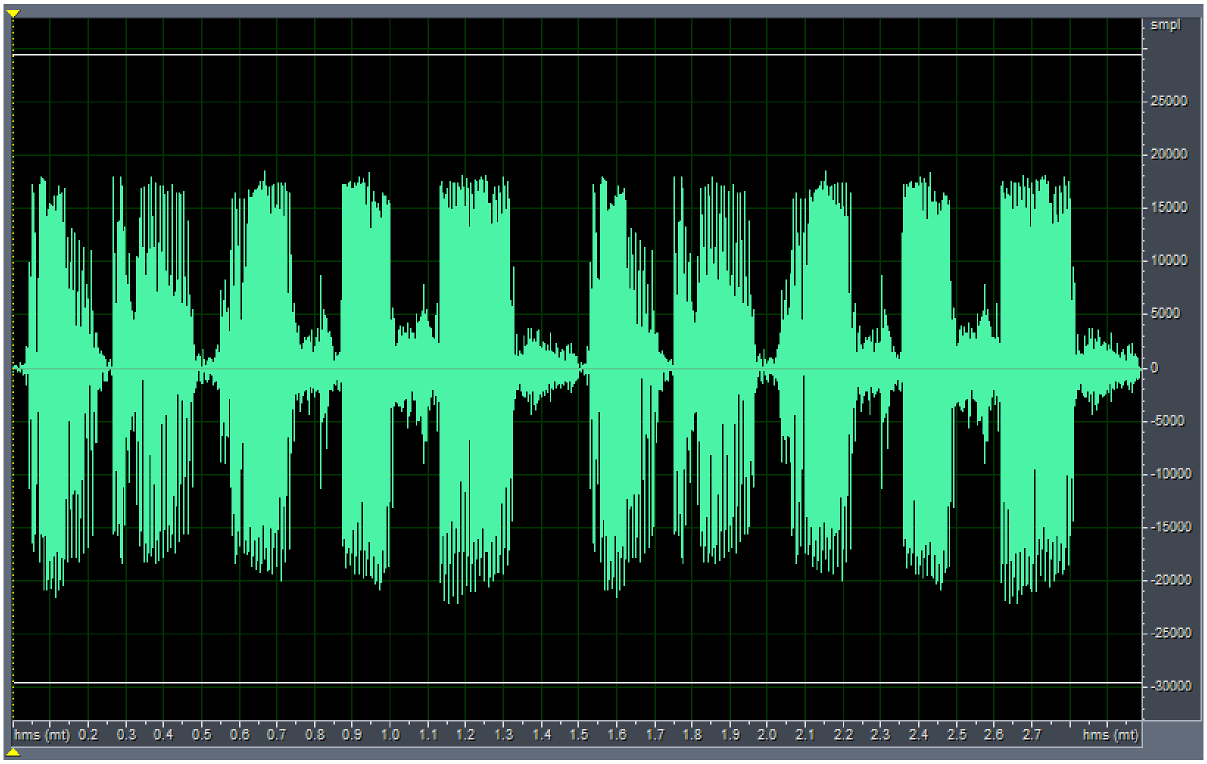


Figure 4-1 Recorded waveform of the English phrase "one two three four five"

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 handsfree microphone and receiver	❑
■ Fully charged handsfree battery or external power source	❑
■ Paired to a mobile phone	❑
GSM mobile phone with Bluetooth Headset Profile	
■ Fully charged phone battery or external power source	❑
■ Noise Suppression and Echo Cancellation disabled on the phone	❑
■ Mobile phone in close proximity to the headset	❑
■ No GSM buzz noise coupling on the headset	❑
Control of noise environment	
■ Near-end and Far-end subjects are acoustically separated	❑
■ Low network traffic test time	❑
■ Set the SPL Meter to the settings listed in 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 handsfree system	❑
Accesories	
■ Ear Plugs	❑
Personal	
■ Far-end (landside) subject	❑
■ Near-end (headset) subject	❑
Documentation	
■ *BCSW-CVC-HF-5-0-3 Parameter Manager User Guide*	❑

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 Microphone configuration

1. Measure and populate the **Mic Separation Distance** parameter as measured from the center point of the two microphones.
2. Measure and populate the **Direction of Arrival** angle parameters for both the driver and passenger.
3. If the vehicle is right-side drive, select the tick box accordingly.

7.3 Set Send Gain path

Left and Right Microphone gains should be linked. Determine if the echo path signal (loudspeaker at max volume) is louder that the speech signal at the microphone.

If echo is louder than speech:

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

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

7.4 Echo/doubletalk performance

For less echo, increase RER Power and/or decrease RER Adjustment.

For increased doubletalk, increase RER Adjustment and/or decrease RER power.

NOTE It may require several attempts to find the best ratio of RER Power/RER Adjustment to achieve optimal doubletalk and echo performance.

8 cVc tuning procedures

8.1 Objective measurement

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

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

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

8.2 Receive Path tuning

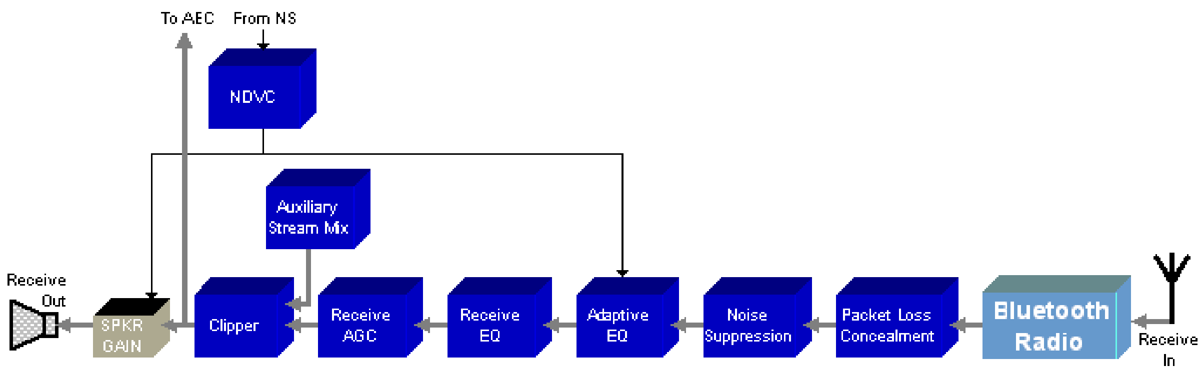


Figure 8-1 Receive Path processing blocks

NOTE Repeat all tuning measurements at least twice.

8.2.1 Load preset parameters

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

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.

NOTE Bypass the advanced processing blocks Adaptive EQ, Clipper and NDVC at this stage in the tuning process.

Alternately, if you have previously tuned the handsfree device and have saved the parameters, you can preload the saved parameters and continue with the tuning process. The loading of saved parameters is described in the Parameters Manager integrated documentation. Click **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**.

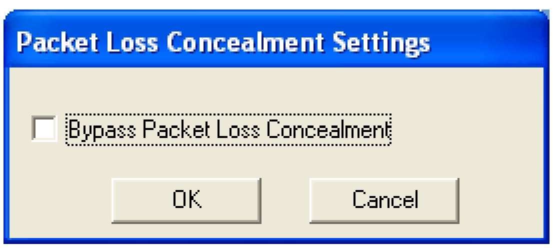


Figure 8-2 Packet Loss Concealment settings window

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.

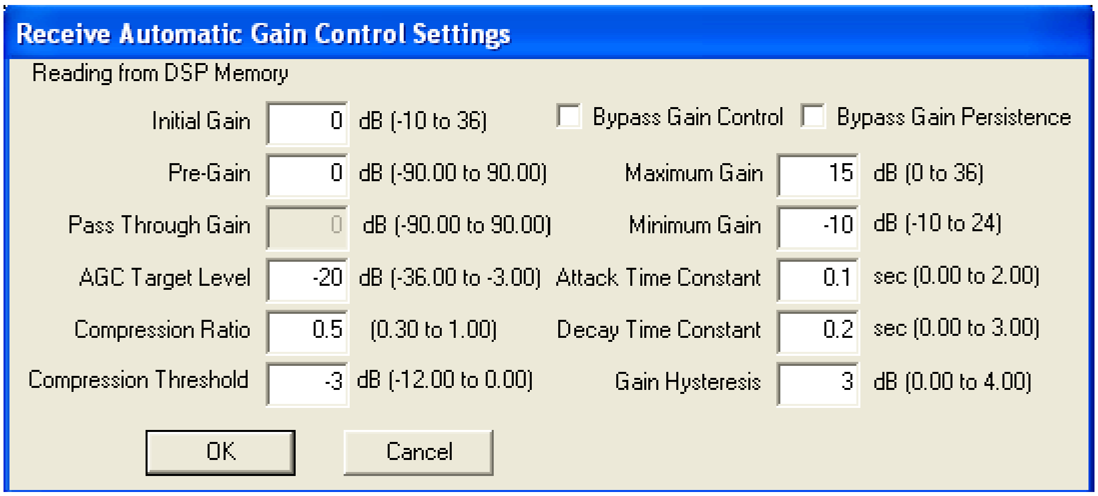


Figure 8-3 Receive Automatic Gain Control default settings

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 handsfree device supports. As the handsfree volume increases, the acoustic coupling between the receiver and the microphone increases. A large amount of acoustic coupling causes echo, which worsens when either the microphone or loudspeaker distorts.

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 handsfree microphone position).

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

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

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

To tune the **SPKR Gain**:

1. Initiate a handsfree call.
2. The far-end subject speaks the level speech phrase.
3. The near-end subject measures or listens to the receiver volume.
4. The near-end subject adjusts the loudspeaker volume by clicking the **SPKR Gain** block, selecting **Temporarily override DAC** and adjusting the **SPKR Gain** to the highest level that does not cause distortion and passes objective and subjective loudness judgment.
5. Using the Headset Configuration Tool, select the **Volume Control** tab to configure the required number of volume steps and set the maximum volume.

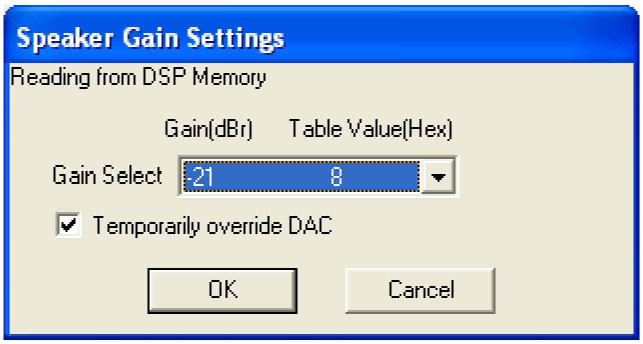


Figure 8-4 Adjusting the speaker gain

If there is distortion:

1. The near-end subject lowers the **SPKR Gain** or enables and adjusts the Clipper:
2. If the clipper is tried, follow the Clipper setup described in Clipper.
3. Optional: The near-end subject adjusts the Boost (located in the Clipper) so that the necessary receiver loudness is maintained.

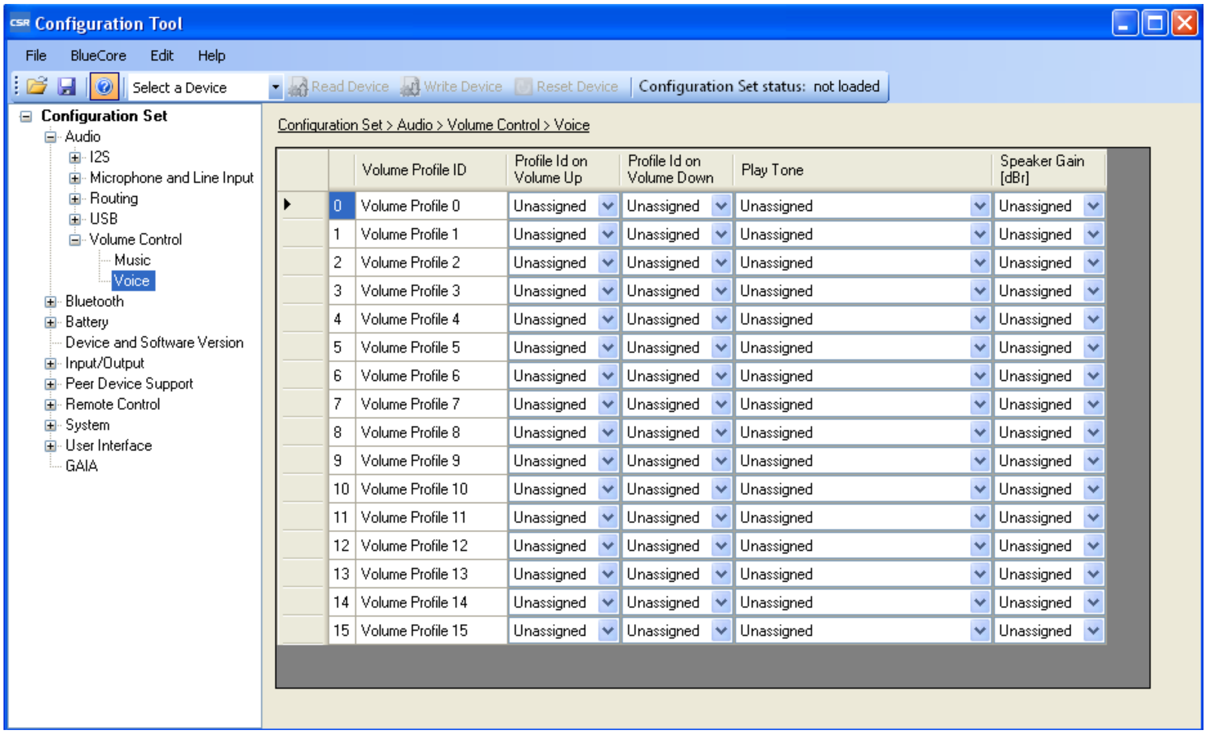


Figure 8-5 Headset configuration tool, audio gains

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

1. Initiate a handsfree call.
2. The far-end subject speaks the level speech phrase.
3. The near-end subject adjusts loudspeaker volume by clicking on the **SPKR Gain** block, selecting **Temporarily Override DAC**, and adjusting the **SPKR Gain** to the highest level that does not overload the audio input to the audio/navigation head-unit and cause distortion.
4. The far-end subject speaks the level speech phrase.
5. The near-end subject adjusts the audio/navigation head-unit's volume level to a level that does not cause distortion and passes subjective loudness judgment, no larger than "110 dB SPL Average Fast Peek" at the handsfree microphone.
6. Using the Sink Configuration Tool, select the **Audio Gains** tab to configure the **SPKR Gain [dB]** values in the VM volume table to be the highest level established in step 3. The **SPKR Gain [dB** > **]** configuration should be the same for all **HFP Levels**.

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 handsfree call.

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

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

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

High Quality Mode is 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.

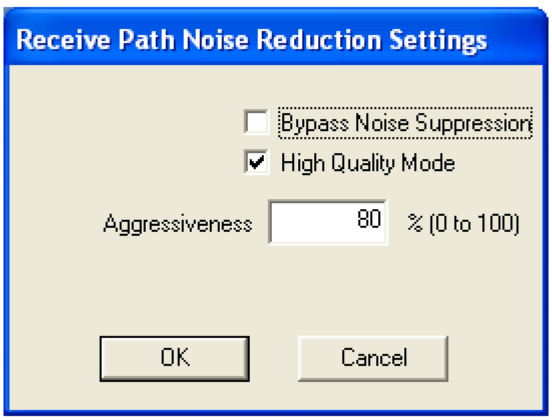


Figure 8-6 Receive Path Noise Suppression settings window

8.3 Send Path tuning

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

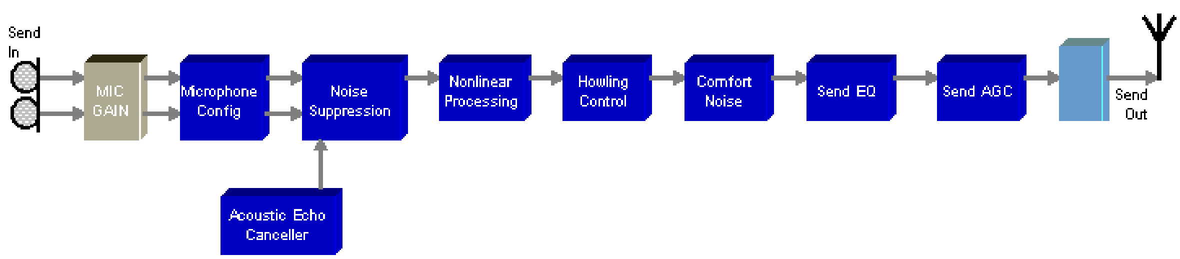


Figure 8-7 Send Path processing block diagram

8.3.1 MIC Gain

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

To tune the **MIC Gain** (HFK Mode):

1. Initiate a handsfree call.
2. The near-end subject sets the vehicle's HVAC Fan setting to either Low or Off.
3. The near-end subject places the handsfree microphone at the closest specified operating distance from the near-end subject.
4. If tuning a Bluetooth handsfree kit whose volume is controlled by the mobile phone, set the phone's volume to maximum. If tuning a Bluetooth handsfree kit integrated with the car's stereo system, set the audio head-unit's volume to maximum.
5. The far-end subject speaks the speech level phrase while the near-end subject monitors the MIC GAIN **Peak** statistics.
6. The near-end subject speaks the level speech phrase at approximately "90 dB SPL Average (C) Weighted Fast" measured 25 mm from the speaker's mouth and monitors the MIC GAIN **Peak** statistics.
7. Is the MIC GAIN **Peak** statistics larger during the far-end speech (echo) or the near-end speech? If the far-end speech (echo) is greater continue. If not, go to step 10.
8. The far-end subject speaks the speech level phrase while the near-end subject monitors the MIC GAIN **Peak** statistics.
9. The near-end subject adjusts the HFK Mode **MIC GAIN** so that the MIC GAIN **Peak** statistics reads no more than -6 dBFS. End **MIC GAIN** tuning for far-end speech (echo) is larger than near-end speech.
10. The near-end subject speaks the speech level phrase while the near-end subject monitors the MIC GAIN **Peak** statistics.
11. The near-end subject adjusts the HFK Mode **MIC GAIN** so that the MIC GAIN **Peak** statistics reads no more than -9 dBFS (-15 to -9 dBFS). End **MIC GAIN** tuning for near-end speech is larger than far-end speech (echo).

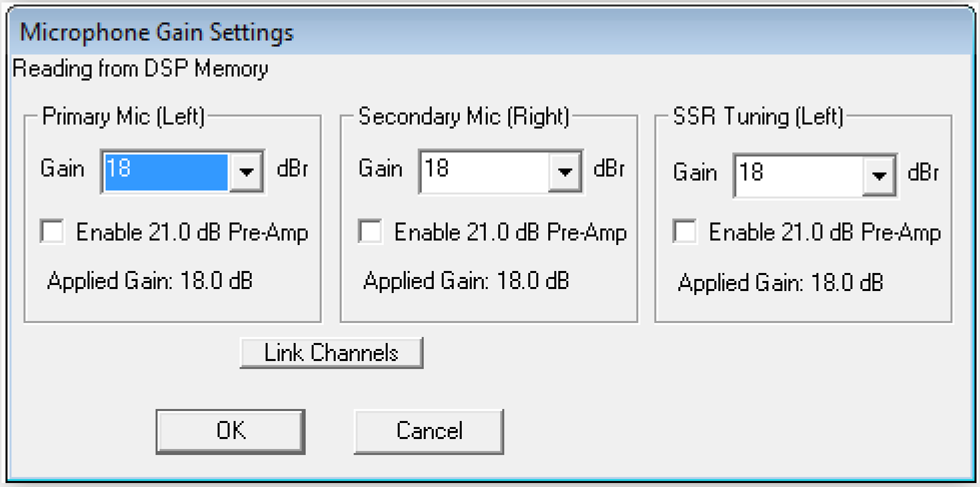


Figure 8-8 Microphone Gain settings window

8.3.2 Microphone Configuration

The Microphone Configuration tells the 2mic cVc handsfree algorithm how the microphones are set up in the vehicle

To tune the Microphone Configuration

1. Initiate a handsfree call.
2. Measure the distance between the center points of the left and right microphone and place this value in the **Mic Separation Distance** parameter.
3. **Driver-side DOA:** Measure the angle from drivers mouth to the center point of microphones and place this angle in the **Driver-side DOA** parameter.
4. **Passenger-side DOA:** Measure the angle from front passengers mouth to the center point of microphones and place this angle in the **Driver-side DOA** parameter.
5. **Right Side Drive:** Check this box if vehicle is right hand drive.
6. **Mic Calibration:** These parameters help the 2-mic processing adapt quicker at the beginning of a call if the left and right microphones are not matched in sensitivity. The **Max Compensation** is the maximum gain that can be applied to the right microphone. The **Malfunction Detection Power** is the threshold which if exceeded assumes one of the microphones has malfunctioned and disables the 2-mic portion of the algorithm. Microphone calibration and gain persist can be disabled by selecting **Bypass Mic Calibration**.

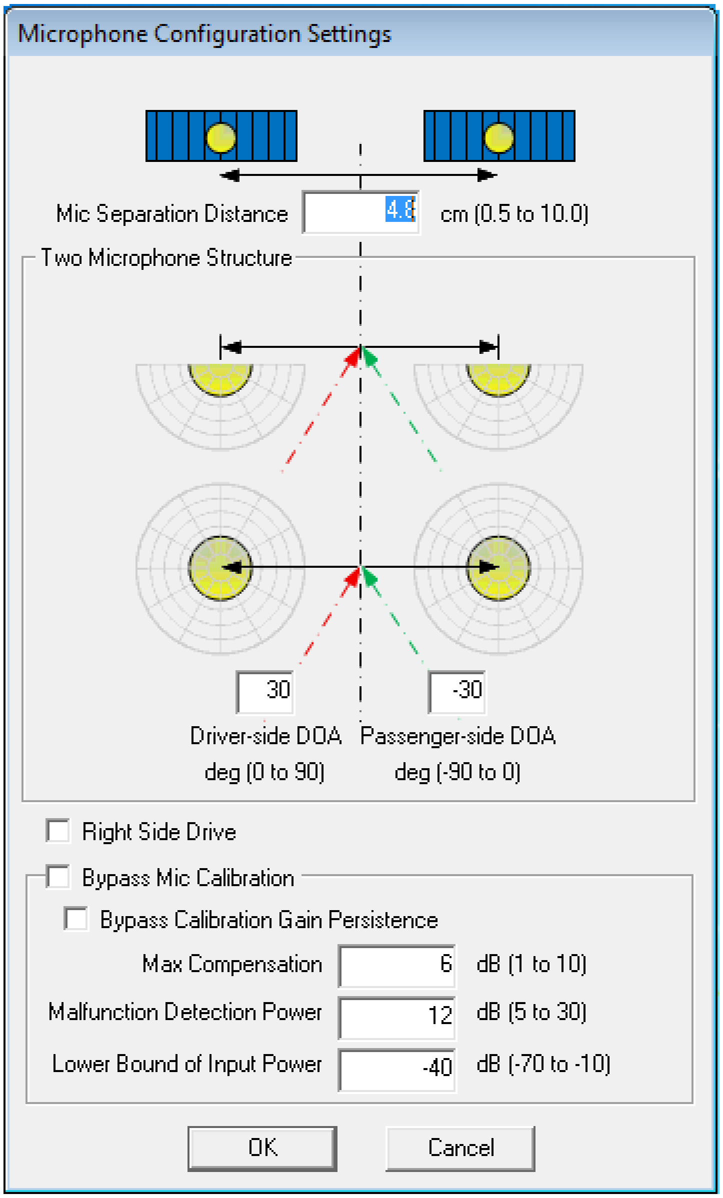


Figure 8-9 Microphone configuration settings

8.3.3 Send AGC

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

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

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

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

1. Initiate a handsfree call.
2. The near-end subject places the primary handsfree microphone at the closest specified operating distance from the near-end subject.
3. The near-end subject sets the vehicle's HVAC Fan setting to either low or off.
4. Next, the near-end subject speaks the level speech phrase at approximately 90 dB SPL Average (C) Weighted Fast, measured 25 mm from the speaker's mouth.
5. The near-end subject adjusts the **AGC Target Leve**l 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.
6. The **Maximum Gain** that can be applied to the signal can also be limited.
7. The **Compression Ratio** can also be specified to suit the needs of the application.
8. Make sure that the far-end subject never hears clipped or saturated speech.

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

1. Initiate a handsfree call.
2. The near-end subject sets the vehicle's HVAC Fan setting to either low or off.
3. The near-end subject places the handsfree microphone at the closest specified operating distance from the near-end subject.
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. The near-end subject monitors the Send In **Peak** MIC statistic.
6. The near-end subject adjusts the **Pre-Gain** > **or** > **Initial** > **Gain** so that the Send In **Peak** MIC statistic added to the **Pre-Gain or Initial Gain** yields levels between (-15 to -9 d BFS).
7. The near-end subject continues to speak the level speech phrase.
8. The near-end subject adjusts the **AGC Target Level** to achieve the necessary listening level at the far-end. Ensure the speech is not clipping by monitoring the Send Out **Peak** statistic, avoid saturation. The Send **AGC Target Level** is limited to -3 dB, allowing for overshoot.
9. The **Maximum Gain** that can be applied to the signal can also be limited.
10. The **Compression Ratio** can also be specified to suit the needs of the application.
11. Ensure that the far-end subject never hears clipped or saturated speech.

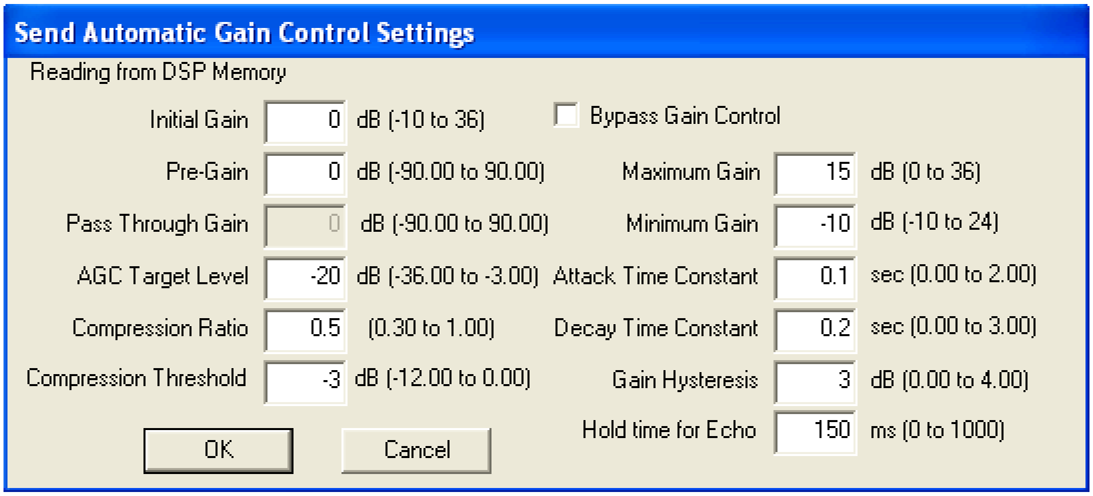


Figure 8-10 Send Automatic Gain Control settings window

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.

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

8.3.4 Send Noise Suppression (includes Two Mic noise reduction)

The **Send Path Noise Reduction Setting** interface controls both the noise suppression and **Two Mic Processing** parameters.

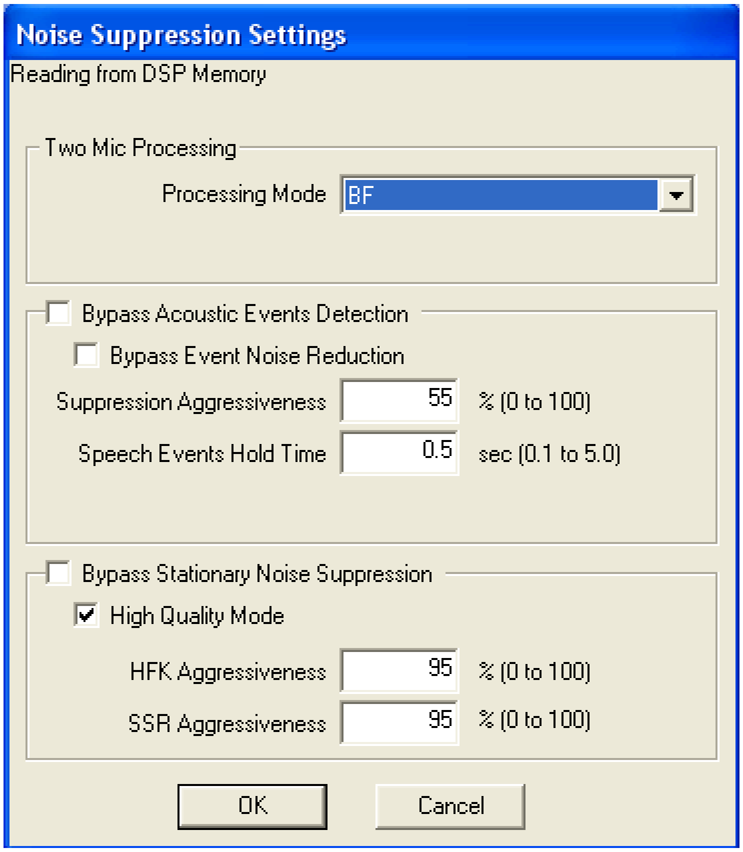


Figure 8-11 Send Path Noise Suppression settings window

The primary use of the **Two Mic Processing** and **Acoustic Events Detection** portions of the noise suppression is to remove non-stationary noise. By default they are enabled and require no additional tuning.

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

To tune for Noise Suppression:

1. Initiate a handsfree call.
2. The near-end subject introduces background noise into the testing environment.
3. The near-end subject continually repeats a test phrase or a normal conversational phrase.

4. Under different noise conditions, the far-end subjectively evaluates the noise level and its quality with and without near-end speech.
5. Increase the **HFK Aggressiveness** for more noise suppression (at the cost of voice quality). Decrease the **HFK Aggressiveness** for less noise suppression.

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

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

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

8.3.5 Acoustic Echo Canceller

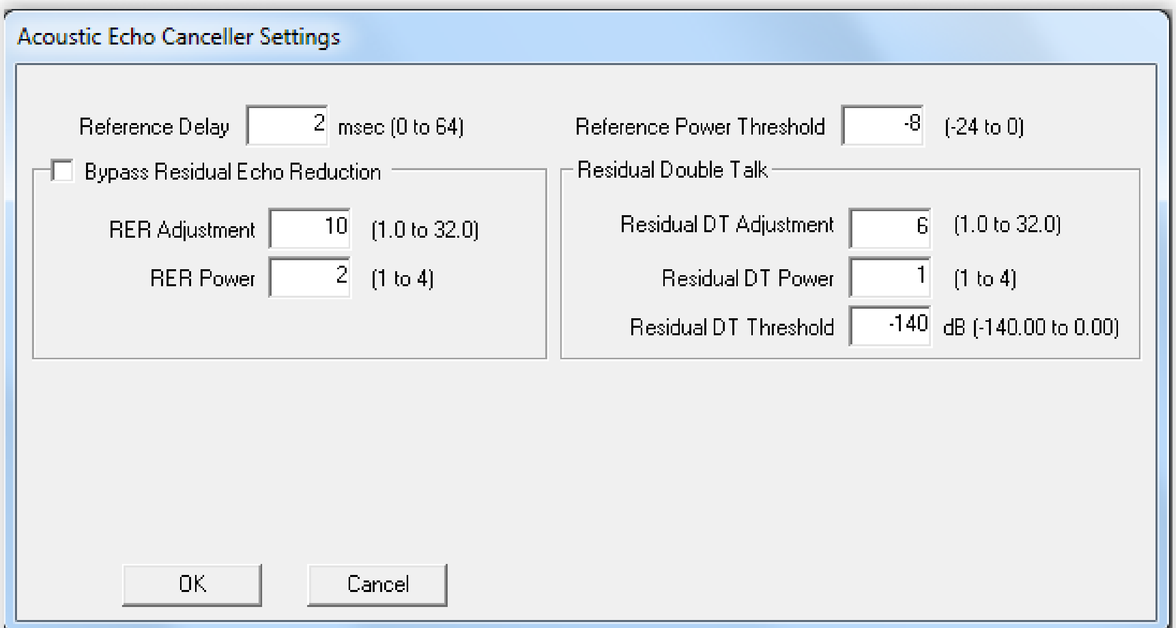


Figure 8-12 Acoustic Echo Cancellation settings window

Table 8-1 Acoustic Echo Cancellation settings field description

Settings	Description
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
Reference Delay	The **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.

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

Settings	Description
Reference Power Threshold	The **Reference Power Threshold** is the level in which the receive energy must exceed for the AEC to adapt. Increasing this value will help keep the AEC from diverging when only noise is present on the receive path for long periods of time.
RER Adjustment	The **RER Adjustment** controls the amount of attenuation when only receive speech is present. Increasing this value can degrade echo cancellation performance.
RER Power	The **RER Power** controls the number of times the RER attenuation is applied when only receive speech is present. Increasing this parameter raises the number of times the RER attenuation is applied.
RER DT Adjustment:	The **RER DT Adjustment** controls the amount of attenuation during Double Talk. Increasing this value can degrade echo cancellation performance.
RER DT Power:	The **RER DT Power** controls the number of times the RER attenuation is applied during Double Talk. Increasing this parameter raises the number of times the RER attenuation is applied.
Residual DT Threshold	The RER DT Threshold controls when to bypass the Residual DT Adjustment and Power. If the noise level statistic exceeds this Residual DT Threshold value only the RER Adjust and Power values is used.

To Tune the Acoustic Echo Canceller

1. Initiate a handsfree call, set the loudspeaker volume to maximum.
2. The far-end subject speaks short bursts of speech (for example, “one”, “two”, “hello”, “ok”, “check”, and “echo”) and checks for echo at the far-end.
3. If echo is heard on the far end, increase **RER Power** by 0.5 and recheck for echo. Increase as needed.
4. The near end speaks continuous while the far-end speaks short bursts. The far end checks for double talk attenuation.
5. To tune for attenuation during double talk, raise **RER Adjustment** by steps of 0.5 and retest. If no increase in performance is achieved, return **RER Adjustment** to default, which is a value of 1.

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

8.3.6 Non-linear processing

The Non-linear Processing block removes residual echo that the primary AEC is not able to cancel due to a distortion. It also compensates for the nonlinearities introduced in the receive path.

More often, the reference signal to the AEC is before the volume control of the speaker. Changes in the volume that introduces non-linear effects in the acoustic path is compensated for by using either the Tier 1 or Tier 2 set of heuristic non-linear processing. The Tier 1 set can be used with lower speaker volume levels and the Tier 2 set at higher speaker volume levels. In each tier, the attenuation, the bandwidths, and the number of bands can be selected to improve the compensation for the non-linearities introduced.

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

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

- Lower Bound
- Middle Bound
- Upper Bound

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

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

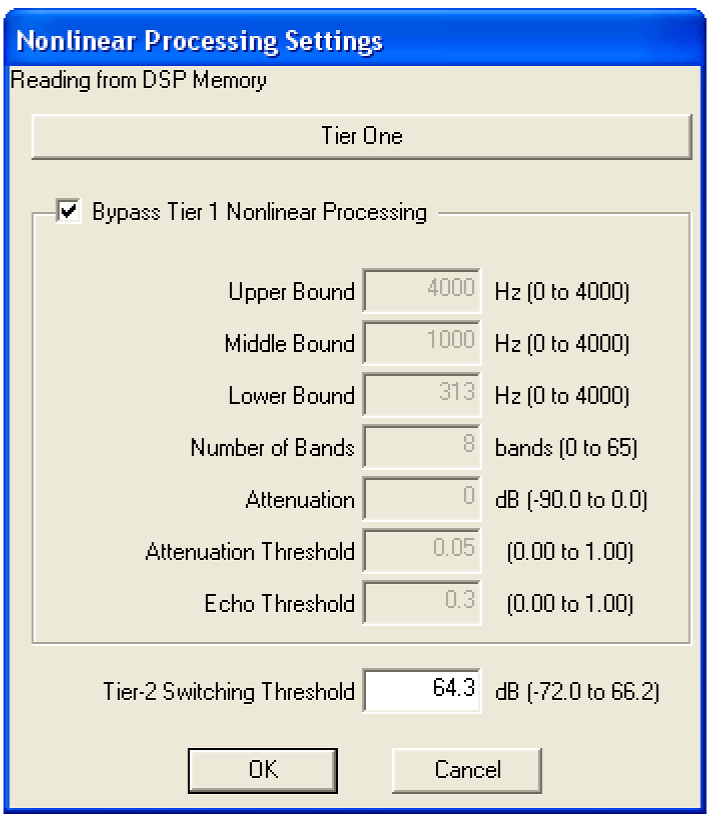


Figure 8-13 Non-linear processing settings window

8.3.7 Howling Control

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

To fine-tune the Howling Control:

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

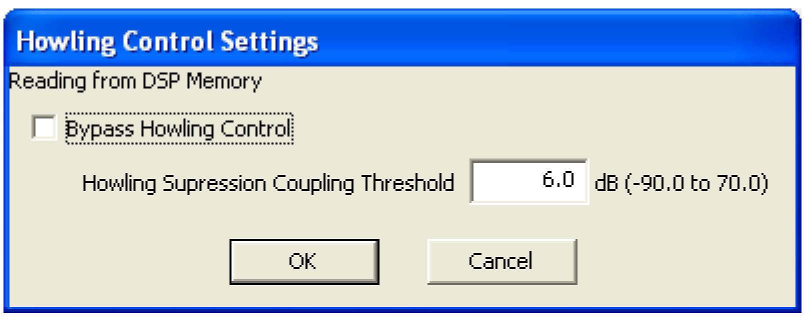


Figure 8-14 3: Howling Control settings window

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.

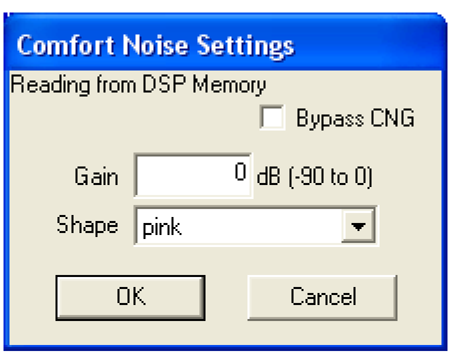


Figure 8-15 Comfort Noise settings window

To tune the Comfort Noise generator:

1. Initiate a handsfree call.
2. The near-end subject introduces steady background noise into the testing environment.
3. The near-end and far-end subjects alternate speaking numbers and letters (for example, “1”, “A”, “2”, “B”, “3”, “C”, and so on) with a 1-second pause between each number and/or letter.
4. The far-end subject listens to any fluctuations in the noise floor during the alternating speech.
5. If fluctuations are heard, increase or decrease the amount of comfort noise.
6. The near-end subject bypasses the Comfort Noise generator.
7. The far-end subject checks for a large decrease in background noise as the Comfort Noise generator is bypassed.
8. If there is a large decrease in background noise, the near-end subject decreases the Comfort Noise generator gain.
9. Shape can be used to choose the weighting of the comfort noise spectrum

9 Fine-tuning

When the cVc handsfree receive and send paths are tuned, minor parameter changes may be necessary to reach a good performance level. Some products have unique acoustic designs or have special handsfree 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 handsfree call.
2. Adjust the phone volume to minimum.
3. The far-end subject speaks the level speech phrase.
4. The near-end subject measures or listens to the loudspeaker volume. Click 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 target level, while the slope of gain curve below the target level is unity.

9.1.3 Receive EQ

The receive path has a parametric equalizer for enhancing audio quality.

By default, the parametric equalizer is set as a high pass filter set to roll off below 125 Hz, but may be used to troubleshoot loudspeaker distortion at specific frequencies or to pass standard measurements (ITU-T). If 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-HF-5-0-3 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 handsfree call.
2. Adjust the handsfree volume to the maximum.
3. The far-end subject speaks the level speech phrase.
4. The near-end subject listens for distortion in the handsfree receiver.
5. The near-end subject lowers the **Clip Point** by -3 dB steps, until additional distortion is heard.
6. Remove the last -3 dB value of added **Clip Point** and set this as the new **Clip Point**. End tuning the clipper.

Optional Steps:

1. If the Clipper is enabled, the near-end subject can adjust the **Boost** so that the required receiver loudness is maintained. Any boost 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.

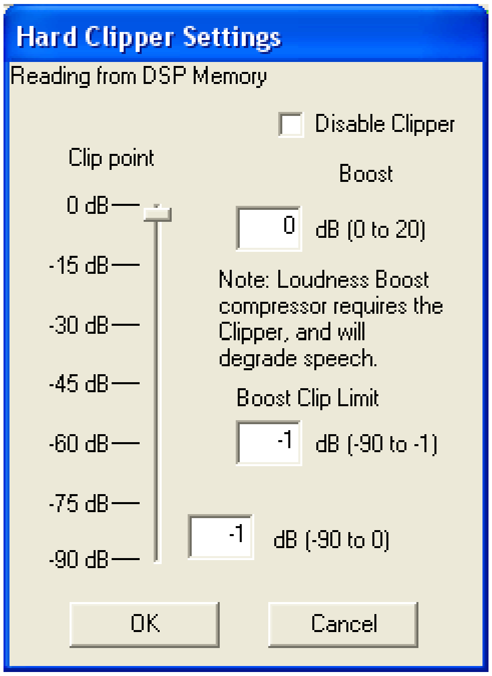


Figure 9-1 Hard Clipper setting window

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.

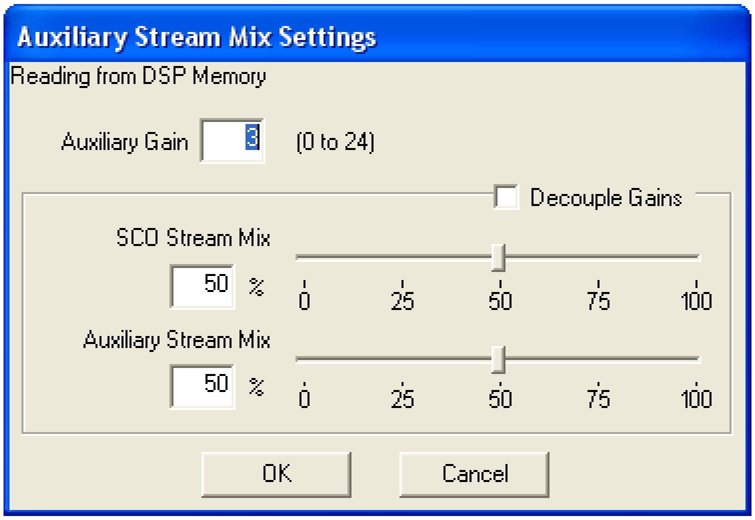


Figure 9-2 Auxiliary Stream Mix setting window

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, see the the *BCSW-CVC-HF-5-0-3 2 mic handsfreeParameter Manager User Guide* for details.

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

9.2.2 Send AGC

To fine-tune the Send AGC:

1. Adjust the **AGC Target Level** to the required value. The default is -25 dB, which provides a good dynamic range with almost full-scale value.
2. Adjust the **Minimum Gain**, which sets the low threshold level for the gain factor. The gain factor 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 target level, while the slope of gain curve below the target level is unity.

10 Advanced tuning

10.1 Noise-Dependent Volume control

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

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

NOTE Tune the NDVC after the MIC Gain.

To tune the NDVC:

1. Initiate a handsfree call.
2. Click on the NDVC processing block and remove the checkmark from the Bypass NDVC check box. This enables the NDVC.
3. Increase the noise floor inside the vehicle to the required level at which the NDVC should start to adjust the volume (for example, driving at a speed of 20 mph).
4. Monitor the Noise Level statistic and type this value in the Min Noise Level field.
5. Increase the noise floor inside the vehicle to the required maximum level at which the NDVC should remain turned on (for example, driving at a speed of 60 mph). Monitor the Noise Level statistic and place this value in the Max noise level field.
6. During the high “road” noise condition, the near-end subject determines the maximum gain that the NDVC can apply and enters this number in the Maximum NDVC Gain Limit field.
7. Set the **Hysteresis** to a value between zero and one. Higher values reduce the NDVC sensitivity when reacting to changes in the background noise and lower values increase the sensitivity. The default value is 0.75.
8. Adjust the **Increasing Noise Attack Time Constant** and **Decreasing Noise Decay Time Constant** to a required level. High time constant values cause the NDVC to react more slowly to changes in the background noise and lower values cause a quicker reaction.

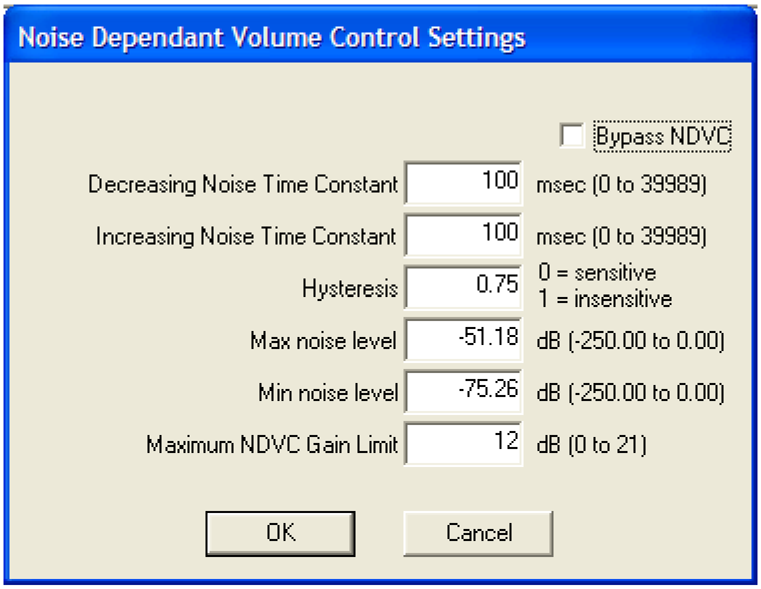
NOTE The NDVC does not adjust the **SPKR Gain** over 0 dB. For example, if the **SPKR Gain** is -12 dB and the Maximum NDVC Gain Limit is 15 dB, the NDVC does not apply more than 12 dB of gain. Under the highest noise condition, the maximum loudness is limited to **SPKR Gain** at 0 dB or the highest level defined in the Volume table.

The Total **SPKR Gain** = **SPKR Gain** + Maximum NDVC Gain Limit

The Total **SPK** > **R Gain**is important when tuning the Send Path.

The NDVC gain change is quantised 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.

Adjust the **Max noise level** and **Min noise level** for your specific handsfreedevice. Noise-Dependent Volume control shows the default settings.



10.1.1 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:

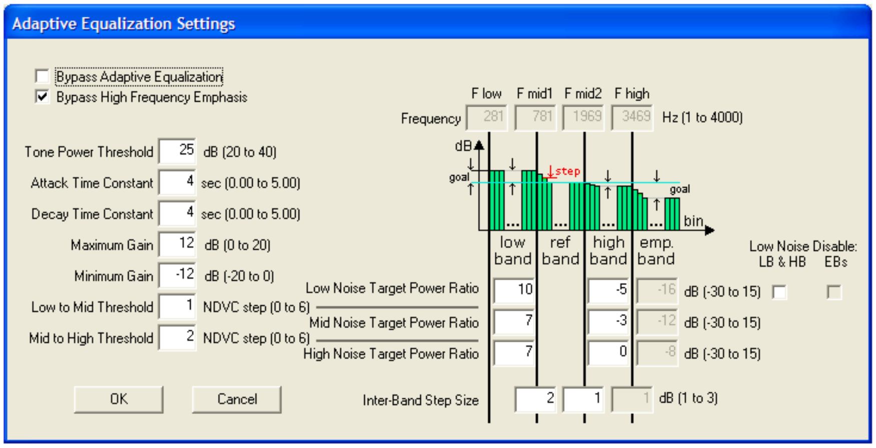
- Narrow Band plus High Frequency Emphasis: consists of AEQ (0 to ~3.5 kHz) plus the addition of an Emphasis Band (~3.5 to 4 kHz).
- Narrow Band plus Frequency Expansion: consists of an AEQ (0 to ~3.5 kHz) plus the addition of Expanded Bands (~3.5 to ~6.2 kHz).
- Wide Band: consists of an AEQ (0 to 8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

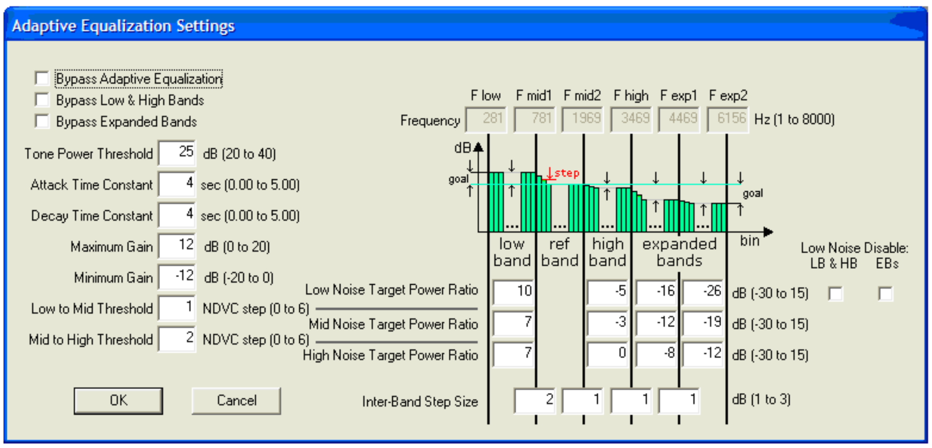
To maintain a consistent amplitude level, the Adaptive Equalization block adaptively increases the high frequencies relative to the middle frequencies while reducing low frequencies accordingly.

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

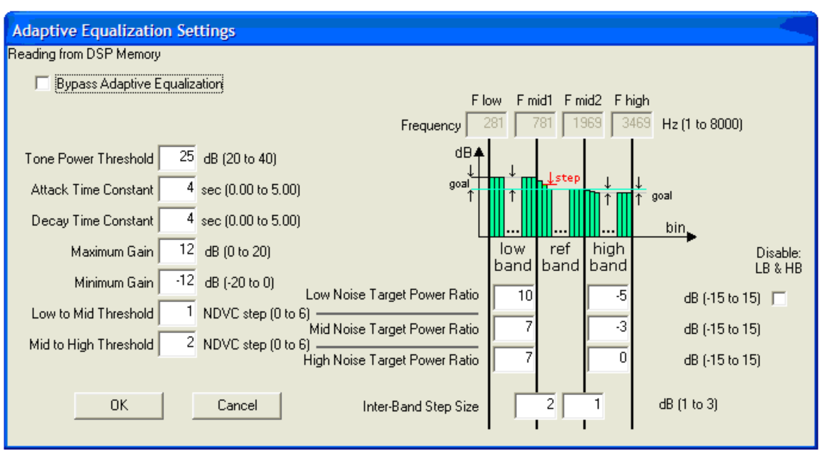
NOTE For the handsfree device to benefit from this feature, the loudspeaker must provide adequate fidelity delivered to the ear of the user and the NDVC has been enabled and tuned.



Narrow Band AEQ plus High Frequency Enhancement



Narrow Band AEQ plus Frequency Expansion



Wide Band AEQ

Figure 10-1 Adaptive Equalizer

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.

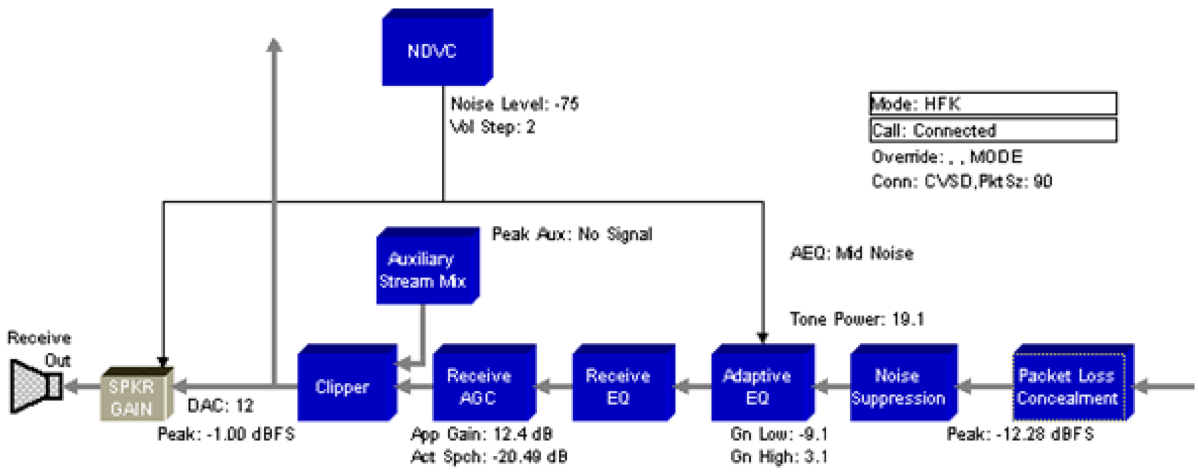


Figure 10-2 Adaptive Equalizer switching to mid-noise tier

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.

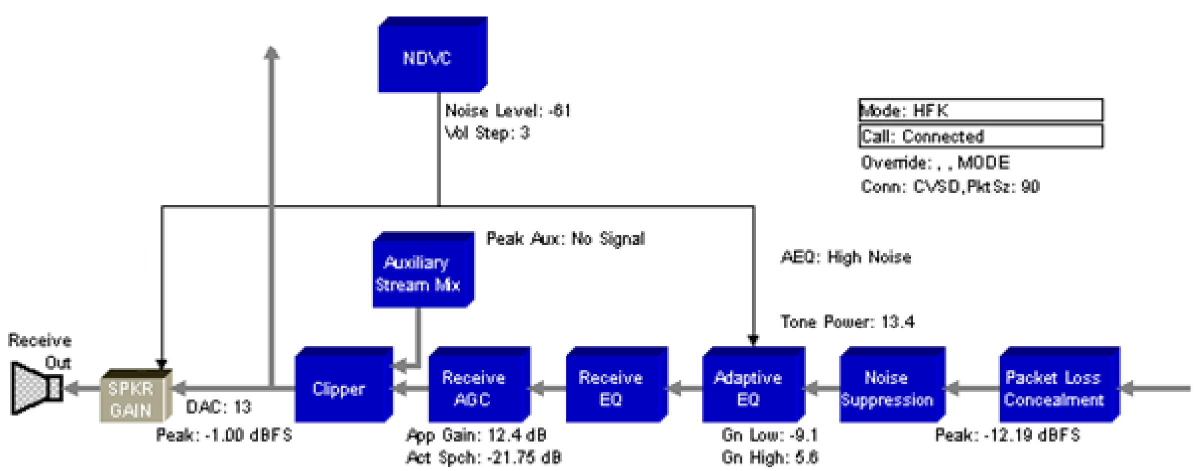


Figure 10-3 Adaptive equalizer switching to high noise ties

To tune the AEQ:

1. Initiate a handsfree 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.

10.1.2 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.

10.1.3 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 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.

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 the value of **Vol. Step** .

To tune the High Frequency Emphasis Portion of AEQ:

1. Initiate a handsfree 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 handsfree 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.

Document references

Document	Reference
BCSW-CVC-HF-5-0-2 2M-HF Parameter Manager User Guide	80-CT413-1 / CS-00309823-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