Qualcomm Technologies International, Ltd.

# BCSW-CVC-HS-6-0-2 2M-HS Parameter Manager

## User Guide

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# Revision history

| Revision | Date | Description |
|---|---|---|
| 1 | December 13 | Original publication of this document. Alternate document number: CS-00309818-UG |
| 2 | December 13 | Editorial updates |
| 3 | March 14 | Editorial updates |
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| 5 | July 14 | Updated for ADK 3.5 |
| 6 | September 16 | Updated for ADK 4.1 |
| AG | November 17 | Updated for ADK 4.3 and to reflect new document/revision numbering scheme |

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# 1 Introduction

The Parameter Manager is part of the UFE, which is a Windows-based application that enables a headset manufacturer to configure and monitor the Qualcomm Technologies International, Ltd. (QTIL) Headset DSP audio application software. The UFE is available with the appropriate version of a Headset Audio Development Kit (ADK) or as a Windows installer for a ROM-based IC.

QTIL provides Clear Voice Capture (cVc™) algorithms that create voice products. The 2-mic Headset Parameter Manager application enables engineers to customize the performance of their headset system.

This document describes the Parameter Manager application, a wizard-like graphical user interface (GUI) which operates in a Windows environment. Use the Parameter Manager application with the cVc Headset application running on a BlueCore™ digital signal processor (DSP). The cVcHeadset application provides these major modules accessible using the Parameter Manager application:

- Microphone Configuration
- Wind Noise Removal (WNR)
- Noise Suppression (NS)
- Acoustic Echo Cancellation (AEC)
- Automatic Gain Controls (AGC) and Equalizers (EQ)
- Stream Mixer
- Clipper
- Near End Audio Enhancement includes Noise Dependant Volume Control (NDVC), Adaptive EQ (AEQ), Packet Loss Concealment (PLC), and noise suppression.

This guide describes how to use the Parameter Manager application for basic tuning and monitoring activities. See the appropriate Headset Tuning Guide for information on the tuning process.

## 1.1 Supported software versions

This Parameter Managers User Guide describes the audio controls of cVc BCSW-CVC-HS-6-0-2 algorithms. The same audio controls and adjustments are used on the ICs listed in Table 1-1.

**Table 1-1 Part number matrix**

| IC supported | cVc Product Code | Version SysID | NB (8 k) | WB (16 k) | cVc license key part number |
|---|---|---|---|---|---|
| CSR8670 (Flash) | BCSW-CVC-HS-6-0-2 | B10E | Yes | Yes | BCSW-CVC-HS-2M-Fx |
| CSR8675 (Flash) | BCSW-CVC-HS-6-0-2 | B10E | Yes | Yes | BCSW-CVC-HS-2M-Fx |

**NOTE** CSR8670 and CSR8675 flash ICs support narrow band (8 KHz sample rate) using CVSD, and include wide band (16 kHz sample rate) using modified sub band coding (mSBC).

Download the UFE installer from www.csrsupport.com.

## 1.2 8th Generation new features

This section lists improvements made since the previous release (BCSW-CVC-HS-5-6-1) that improve performance and affect the tuning process.

New features and improvements include:

- cVc Generation 7 feature support
- Optimized latency using 60 sample Frame Size
- Added **Simple Speech Recognition** (SSR) functionality
- Optimized latency using 60 sample Frame Size
- ■ In the Dual Mic Signal Separation (DMSS) block, added parameter for Front Mic Bias and updated defaults
- Updated defaults in the Acoustic Echo Cancelation (AEC) block
- Updated defaults in the Comfort Noise (CNG) block, added parameter for noise **Shape**
- Updated defaults in the Auxiliary Stream Mix
- Microphone Config tool
- Reduced tuning compatibility for WNR
- Reduced tuning compatibility for Noise Suppression
- Added half duplex mode

## 1.3 Assumptions

This document assumes:

- You have built and downloaded cVc headset software to a suitable development hardware platform. See the release note and/or online help for details on development board compatibility.
- You are using the correct cVc headset software version
- You clicked the **Documents** link on Parameter Manager's opening window and read the **Quick Start Guide**.

# 2 Getting started

The basic steps for using the Parameter Manager application are:

1. Install the Qualcomm® Technologies International Ltd. (QTIL) Audio Development Kit or the UFE Installer from www.csrsupport.com. Run the Universal Front End (UFE) application.
2. Use the **Quick Start** link in the UFE Documents Section page that opens when UFE application is accessed to familiarize yourself with the Parameter manager application.
3. Access the Headset Parameter Manager application.
4. Connect the Parameter Manager application using an active SPI.
5. Enter the security key if necessary.
6. Pair and connect the hands-free device (typically a Bluetooth headset) with a Bluetooth device (usually a Bluetooth phone) as an audio gateway.
7. Use parameters and metrics information for tuning and/or monitoring.

## 2.1 Installing the stand-alone universal front-end application

QTIL has designated a location for the download of the ADK or UFE Installer. Ensure that location is accessible.

### 2.1.1 Installing a Headset UFE Installer for ROM ICs

By default the installer creates a subdirectory on the program files of the PC: `C:\Program Files\CSR\<Installer Name>`

Select a corresponding **Start** menu folder and desktop icon during the installation process: `Start -> All Programs -> <Installer Name>`

### 2.1.2 Installing a Headset UFE 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**

## 2.2 Accessing the Universal Front End application

The UFE application is the main application, which contains these Parameter Manager applications:

- 1-mic Headset
- 2-mic Headset
- 1-mic Handsfree
- 2-mic Handsfree
- Low Latency 1-mic Headset
- Low Latency 2-mic Headset
- Music Manager

Access this application from the Windows **Start** menu.

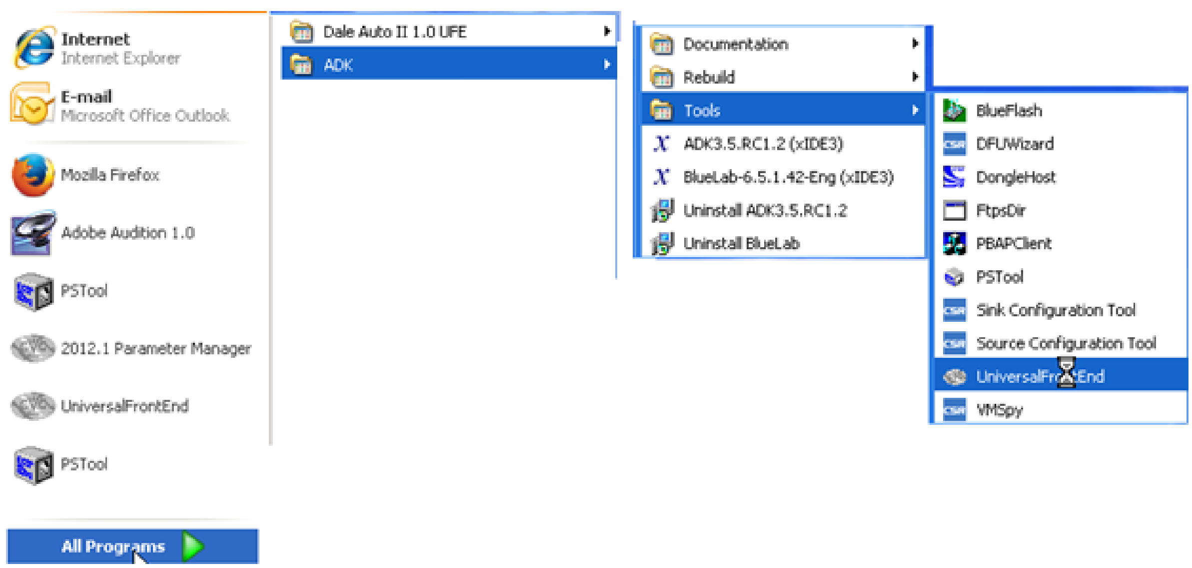



**Figure 2-1 Accessing UFE from Windows Start menu**

Click **CSR CSR86xx Parameter Manager** to open an HTML page.

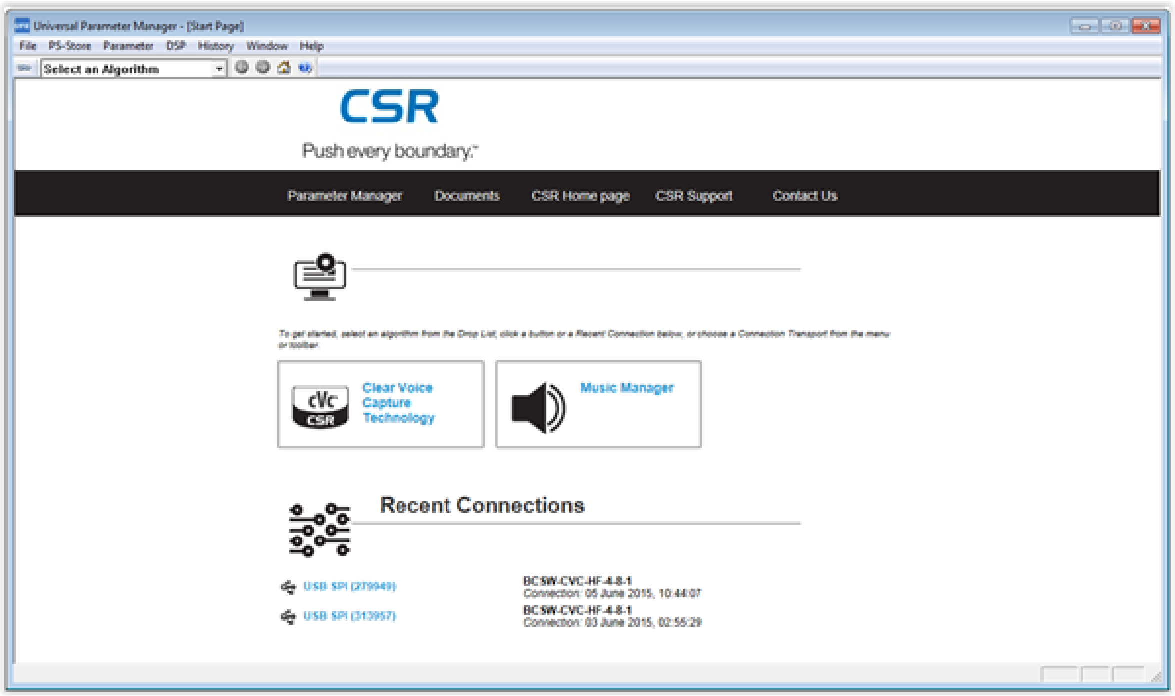



**Figure 2-2 UFE application opening HTML showing Quick Start link**

## 2.3 Viewing the UFE Quick Start

1. Click the **Documents** link on the opening HTML page then select **Quick Start**.

2. Click the  (Home icon) to return to the opening HTML page.

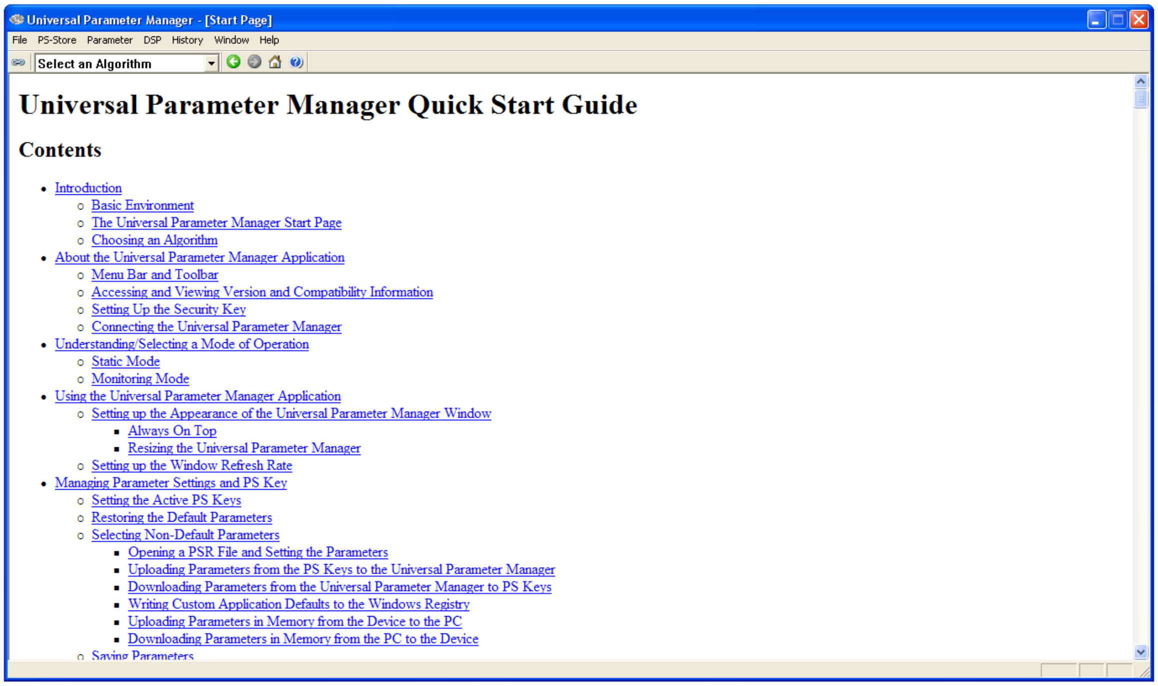



**Figure 2-3 UFE Quick Start embedded documentation**

## 2.4 Accessing the Headset Parameter Manager

To access the Headset Parameter Manager from the UFE Opening HTML page, select the **Select an Algorithm** from the drop-down list in the menu bar.

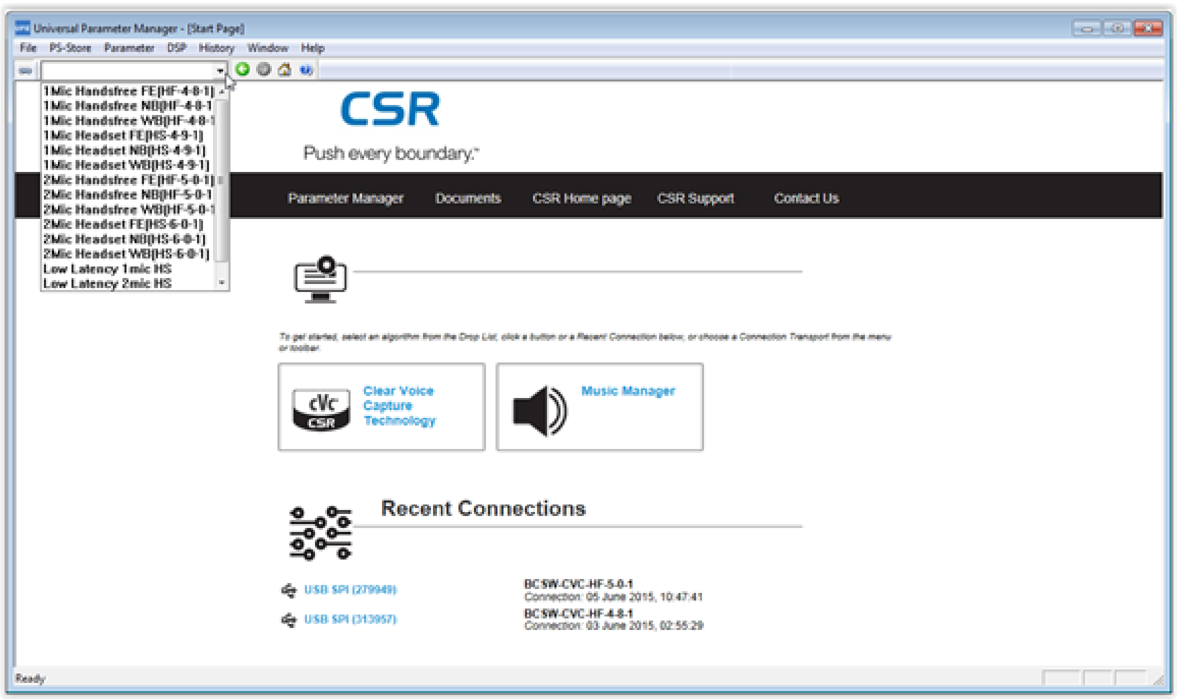



**Figure 2-4 UFE application showing Select an Algorithm drop-down list (ADK)**

The drop-down list contains the following options for the QCC3xxx:

- 1Mic Gaming Headset
- 1Mic Handsfree FE(HF-4-8-2)
- 1Mic Handsfree NB(HF-4-8-2)
- 1Mic Handsfree WB(HF-4-8-2)
- 1Mic Headset FE(HS-4-9-2)
- 1Mic Headset NB(HS-4-9-2)
- 1Mic Headset WB(HS-4-9-2)
- 2Mic Handsfree FE (HF5-0-3)
- 2Mic Handsfree NB (HF5-0-3)
- 2Mic Handsfree WB (HF-5-0-3)
- 2Mic Headset FE(HS-6-0-2)
- 2Mic Headset NBHS-6-0-2)
- 2Mic Headset WB(HS-6-0-2)

- Low Latency 1mic HS
- Low Latency 2mic HS
- Music Manager 44.1kHz
- Music Manager 48kHz

To select the 2-mic Headset Parameter Manager application, click **2 Mic-Headset NB (HS-6-0-2)** algorithm from the drop-down list.

## 2.5 Connecting Parameter Manager via SPI

See the **Quick Start** for descriptions on how to connect the Parameter Manager via a Serial Peripheral Interface (SPI) and to set up the security key for the algorithms.

## 2.6 Setting up a security key

A security key protects the cVc library. When the ADK application is used, the BCSW-CVC-HS-6-0-2 mutes the audio until a valid security key is stored in the appropriate PS Key location. Contact a QTIL sales representative to obtain valid keys for use in production devices. The CSR86xx ROM-based chips do not require a security key

**NOTE** NOTE: The cVc DSP software requires a valid security key. When a valid security key is not available, the system audio mutes immediately.

For temporary license keys, see *Enabling cVc for Headset ADK*s for instructions on how to activate the cVc algorithms for development purposes. When the Mono Headset ADK has been installed navigate from the Windows Start menu to: All Programs\<ADK Name>\Documentation\Support Documentation\ for this document. This opens the Support Document Index page. Click the link for the required document that accompanies the ADK.

# 3 Headset Parameter Manager application

The BCSW-CVC-HS-6-0-2 B10E represents the 2-mic Headset Parameter Manager. Table 3.1, Table 3.2 and Table 3.3 describe the gain settings at various points along with available modules of the Headset system.

**Table 3-1 Gain parameters and metrics**

| Option | Description |
|---|---|
| Gains – Adjustable Tuning Parameters | |
| **MIC Gain** | Analog and Digital gain stage. Determines the gain applied to an incoming microphone signal.<br>See MIC Gain. |
| **SPKR Gain** | Used during the tuning process to set the overall gain of the DAC. This value is then placed into the volume table.<br>See SPKR Gain |

**Table 3-2 Receive Path parameters and metrics**

| Option | Description |
|---|---|
| Receive Path Processing Parameters | |
| **Packet Loss Concealment** | The Packet Loss Concealment improves the receive path audio quality in the presence of bit and packet errors within the Bluetooth link by using various techniques, such as pitch-based waveform substitution.<br>The user can bypass the processing block.<br>See Packet Loss Concealment. |
| **Noise Suppression** | See the description for Send Noise Suppression.<br>This block removes unwanted noise during hands-free conversation, cleaning the audio for the near-end listener.<br>See Noise Suppression. |
| **Adaptive EQ** | The Adaptive EQ is to improve speech intelligibility and loudness in quiet and in noisy environments.<br>Three AEQ curves can be defined, and are switched based on the near-end noise.<br>See Adaptive EQ. |
| **Receive EQ** | See the description for Send EQ.<br>See Receive EQ. |

### Table 3-2 Receive Path parameters and metrics (cont.)

| Option | Description |
|---|---|
| **Receive AGC** | Automatic Gain Control (AGC) combined with audio compression is provided on the Receive channel.<br>The goal is to adjust speech to a consistent level and provide a limiter to help avoid clipping. This block reduces the sound level variances introduced by various networks and phones.<br>The AGC has 11 parameters that can be modified to obtain the required signal level with required response times.<br>See Receive Automatic Gain Control. |
| **Clipper** | Pre-clips the reference signal before the echo canceler. This feature offsets any nonlinearities that would occur after the echo canceler.<br>The developer can control the amount of clipping.<br>See Clipper. |
| **Aux** (Auxiliary) **Stream Mix** | Digital audio streams can be mixed into the receive path from other sources to enable ringtone and prompt playback, without interrupting the receive voice audio.<br>The developer can control the mix of SCO audio and Auxiliary tones.<br>See Auxiliary Stream Mix |
| **NDVC**<br>Noise-Dependent Volume Control) | The Speaker Gain can be controlled in 3 dB analog steps so that the near-end volume adjusts according to the current noise level on the near-end.<br>This block sets fixed gains according to standards and then programs a range that the volume increases further if near-end noise is present.<br>**Note:**<br>This relies on adequate speaker sensitivity designed into the overall system.<br>See Noise-Dependent Volume Control. |
| **Side Tone** | This block is provided in the digital domain to enable sidetone capability in the near-end speaker.<br>CVS automatically levels the sidetone to deliver a consistent loudness, independent of the Bluetooth volume.<br>The developer can enable or disable this function.<br>The amount of injected sidetone is programmable.<br>See Side Tone. |

### Table 3-3 Send Path parameters and metrics

| Option | Description |
|---|---|
| Send Path Processing Parameters | |
| **Wind Noise Reduction** | This block contains a wind noise reduction feature that can clean the speech in the presence of wind.<br>This block removes unwanted noise during hands-free conversation, cleaning the audio for the far end listener.<br>See Wind Noise Reduction. |

**Table 3-3 Send Path parameters and metrics (cont.)**

| Option | Description |
|---|---|
| **Microphone Configuration** | This is the main signal separation block that reduces many types of unwanted dynamic noise.<br>It uses the two microphones as inputs and results in a cleaned single channel audio input to the next noise suppression block.<br>The amount of noise suppression can be controlled to achieve optimum suppression vs. voice distortion levels for the intended application.<br>See Microphone Configuration Settings |
| **Noise Suppression** | This block reduces noise with temporal characteristics uncorrelated with speech. The amount of noise suppression can be controlled to achieve optimum suppression vs. voice distortion levels for the intended application.<br>This block removes unwanted noise during hands-free conversation, cleaning the audio for the far end listener.<br>See Noise Suppression |
| **Acoustic Echo Canceller** | This block accesses the echo cancellation settings.<br>It includes a sub-band adaptive linear filter that models the acoustic path from the receive reference point to the microphone input.<br>It also provides a nonlinear processing function that applies narrowband and wideband attenuation. This is done adaptively as a result of residual echo present after the linear filter.<br>See section Acoustic Echo Canceller. |
| **Comfort Noise** | The Comfort Noise block mitigates the noise floor modulations introduced by the residual echo reduction, generated by the AEC.<br>This feature can be parametrically bypassed and the gain control.<br>See Comfort Noise |
| **Send EQ** | Five-stage parametric and graphic equalizations are provided for both the send and receive channels, which can be independently enabled and programmed to achieve the required frequency response.<br>See Send EQ. |
| **Send AGC** | Maintains consistent listener experience regardless of the user speech level.<br>This AGC has multiple parameters that can be modified to obtain the required signal level with required response times.<br>See Send AGC. |

Figure 3-1 shows the normal full processing mode (HFK) displayed in the monitoring mode.

**NOTE** Several different Statistics are displayed and updated during an active call, such as Peak and Noise Level.

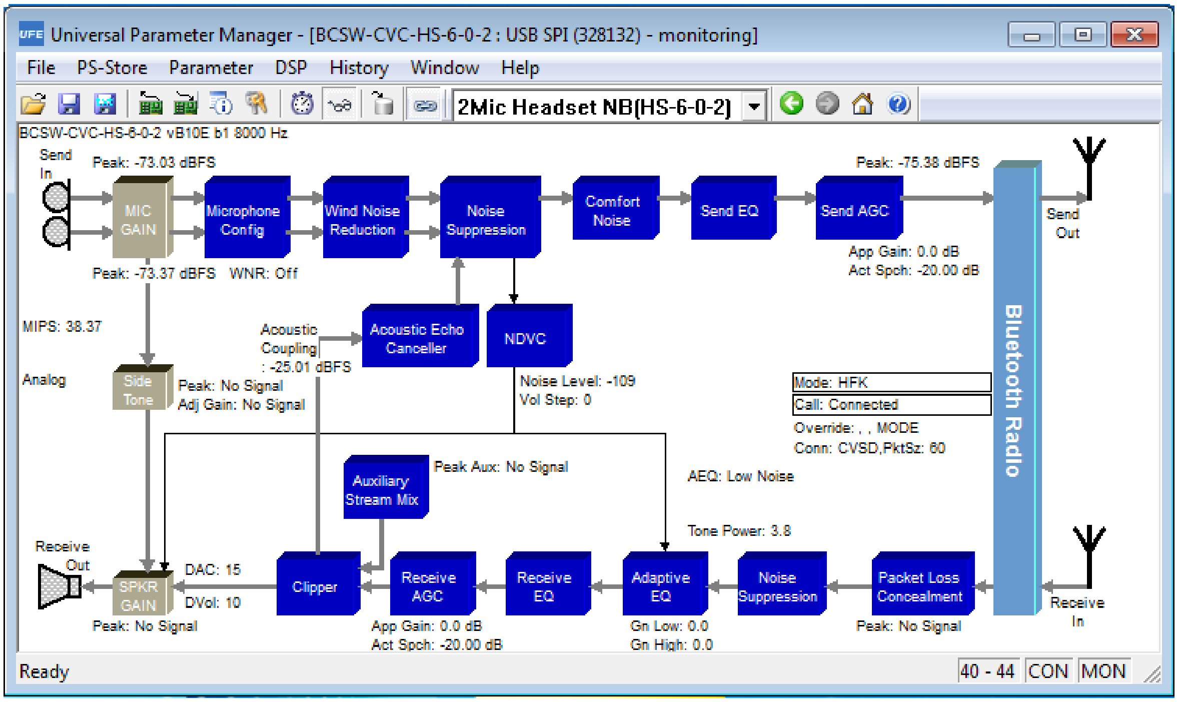



**Figure 3-1 Parameter Manager window showing all active modules**

The virtual machine (VM) controls the cVc SysMode. If the SysMode changes during the monitoring mode, the inactive areas of the Parameter Manager application are grayed out.

To manually override the current mode to assist with diagnostics, right-click the **Mode:** field and select the required override mode.

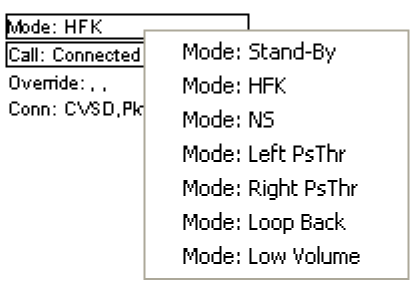



**Figure 3-2 Selecting an override mode**

Overrides modes are:

- Stand-by: Processing in standby, no audio flow
- HFK: Full processing
- NS: Noise Suppression, only NS in the send path
- Left PsThr: Left microphone pass-through
- Right PsThr: Right microphone pass-through

- Loop Back: Loop microphone to speaker, ADC to DAC
- Low Volume: Enables power savings by bypassing the AEC during low loudspeaker output volumes

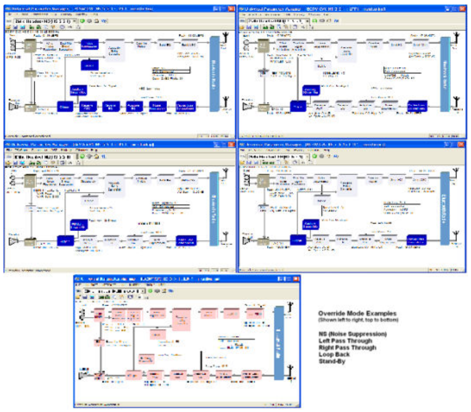



**Figure 3-3 Sample override mode examples during monitoring**

# 4 Using gains tuning controls

Parameter Manager provides access to the following gain tuning controls:

- MIC Gain
- SPKR Gain

In the **Monitoring** mode, the values that populate these screens are based on the default values stored in the DSP memory.

In the **Static** mode, the values that populate these screens are based on the default values stored in the Parameter Manager application.

## 4.1 MIC Gain

The MIC Gain option determines the gain applied to the incoming microphone signal.

To adjust the MIC Gain settings:

1. From the **Parameter Manager** window, click the **MIC GAIN** block. The **Microphone Gain Settings** window opens.

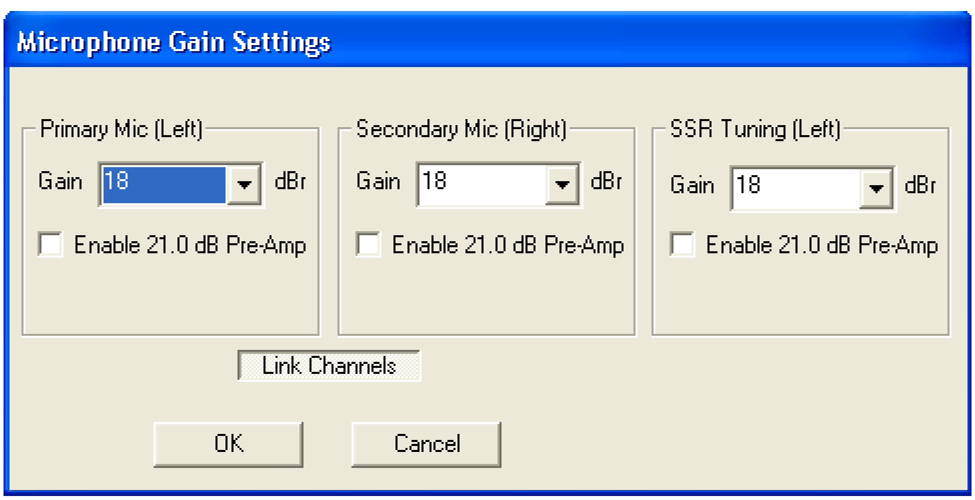



**Figure 4-1 Microphone Gain settings window**

This window controls the ADC gain. Select the required microphone gain value for each microphone.

The **Microphone Gain Settings** window displays with the current **Gain dBr** value and an **Enable 21.0 dB Pre-Amp** option. The sum of these values appears in the **Applied Gain: xx.x dB** read-only area, which the software calculates and uses.

The **Primary Mic** settings control the front (left channel) microphone, while the **Secondary Mic** settings control the rear (Right channel) microphone, SSR Tuning (left channel) settings control the SSR microphone gain.

1. Select the required settings based on the Microphone Gain settings options.

**Table 4-1 Microphone Gain settings options**

| Option | Description |
|---|---|
| **Gain** | Use the drop-down arrow to select the required gain setting for either the Primary or Secondary Microphone. These settings reflect a combination of analog gains (black text) and digital gains (red text). |
| **Enable 21.0dB Pre-Amp** | When these check boxes are unchecked (default), this means that the microphone pre-amplifier is not enabled.<br>Check the boxes to enable the microphone pre-amplifier to apply analog gain to the microphone signal. |
| **Applied Gain** | This read-only area shows a sum of the selected values, including the **Pre-Amp** check boxes, for each mode. |
| **Link Channels** | Selecting this enables the gain values of the left channel and right channel to change in tandem. |

2. When the required settings are selected, click **OK**.

## 4.2 SPKR Gain

For debugging purposes, this option enables you to temporarily adjust the Speaker Gain setting. Speaker Gain determines the gain applied at the DAC, which drives the loud speaker.

**NOTE** To set the Speaker Gain in the final configuration, edit the volume table in the Headset VM application.

To temporarily adjust the Speaker Gain Parameter Settings:

1. From the **Parameter Manager** window, click the **SPKR GAIN** block. The **Speaker Gain Settings** window opens.

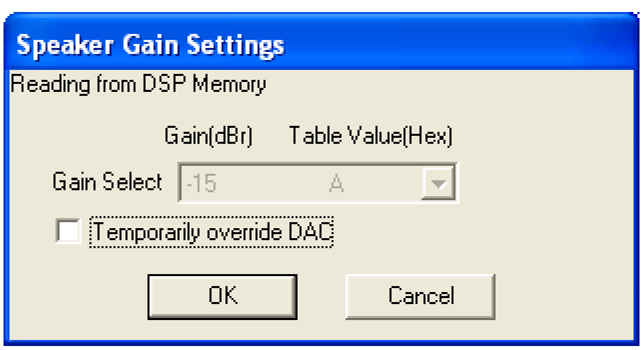



**Figure 4-2 Speaker Gain options**

2. Select the required settings based on the Speaker Gain options.

**Table 4-2 Speaker Gain options**

| Option | Description |
|---|---|
| **Gain Select** | The default Gain Select value is a read-only value based on the phone volume index. After the volume is changed, closing and re-opening the **SPKR** GAIN block updates the read-only Gain Select value.<br>To manually enter a Gain Select value:<br>Check the **Temporarily override DAC** option box.<br>Select the appropriate value in the **Gain Select** field.<br>**Note:**<br>The value in the **Gain Select** field can be overridden by a volume change. |
| **Temporarily override DAC** | Enables the **Gain Select** field to enable a manual entry.<br>**Note**:<br>The value in the **Gain Select** field can be overridden by a volume change. |

3. When the required settings are selected, click **OK.**

# 5 Using Receive Path audio tuning controls

The Parameter Manager provides access to these receive path audio tuning blocks.

- Packet Loss Concealment
- Noise Suppression
- Adaptive EQ (Adaptive Equalization that improves speech intelligibility)
- Receive EQ (Parametric Equalization for loudspeaker frequency correction)
- Receive AGC
- Clipper (includes Boost)
- Aux Stream Mix
- NDVC
- Side Tone Generation

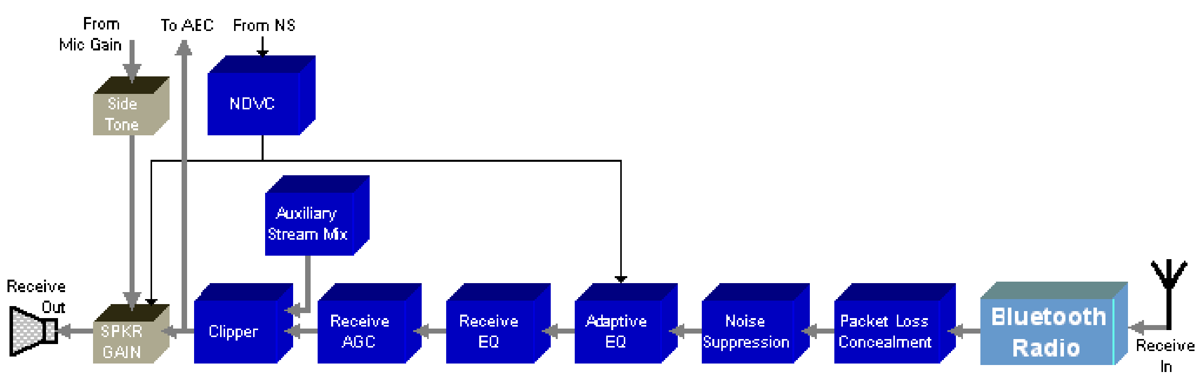



**Figure 5-1 Receive Audio Path Processing Blocks**

## 5.1 Packet Loss Concealment

The Packet Loss Concealment block improves the receive path audio quality in the presence of bit and packet errors within the Bluetooth link by using various techniques such as pitch-based waveform substitution.

To adjust the Packet Loss Concealment (PLC) settings, check the **Bypass Packet Loss Concealment** to disable the module if necessary. No tuning is provided.

**NOTE** Leave the Packet Loss Concealment enabled to achieve the best audio quality.

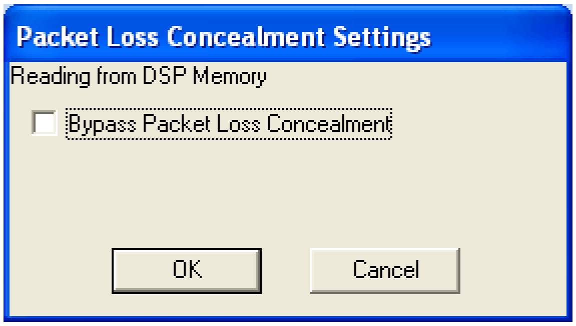



**Figure 5-2 Packet Loss Concealment settings window**

## 5.2 Noise Suppression

The **Noise Suppression** block defines the aggressiveness and quality of the noise suppression algorithm.

To set Noise Suppression options:

1. From the **Parameter Manager** window, select the **Noise Suppression** block. The **Noise Suppression Settings** window opens.

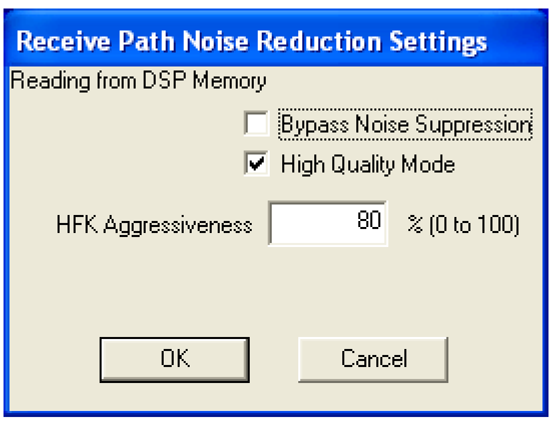



**Figure 5-3 Noise Suppression settings window**

2. Select the required settings based on the Noise Suppression settings options.

**Table 5-1 Noise Suppression settings window options**

| Option | Description |
|---|---|
| **HFK Aggressiveness** | Controls the amount of noise suppression applied to the receive signal during HFK mode. Setting this parameter to 80% yields > 10 dB of SNR improvement.<br>**Note:**<br>The receive signal has been processed by the cellular network, and transmitted over Bluetooth to avoid overprocessing the voice. Set the HFK Aggressiveness < 80%. |
| **Bypass Noise Suppression** | If checked, bypasses the NS feature reducing, processor cycles. |
| **High Quality Mode** | If checked, invokes extra algorithm processing improving the quality of the voice. |

3. Click **OK**.

## 5.3 Adaptive EQ

The adaptive equalizer improves speech intelligibility and loudness in quiet and noisy environments. Enabling the Adaptive EQ block improves the intelligibility of the receive path voice signal in the presence of near-end noise by altering the spectral shape of the receive path signal while maintaining the overall power level.

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

To maintain a consistent amplitude level, the Adaptive EQ block adaptively increases the high frequencies relative to the middle frequencies and reduces the low frequencies accordingly.

Figure 5.4 shows that in the lower right section of the diagram, the AEQ speech signal is divided into three different frequency regions. They are defined as:

- Low band: 281 Hz to 780 Hz
- Ref band: 781 Hz to 1968 Hz
- High band: 1969 Hz to 3469 Hz or 1969 Hz to 6938 Hz for wideband

The AEQ has a fixed power ratio for the ref band, and the user sets the low and high band goals to improve intelligibility and loudness.

The three bands combine to create the required spectral shape or curve. One of three power ratio curves are applied to dynamically shape to the receive speech. These curves are then switchable based on the near end noise level as measured by the NDVC.

**NOTE** For the Headset to benefit from this feature, the loudspeaker must provide adequate fidelity delivered to the user’s ear. Good examples are headsets fitted with gel ear buds that seal the ear canal. An on-ear hard plastic speaker case is not a good design for use with the AEQ.

To adjust the Adaptive EQ Settings:

1. From the **Parameter Manager** window, click the **Adaptive EQ** block. The **Adaptive Equalizer Settings** window opens.

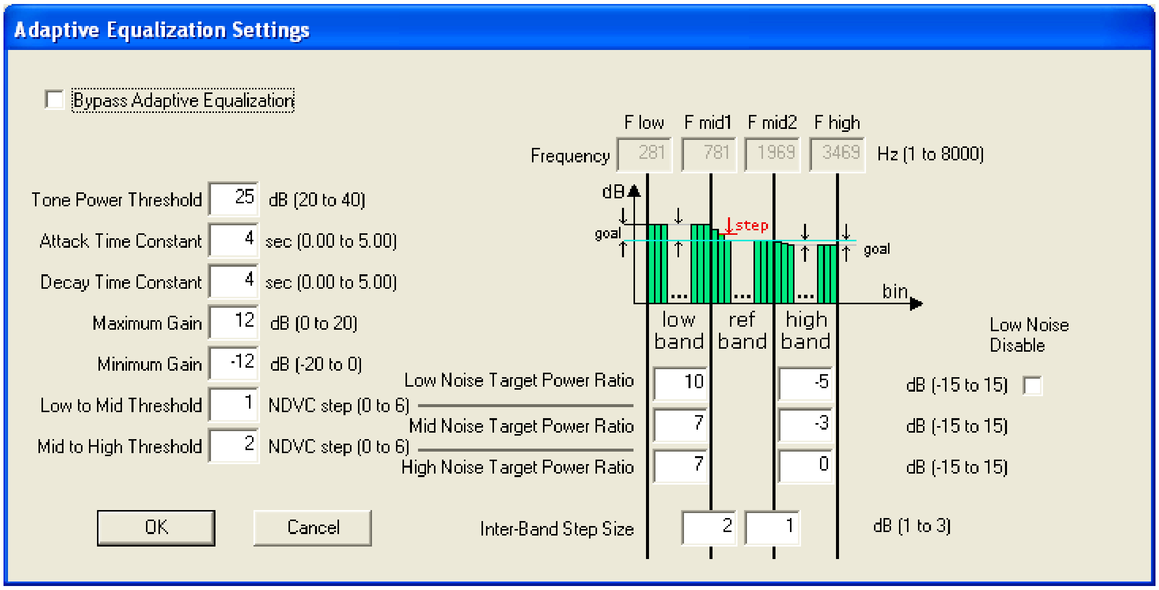



**Figure 5-4 Adaptive Equalizer Settings Window (for Wide Band)**

2. Select the required settings based on the Adaptive Equalizer options.

**Table 5-2 Adaptive Equalizer options**

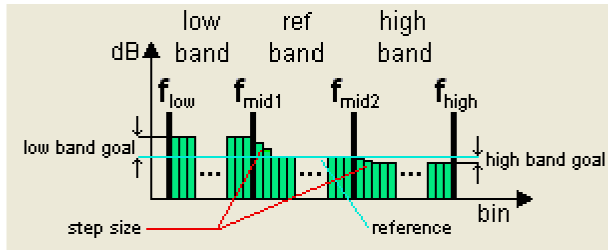
| Option | Description |
|---|---|
| **Tone Power Threshold** | If a tone appears in the receive audio path, the AEQ or the NS should not adapt during the tone. This causes an unwanted distortion in the tone.<br>Based on the Tone Power: Statistic, the user sets the threshold to identify a tone. When the Tone Power Threshold has been exceeded, the Tone Detected statistic appears.<br>Setting the threshold to low may result in some speech being falsely detected. The AEQ and NS will not be operating and speech distortion could result.<br>Setting the threshold to high may result in some tones being attenuated by the NS and shaped by the AEQ. |
| **Attack Time Constant** | Sets the adaptation rate at which the AEQ applies frequency gains. If the rate is too slow, set the Attack Time Constant lower, speeding up the frequency adaptation rate. |
| **Decay Time Constant** | Sets the adaptation rate at which the AEQ attenuates frequency gains. If the rate is too slow, set the Decay Time Constant lower, speeding up the frequency adaptation rate. |
| **Maximum Gain** | Sets the high threshold limit for the gain applied to any frequency bin. No output will have more than this gain value applied. |
| **Minimum Gain** | Sets the low threshold limit for the gain applied to any frequency bin. No output will have less than this gain value applied. |
| **Low to Mid Threshold** | Set the NDVC step at which the AEQ switches from the Low to Mid Noise Target Power Ratio curve. The switch point is based on the Vol Step statistic. When switched the AEQ statistic reads AEQ: Mid Noise<br>Note:<br>This field is only effective if the NDVC is enabled and the DAC has available headroom. |

**Table 5-2 Adaptive Equalizer options (cont.)**

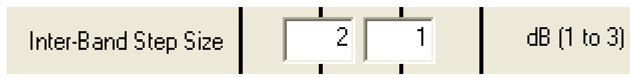
| Option | Description |
|---|---|
| **Mid to High Threshold** | Sets the NDVC step, at which the AEQ switches from the Mid to High Noise Target Power Ratio curve. The switch point is based on the Vol Step statistic. When switched the AEQ statistic reads AEQ: High Noise<br>Note:<br>This field is only effective if the NDVC is enabled and the DAC has available headroom. |
| **Low Noise Target Power Ratio** | If the AEQ is enabled, this Power Ratio curve is always used. It is independent of the NDVC. If NDVC is active, this curve is applied until the Low to Mid Threshold is reached.<br><br>The low band and high band goals are user-defined. Enter the value in dB and tune to suit the required frequency response. |
| **Mid Noise Target Power Ratio** | If the AEQ is enabled, this Power Ratio curve is applied when the Low to Mid Threshold is reached.<br><br>Two values are user-defined, the low band and high band goals. Enter the value in dB, tune to suit the required frequency response. |
| **High Noise Target Power Ratio** | If the AEQ is enabled, this Power Ratio curve is applied when the Mid to High Threshold is reached.<br><br>Two values are user-defined, the low band and high band goals. Enter the value in dB, tune to suit the required frequency response. |

**Table 5-2 Adaptive Equalizer options (cont.)**

<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Inter-Band Step Size</td><td>The change (in dB) per bin enabled at the band transition boundaries. This ensures a smooth response.




Two values are user-defined. The first value defines the step size that smooths the low to ref band transition and the second step size variable defines ref to high band transition.


</td></tr>
<tr><td>Low Noise Disable</td><td>If checked, disables the AEQ frequency shaping for the low noise condition only to preserve speech quality. If near end noise is present the AEQ can still be applied to improve intelligibility to the mid and high noise conditions.</td></tr>
</table>

3. When the required settings are selected, click **OK**.

### 5.3.1 AEQ with High Frequency Emphasis or Frequency Expansion

To complement the AEQ, add High Frequency Emphasis or Frequency Expansion to improve the intelligibility of the far-end caller. See the *QCC3xxx 2-Mic Headset Tuning Guide* for details.

This section defines the relative additions of the base AEQ.

High Frequency Emphasis can be used with any standard narrowband call, when the DAC is operating at a sample rate of 8 kHz. The High Frequency Emphasis enables the user to add in frequencies that were lost due to the band limiting filters of the cellular network and Bluetooth link. These recovered frequencies are added between 3.5 kHz to 4 kHz.

To adjust the Adaptive EQ with High Frequency Emphasis Settings:

1. Choose **2Mic Headset NB[HS-6-0-2)** from the UFE drop list or make a narrowband call, and place the UFE into monitoring mode.
2. From the **Parameter Manager** window, click the **Adaptive EQ** block. The **Adaptive EQ Settings** window opens.

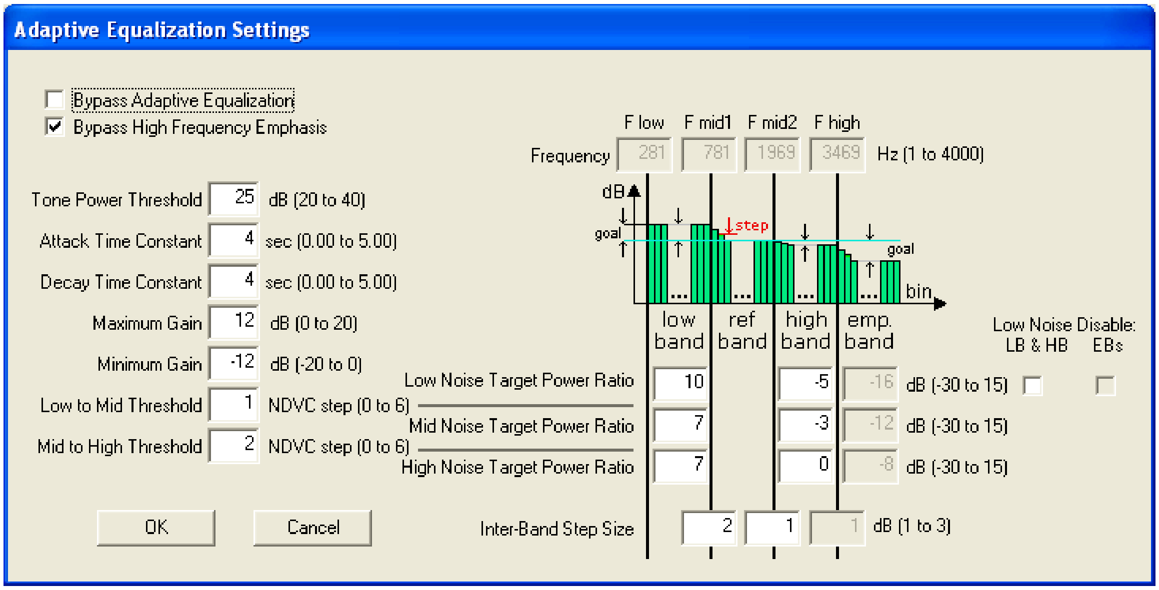



**Figure 5-5 Adaptive Equalization with High Frequency Emphasis settings window**

3. Select the required settings based on the Adaptive Equalizer with High Frequency Emphasis options.

**Table 5-3 Adaptive Equalizer with High Frequency Emphasis options**

| Option | Description |
|---|---|
| **Bypass High Frequency Emphasis** | Leave this option unchecked to use the High Frequency Emphasis under all near end noise conditions. |
| **Low Noise Disable EBs** | If checked, you can disable the emphasis band for the low noise condition only, to preserve speech quality. If near-end noise is present the emphasis band can still be applied to improve intelligibility to the mid and high noise conditions. |
| **Low Noise Target Power Ratio**<br>**Med Noise Target Power Ratio**<br>**High Noise Target Power Ratio**<br>**Inter-Band Step Size** | The Emphasis Band has its own set of Noise Target Power Ratios and Inter-Band Step Size. They operate similar to the AEQ parameters previously described but applied to the Emphasis Band. |

Frequency Expansion can be used with any standard narrowband call, but a special mode is invoked when the DAC and ADC operate at a sample rate of 16 kHz. The Frequency Expansion enables the user to add in frequencies far beyond the normal band limits caused by the cellular network and Bluetooth link. These expansion frequencies are added between 3.5 kHz to 6.5 kHz.

To Adjust the Adaptive EQ with Frequency Expansion Settings:

1. Choose **2Mic Headset Freq. Exp.** from the UFE drop list or make a narrowband call, and place the UFE into monitoring mode (assuming the VM Plug-in has been set to support the Frequency Expansion mode).
2. From the **Parameter Manager** window, click the **Adaptive EQ** block. The **Adaptive Equalizer Settings** window opens.
3. Select the required settings based on the Adaptive Equalization with High Frequency Emphasis options.

**Table 5-4 Adaptive Equalization with High Frequency Emphasis options**

| Option | Description |
|---|---|
| **Bypass Low & High Bands** | Leave this option unchecked to use the Adaptive EQ under all near-end noise conditions. |
| **Bypass Expanded Bands** | Leave this option unchecked to use the Frequency Expansion under all near-end noise conditions. |
| **Low Noise Disable LB & HB** | If checked, you can disable the low and high bands (AEQ bands) for the low noise condition only, to preserve speech quality. If near-end noise is present the low and high bands (AEQ bands) can still be applied to improve intelligibility to the mid and high noise conditions. |
| **Low Noise Disable EBs** | If checked, you can disable the expanded bands for the low noise condition only, to preserve speech quality. If near-end noise is present the expanded bands can still be applied to improve intelligibility to the mid and high noise conditions. |
| **Low Noise Target Power Ratio**<br>**Med Noise Target Power Ratio**<br>**High Noise Target Power Ratio**<br>**Inter-Band Step Size** | The Expanded Bands has their own set of Noise Target Power Ratios and Inter-Band Step Sizes. They operate similar to the AEQ parameters previously described but applied to the Expanded Bands. |

## 5.4 Receive EQ

The Receive EQ parameter is a graphical user interface designed to alter the frequency response by configuring up to five biquad filter stages to achieve the required correction response.

**NOTE** The Receive EQ settings and the Send EQ settings are identical.

To adjust the Receive EQ Settings:

1. From the **Parameter Manager** window, click the **Receive EQ** block. The **Receive Equalizer Settings** window opens.
2. The current state of the screen is the default state before presets are selected (the **Presets** field is blank by default). The Red/Green line shown in the plot area represents the equalization curve, which changes when a **Preset** is selected. The EQ interface supports multiple views of the equalizer without needing to close the window.

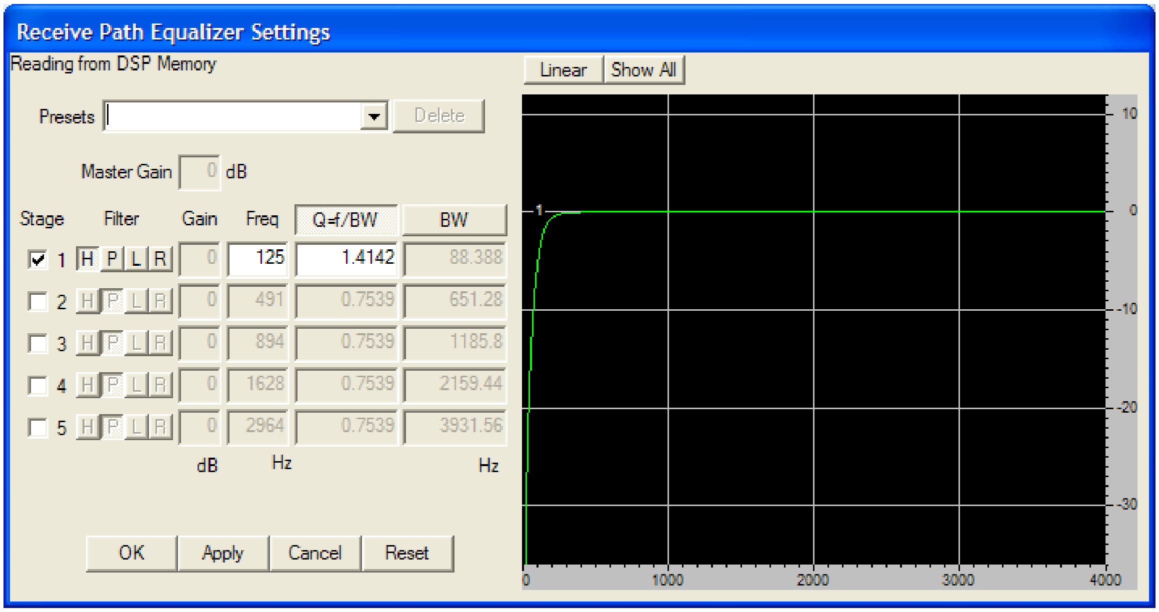



**Figure 5-6 Receive Equalizer settings window**

3. Select the required settings based on the Equalization options.

**Table 5-5 Equalization options**

| Option | Description |
|---|---|
| **Presets** | This drop-down list provides a collection of sample equalizer filters. Available options include:<br>Default<br>Low Boost<br>Mid Softener<br>Gentle High Boost<br>V curve<br>5 State Graph EQ<br>Each Preset option can be altered using the five stages listed in the Graphic EQ/Parametric EQ area on the screen. |
| **Add / Delete** | You can create an equalizer filter, once satisfied in the **Presets** type a preset name. The **Add** button highlights, if pressed the name is saved and appended to the Presets list and stored in the PC registry for later recollection.<br>Select user-defined presets from the **Presets** drop-down list. Once selected, the Delete button is activated.<br>Note:<br>If the **Delete** is pressed the Preset is deleted from the drop-down list and the registry. It is not recoverable. |
| **Linear / Log** | This option controls the appearance of the plot area on the screen.<br>Click to toggle the curve plot between the **Linear** (default) and **Log** (Logarithmic) views. |

**Table 5-5 Equalization options (cont.)**

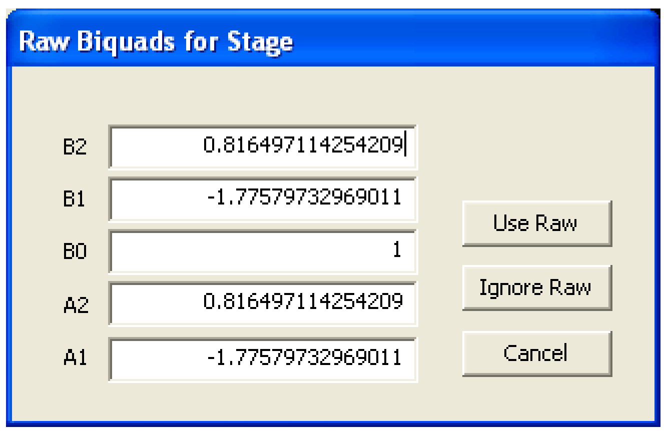
| Option | Description |
|---|---|
| **Show All** | This toggle option enables the plotting of the individual stages as well as the combined filter.<br>Selecting **Show All** displays plots of all Stages and the final filter. |
| **Master Gain** | This field is only available when one or more of the **Stage** check boxes is selected. Use this field to shift the curve up or down without changing the shape.<br>The range that can be entered in this field is -90 dB to 12 dB. |
| **Stage** | The **Stage** check box enables you to define the number of biquad stages to use in the equalizer filter configuration. |
| **Filter** | The **Filter** option enables the Stage Filter to be set to H, P, L (High Pass, Parametric, Low Pass and Raw).<br>A filter type can be set for each stage enables creation of complex curves.<br>High Pass<br>Parametric<br>Low Pass<br>Raw<br><br>If **Use** > **Raw**is selected, enter the stage coefficients directly using the GUI provided.<br>Parametric mode:<br>The fields in the Gain, Freq, Q=f/BW and BW columns are editable. Enter or select data directly from the fields, or from the plot area. Users can drag and drop the corresponding numbers on the curve to adjust the settings in the fields. |
| **Gain** | In Parametric EQ mode, the **Stage** and **Gain** fields are editable for the filter creation. **Gain** is limited between -90 dB to 12 dB. |
| **Q=f/BW** | This option is mutually exclusive to the BW option. Only one option can be selected at a time. When this option is selected, the Q=f/BW column data is editable.<br>**Q: Sharpness of curve**<br>**f: frequency**<br>**BW: Bandwidth** |

**Table 5-5 Equalization options (cont.)**

| Option | Description |
|---|---|
| **BW** | This option is mutually exclusive to the Q=f/BW option. Only one option can be selected at a time. When this option is selected, the Q=f/BW column data is editable.<br>**BW: Bandwidth** |
| **Reset** | Click this button to create a filter curve from the last saved state, such as the state last saved by clicking **OK**.<br>When this button is clicked, the settings in the stages area are updated with the last saved settings. |
| **Apply** | In the Monitoring mode, click this button to write the EQ parameters to the DSP memory where the changes take immediate effect.<br>Changes made in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory.<br>The **Apply** button can write unlimited parameters to the DSP memory without closing the **Receive Equalizer Settings** window.<br>In the Static mode, this button is not available (grayed out). |
| **OK** | In the Monitoring mode, click this button to write the EQ parameters to the DSP memory where the changes take immediate effect. The **Receive Equalizer Settings** window closes.<br>Changes made in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory.<br>In the Static mode, this button is not available (grayed out). |
| **Cancel** | Click this button to close the **Receive Equalizer Settings** window without saving any of the latest changes. |
| **Vertical and Horizontal Scale Bar User Controls** | |
| **Zoom In** | Use the mouse, on the grey Scale bar in the plot area, to select a starting zoom point.<br>Left-click and drag to the required ending zoom point. Release the mouse button and the plot area zooms to the selected area. |
| **Zoom Out** | To return to the default scale, on the grey scale bar in the plot area, double-click the left mouse button. |
| **Scale Drag** | To change the curve in the plot area, right-click, hold, and drag the mouse button within the plot area, to the required location.<br>Release the mouse button.<br>Note:<br>The horizontal scale only drags if the scale is zoomed. |

## 5.5 Receive Automatic Gain Control

To adjust Receive AGC Settings:

1. From the **Parameter Manager** window, click the **Receive AGC** block. The **Receive Automatic Gain Control Settings** window opens.

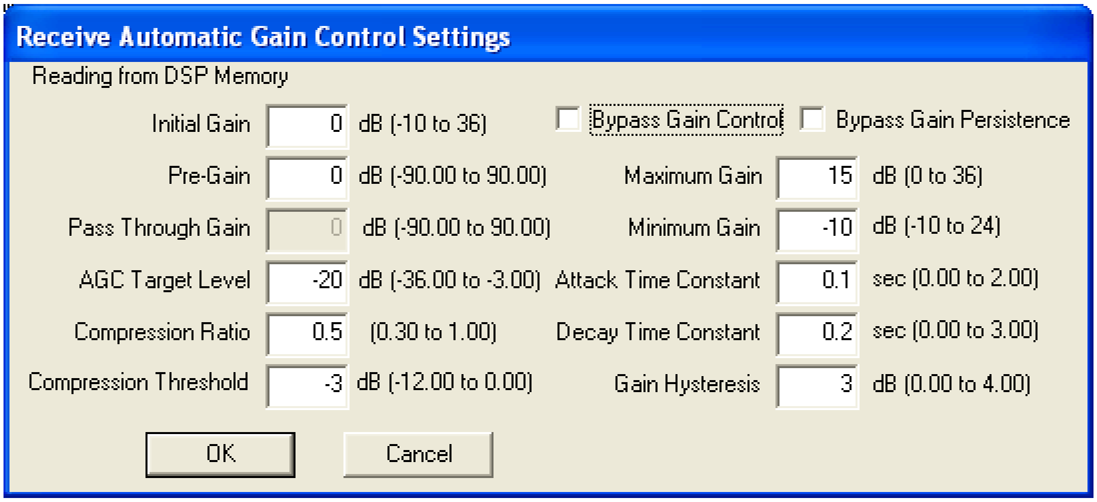



**Figure 5-7 Receive Automatic Gain Control settings windows**

2. Select the required settings based on the Receive AGC options.

**Table 5-6 Receive AGC options**

| Option | Description |
|---|---|
| **Bypass Gain Control** | Leave this option unchecked to use the AGC for signal level control. |
| **Bypass Gain Persistence** | The Gain Persistence stores the last known Receive AGC applied gain in a psKey to be used at the initiation of a new call, helping the Receive AGC adapt quicker at startup.<br>To disable the Calibration Persistence option, check this option. |
| **Initial Gain** | The applied gain of the AGC when cVc is first initialized. |
| **Pre-Gain** | A digital gain applied before the AGC. |
| **Pass Through Gain** | When the system mode is changed to Pass-Through (PT), this digital gain enables you to set the Pass Through Gain since the other blocks are bypassed. This is mainly to compensate for the loss of the AGC block. It is typically used for demonstration when toggling between HFK and PT modes, or for power-saving operation. |
| **AGC Target Level** | Sets the required signal level of the receive output, below which no compression of the input signal occurs. |
| **Compression Ratio** | The Compression Ratio defines the slope of the compression curve used for applying gain to the input signal above the Compression Threshold. Setting the Compression Ratio to 1 results in no compression. As the Compression Ratio values decrease, compression increases. |
| **Compression Threshold** | The point at which compression begins (peak from full scale). |
| **Minimum Gain** | Sets the low threshold level for the gain, so the AGC acts to maintain this value as the minimum gain level. No output has lless than this gain value applied. |
| **Maximum Gain** | Sets the high threshold level for the gain, so the AGC acts to maintain this value as the maximum gain level. No output has less than this gain value applied. |
| **Attack Time Constant** | Sets the rate of attenuation (decreasing gain). If the AGC gain is too high and must decrease, set the Attack Time Constant lower to increase the rate of change. |

**Table 5-6 Receive AGC options (cont.)**

| Option | Description |
|---|---|
| **Decay Time Constant** | Sets the rate when increasing gain. When the voice is low, the AGC wants to slowly increase the gain. By Setting the Decay Time Constant larger the AGC gain increase will be slower to react. |
| **Gain Hysteresis** | Sets the upper and lower boundaries for the gain to change.<br>For example, a value of 4 means that the AGC adjusts only when the speech signal has changed by 4 or more dB above or below the target level. |

3. Click **OK**.

## 5.6 Clipper

If the audio path distorts before the codec reaches full scale, the Clipper may be used as a limiting mechanism to hard-clip the codec output. It also provides a preclipped reference signal to the primary AEC so that optimal echo cancellation can be performed.

If the clip saturation point is not set so that the actual audio path clip point is below the clip saturation point, then optimum function of the primary acoustic echo filter is not achieved.

If the clip saturation point is set so that the actual audio path clip point is above the clip saturation point, then the primary filter can cancel the maximum acoustic echo, but this causes more distortion in the loudspeaker.

To use the Clipper option:

1. From the **Parameter Manager** window, select the **Clipper** block. The **Hard Clipper Settings** window opens.

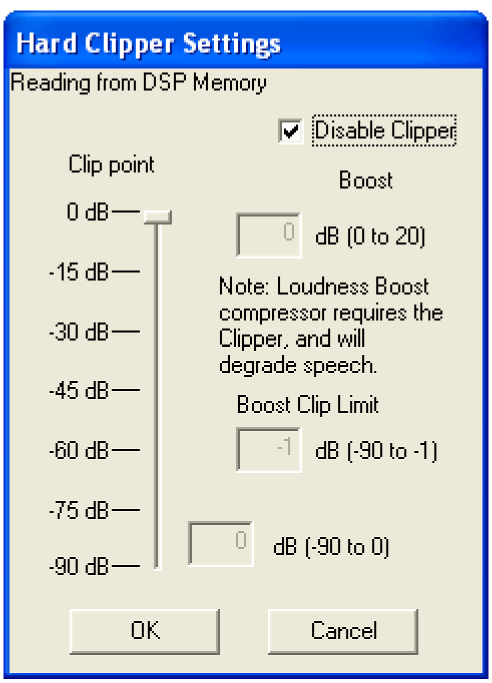



**Figure 5-8 Hard Clipper settings window**

The available Clipper variables are:

- Clip Point/dB Full Scale
- Loudness Boost
- Boost Clip Limit

2. Select the required settings based on the Clipper options.

**Table 5-7 Clipper options**

| Option | Description |
|---|---|
| **Disable Clipper** | By default, this option is selected and the Clipper is disabled.<br>In this state, all the fields and controls on the screen are disabled. The clipper should remain disabled if the system is used with a high-quality (low distortion) audio path.<br>Note:<br>All other parameters are disabled unless the **Disable Clipper** check box is unchecked. |
| **Clip point (slider)**<br>**(also see dB (Full Scale))** | The **Clip point** option sets the clip point. The clip point can be manually set using the dB (Full Scale) field at the bottom of the screen.<br>The clip point or saturation amplitude for the receive signal provides a saturation threshold value for RCV-OUT. Any RCV-OUT signal above the saturation point is clipped to the selected clip point.<br>To set the clip point, move the Clip point slider to the appropriate location on the ruler. The Clip point slider populates the dB (Full Scale) field.<br>Important Note:<br>The Clip point slider and the dB (Full Scale) field are mutually exclusive. The slider location populates the dB (Full Scale) field. A manual entry in the dB (Full Scale) field forces the Clip point slider location to move. |
| **Boost (Loudness Boost)** | The Loudness **Boost** is a digital gain applied to the receive signal, above the saturation point.<br>This setting raises the loudness at the loudspeaker without overdriving the saturation point for the loudspeakers.<br>Increasing the gain in the loudness boost can further degrade the audio quality.<br>To set the loudness boost, type the appropriate setting in this field (the range is from 0 dB to 20 dB). |

**Table 5-7 Clipper options (cont.)**

| Option | Description |
|---|---|
| **Boost Clip Limit** | The **Boost Clip** Limit sets the maximum scale a receive signal can achieve to avoid the saturation (hard limiter).<br>Any boosted audio is hard clipped to the **Boost Clip Limit** setting. This setting raises the loudness at the loudspeaker without overdriving the saturation point for the loudspeakers.<br>Lowering the Boost Clip Limit further degrades the audio quality.<br>The Boost Clip Limit range is -1 dB and adjustable down to -90 dB. |
| **dB (Full Scale) (also see Clip point)** | This field enters the clip point (from -90 to 0). The clip point can be manually entered using this field, or set using the Clip point slider.<br>Important Note:<br>The Clip point slider and the dB (Full Scale) field are mutually exclusive. The slider location populates the dB (Full Scale) field. A manual entry in the dB (Full Scale) field forces the Clip point slider location to move. |

3. Click **OK**.

## 5.7 Auxiliary Stream Mix

The **Auxiliary Stream Mix** block enables the developer to adjust the mix of auxiliary tones and SCO In. In addition to the mixing feature, you can also control the gain applied to the auxiliary tones.

To use the Aux Stream Mix:

1. From the **Parameter Manager** window, click the **Aux Stream Mix** block. The **Auxiliary Stream Mix Settings** window displays.

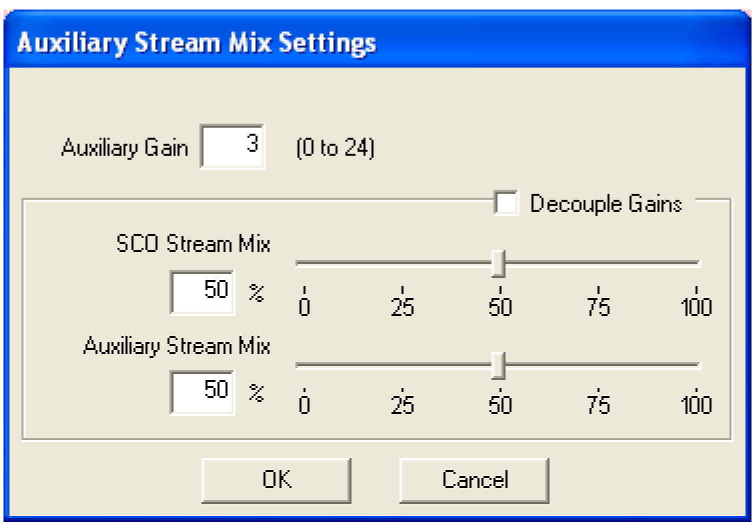



**Figure 5-9 Auxiliary Stream Mix settings window**

2. Select the required settings based on the Auxiliary Stream Mix options.

**Table 5-8 Auxiliary Stream Mix options**

| Option | Description |
|---|---|
| **Auxiliary Gain** | Used to set the amount of gain that is applied to the Auxiliary Signal.<br>To set the Auxiliary Gain, type the appropriate setting in this field.<br>The range is from 0 to 24. |
| **Decouple Gains** | Check this option to decouple the gains. This enables independent mix levels for the SCO and Auxiliary Stream to be set.<br>Do not check this option if you want the SCO and auxiliary signal mix to be coupled. |
| **SCO Stream Mix (slider)** | Sets the ratio in which the SCO and auxiliary streams are mixed. For example, 75% SCO stream mix gives the user 25% auxiliary stream mix. |
| **Auxiliary Stream Mix (slider)** | Sets the ratio in which the SCO and auxiliary streams are mixed. For example, 25% auxiliary stream mix gives the user 75% SCO stream mix. |

3. Click **OK**.

## 5.8 Noise-Dependent Volume Control

The **Noise-Dependent Volume Control (NDVC)** block monitors the noise estimate at the send path. Based on this noise estimate, it attempts to adjust the DAC gain if there is available headroom.

To use the NDVC:

1. From the **Parameter Manager** window, click the **NDVC** block. The **Noise Dependant Volume Control Settings** window opens.

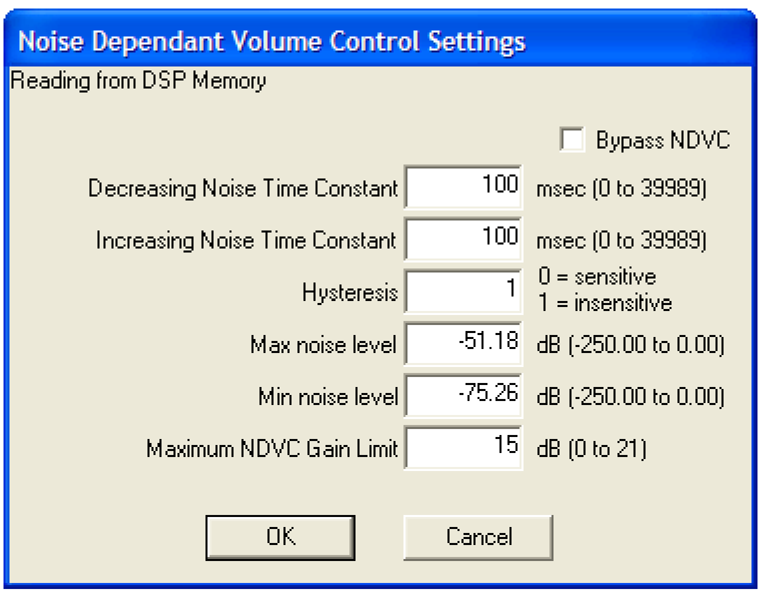



**Figure 5-10 Noise-Dependent Volume Control settings window**

2. Select the required settings based on the Noise-Dependent Volume Control options. See the associated Tuning Guide for alternative settings.

**Table 5-9 Noise-Dependent Volume Control options**

| Option | Description |
|---|---|
| **Bypass NDVC** | By default, the Bypass NDVC option is checked, so the NDVC feature is disabled and the fields on this screen are read-only (grayed out).<br>When this option is unchecked the NDVC feature is enabled and the fields on this screen are available. |
| **Increasing Noise Attack Time Constant** | Sets the attack time constant that increases the volume steps based on the noise level. |
| **Decreasing Noise Decay Time Constant** | Sets the decay time constant that decreases the volume steps based on the noise level. |
| **Hysteresis** | Sets the sensitivity when switching between adjacent volume states (Range 0.00 to 1.00).<br>Lowering the value gives higher sensitivity. For example, a value of 0.75 means the hysteresis is more sensitive than when set to 1.00. |
| **Max noise level** | This sets the noise level threshold at which the NDVC adds the maximum gain as specified in the Maximum NDVC Gain Limit option. |
| **Min noise level** | This sets the noise level threshold at which the NDVC adds the minimum gain and Gain is added. |
| **Maximum NDVC Gain Limit** | This setting limits the maximum gain that the NDVC applies.<br>**Note:**<br>Set up the system code to accommodate the appropriate gain limit. For example, if a gain change of 9 dB is required, then the maximum volume level must be -9 dB in the system code. |

3. Click **OK**.

## 5.9 Side Tone

The Side Tone option determines the gain applied to the sidetone signal. Side tone is the signal picked up by the headset's front microphone and reproduced at the headset's receiver.

To adjust the Side Tone setting:

Side tone can be fixed or has a built-in mechanism that adjusts the amount based on the Bluetooth volume. The leveling mechanism (**Gain Adjustment Limit**) adds proportional sidetone as the Bluetooth volume drops to maintain a "leveled" sidetone. Automatic limits are computed to avoid DAC saturation.

1. From the **Parameter Manager** window, click the **Side Tone** block. The **Side Tone Generation Settings** window opens.

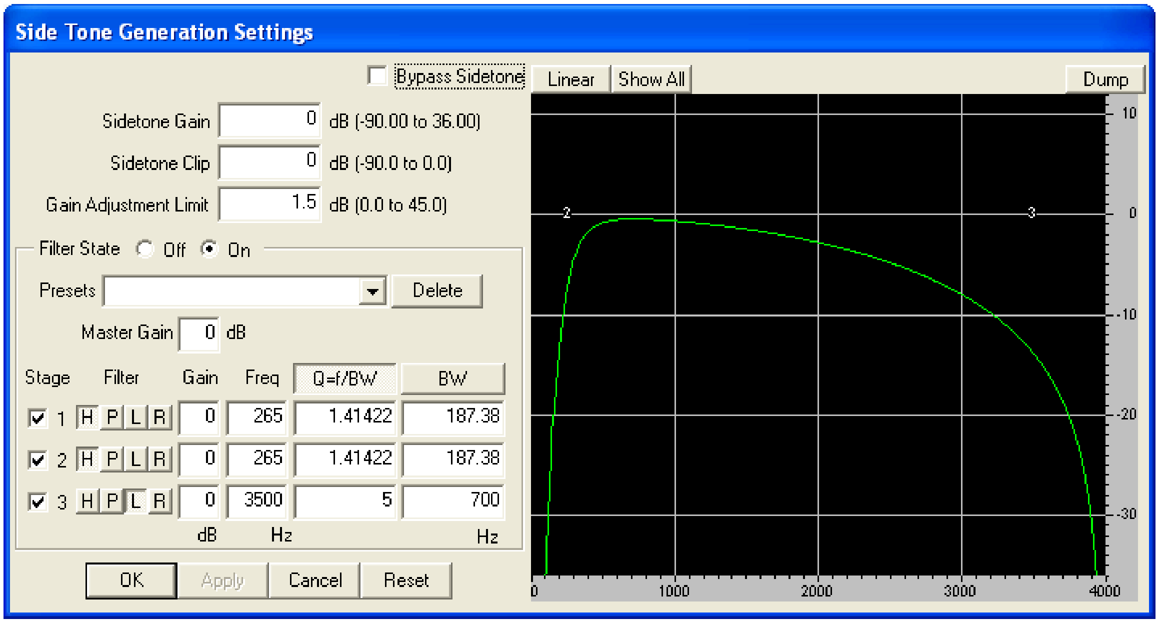



**Figure 5-11    Side Tone Generation settings window**

2. Select the required settings based on the Side Tone Generation, gain options..

**Table 5-10    Side Tone Generation, gain options**

| Option | Description |
|---|---|
| **Bypass Sidetone** | If checked, bypasses the sidetone feature reducing processor cycles. |
| **Sidetone Gain** | Select the required **Sidetone Gain**. The higher the gain the more sidetone is applied. The Default is 0 dB and the allowable range is from -90 dB to 36 dB. |
| **Sidetone Clip** | If you require a large amount of sidetone gain and the DAC could saturate, **adjust the Sidetone Clip** to avoid saturation.<br>The Default is 0 dB (no clip or DAC full scale). As the clip value becomes more negative, more clipping occurs.<br>The allowable range is from -90 dB to 0 dB.<br>Note:<br>The **Sidetone Clip** invokes a hard clipper that causes some audio distortion. |
| **Gain Adjustment Limit** | Adjust the **Gain** > **Adjustment Limit** parameter to adjust the amount of sidetone gain based on Bluetooth volume. The amount of leveling is controlled by the **Gain** > **Adjustment Limit**parameter.<br>The allowable range is from 0 (no leveling) to 45 dB.<br>Note:<br>The **Gain** > **Adjustment Limit** leveling mechanism adds proportional sidetone as the Bluetooth volume drops to maintain a leveled sidetone. |

**Table 5-10 Side Tone Generation, gain options (cont.)**

| Option | Description |
|---|---|
| **Filter State** | The **Filter State** can be set **Off** or **On**. If **On**, then a programmable three stage filter is applied to the sidetone audio. If set **Off**, than the raw sidetone audio is passed. |
| **Presets**<br>**Master Gain**<br>**Stage**<br>**Filter**<br>**Gain**<br>**Freq.**<br>**Q=f/BW**<br>**BW** | If the **Filter State** is **On**, then a programmable filter is applied to the sidetone audio. The operation of the filter is the same as described in Section Receive EQ the difference being only 3 stages are available. See Figure 5-11 for an example filter and the GUI. |

3. Make the required changes and click **OK.**

# 6 Using Send Path tuning controls

Parameter Manager provides access to these send path tuning controls:

- Wind Noise Reduction
- Dual Microphone Signal Separation
- Noise Suppression
- Acoustic Echo Canceller
- Comfort Noise
- Send EQ
- Send AGC

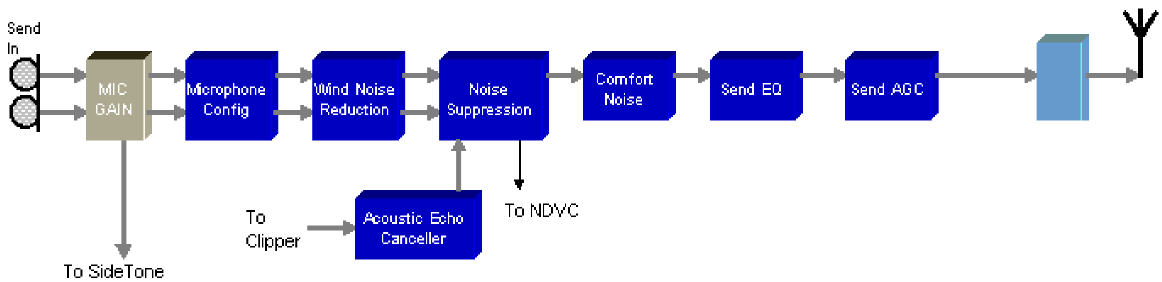


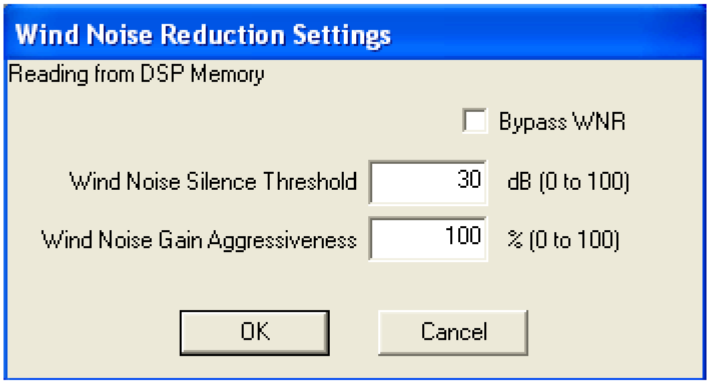

**Figure 6-1 Send path tuning controls**

## 6.1 Wind Noise Reduction

The Wind Noise Reduction (WNR) block detects and reduces the wind noise in the presence of wind while attempting to improve speech quality.

To access the Wind Noise Reduction features:

1. From the **Parameter Manager** window, click the **Wind Noise Reduction** block. The **Wind Noise Reduction Settings** window opens.




2. Select the required settings based on Wind Noise Reduction options.

**Table 6-1 Wind Noise Reduction options**

| Option | Description |
|---|---|
| **Bypass WNR** | By default, this option is unchecked and Wind Noise Reduction is enabled.<br>To disable the residual echo reduction, check this option. Disabling the wind noise reduction saves ~1.7 MIPS. |
| **Wind Noise Silence Threshold** | The Wind Noise Silence Threshold value is used with the Mic Internal Noise Level to control the wind detection mechanism. |
| **Wind Noise Gain Aggressiveness** | Wind Noise Gain Aggressiveness controls the amount of wind noise suppression provided by the algorithm. Increasing this value increases the amount of wind noise reduction.<br>The closer this field is set to 100%, the more reduction is applied. |

3. Click **OK**.

## 6.2 Microphone Configuration Settings

The Microphone Configuration Settings block may be tuned for considerations such as microphone set-up, microphone internal noise levels, microphone calibration, and aggressiveness.

To set Microphone Configuration options:

1. From the **Parameter Manager** window, select the **Microphone Config** block. The **Microphone Configuration Settings** window opens.

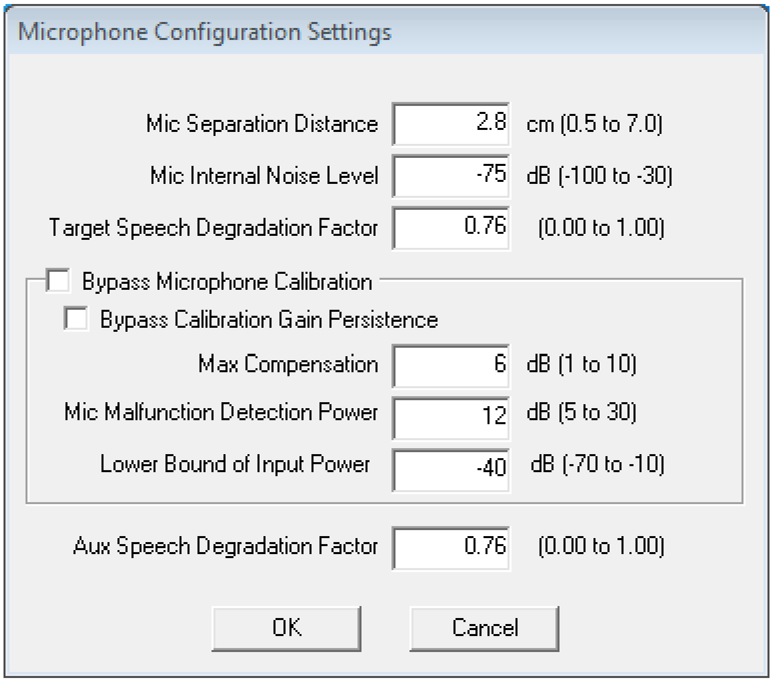



**Figure 6-2 Microphone Configuration settings**

2. Select the required settings based on the Microphone Configuration Setting options.

**Table 6-2 Microphone Configuration Setting options**

| Option | Description |
|---|---|
| **Mic Separation Distance** | Simply the distance between the center points of each microphone on the Headset (in centimeters) |
| **Mic Internal Noise Level** | Power level of microphone noise in dB, measured in a quiet room, such as an office-type environment |
| **Target Speech Degradation Factor** | Amplitude degradation of target speech from primary microphone to secondary microphone |
| **Bypass Microphone Calibration** | When left unchecked, the algorithm uses the microphone calibration parameters |
| **Bypass Calibration Gain Persistence** | When left unchecked, the headset stores the MGDC information for future calls (after a power-up). Otherwise, it recomputes starting from zero |
| **Max Compensation** | Maximum level compensation between the primary and secondary microphones |
| **Mic Malfunction Detection Power** | Power threshold to detect the malfunction of microphones. If the level difference between the two microphones exceed this value, then the algorithm switches to 1 mic mode |

**Table 6-2 Microphone Configuration Setting options (cont.)**

| Option | Description |
|---|---|
| **Lower Bound of Input Power** | MGDC adapts calibration gain based on signal with power greater than this lower bound. If the input channels have the same power of noise floor, the lower bound can be set below the noise floor and MGDC can adapt calibration gain all the time. |
| **Aux Speech Degradation Factor** | The target speech degradation factor used in rejecting (auxiliary) beam-forming. |

3. Click **OK**.

## 6.3 Noise Suppression

The Noise Suppression block defines the aggressiveness of the noise suppression algorithm.

To set Noise Suppression options:

1. From the **Parameter Manager** window, select the **Noise Suppression** block. The **Noise Suppression Settings** window opens.

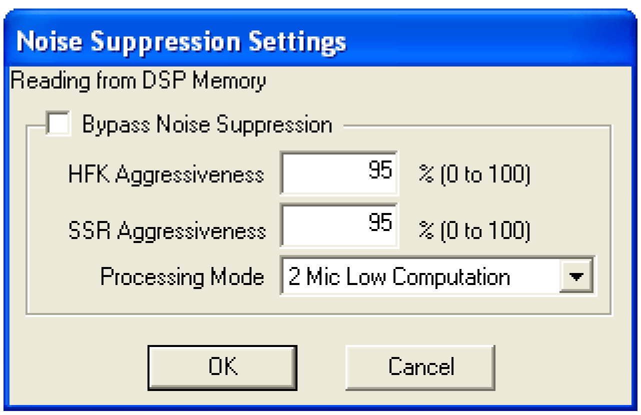



**Figure 6-3 Noise Suppression settings window**

2. Select the required settings based on the Noise Supression Settings options.

**Table 6-3 Noise Suppression settings window options**

| Option | Description |
|---|---|
| **Bypass Noise Suppression** | If checked, bypasses the NS feature reducing processor cycles.<br>Important Note:<br>Other modules depend on intelligence within the Send Noise Suppression block, if bypassed system performance could degrade. |
| **HFK Aggressiveness** | Controls the amount of noise suppression applied to the send signal during main processing- HFK mode.<br>To achieve a good balance between voice quality and noise suppression, set the **HFK Aggressiveness** to **95%** which yields > 17 dB of SNR improvement.<br>Setting this parameter to 100% maximizes noise suppression. |

**Table 6-3 Noise Suppression settings window options (cont.)**

| Option | Description |
|---|---|
| **SSR Aggressiveness** | Controls the amount of noise suppression applied to the send signal during the noise suppression mode of operation.<br>Setting this parameter to 100% maximizes suppression. |
| **Processing Mode** | Selection between 2 Mic Low Computation, 2 Mic Normal Computation and 2 Mic High Computation modes. Increased Computation mode increases noise suppression and voice quality at the trade-off of MIPS |

3. Click **OK**.

## 6.4 Acoustic Echo Canceller

The Acoustic Echo Canceller reduces echo that is caused by the acoustic coupling of the loudspeaker to the microphone.

To access The Echo Cancellation features:

1. From the **Parameter Manager** window, click the **Acoustic Echo Canceller** block. The **Acoustic Echo Cancellation Settings** window opens.

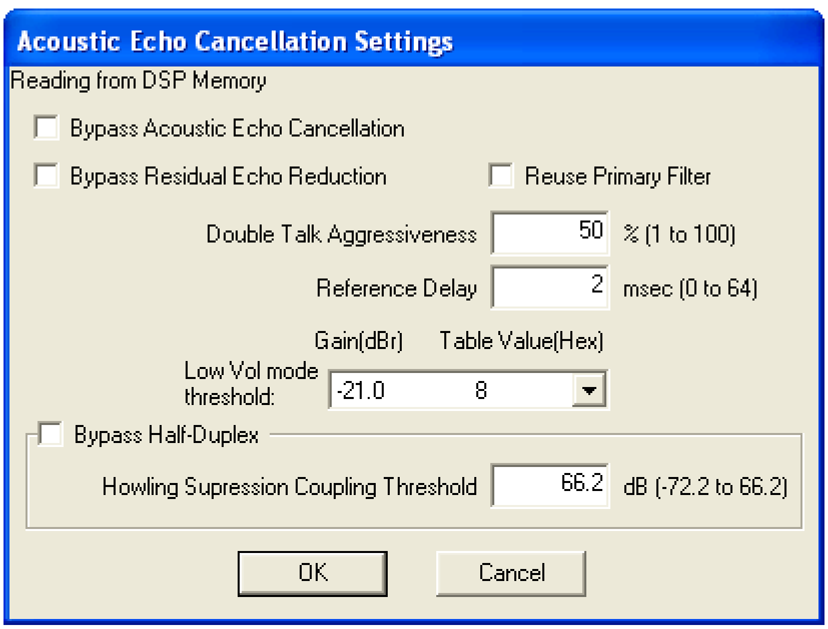



**Figure 6-4 Acoustic Echo Cancellation settings window**

2. Select the required settings based on the Acoustic Echo Cancellation options.

**Table 6-4 Acoustic Echo Cancellation options**

| Option | Description |
|---|---|
| **Bypass Acoustic Echo Cancellation** | By default, this option is unchecked and the AEC module is included in the system.<br>To disable AEC processing, check this option. |
| **Bypass Residual Echo Reduction** | By default, this option is unchecked and the Residual Echo Reduction is enabled.<br>This feature is best used when there are no nonlinearities in a system and the acoustic coupling of the system is minimal.<br>To disable the residual echo reduction, check this option. Disabling the residual echo reduction saves ~1.4 MIPS. |
| **Reuse Primary Filter** | Check this option when the echo coupling is high. Selecting this option increases the DSP processing load because the audio is recycled through the primary AEC filter to further reduce echo.<br>Under certain conditions where processing power is not a concern, then this option is enabled to ensure that the AEC primary filter converges as close as possible to the optimal level. |
| **Double Talk Aggressiveness** | Determines the amount of attenuation that is applied during double talk.<br>The closer this field is set to 100%, the less echo attenuation is applied. |
| **Reference Delay** | **Reference Delay** is a delay buffer that compensates for the latency in the signal as it travels:<br>From the DAC to the loudspeaker<br>Over the acoustic enclosure<br>To the ADC<br>This delay in milliseconds is presented to the AEC as the echo component in the Microphone signal.<br>Type the appropriate reference delay setting in this field if other than the default is required.<br>**Note:**<br>If the delay is longer than the actual latency, the microphone signal with echo arrives at the AEC before the delayed reference signal, resulting in no cancellation of the echo.<br>If the entered delay is shorter than the actual latency, the algorithm's effective echo tail length is reduced. |
| **Low Vol mode threshold** | This is a switch threshold that disables the AEC and Comfort Noise when they are no longer required.<br>This feature disables the AEC when the speaker's acoustic loudness is decreased due to a decrease in the headset volume.<br>The **Low Vol mode threshold** is set to the Bluetooth step where the echo is tolerable without the software AEC.<br>When the Bluetooth volume step is at or above the threshold, the AEC and Clipper are enabled.<br>This mechanism is a power-saving technique. |

**Table 6-4 Acoustic Echo Cancellation options (cont.)**

| Option | Description |
|---|---|
| **Bypass Half-Duplex** | By default, this option is checked and the feature is disabled.<br>Uncheck the option to enable the Howling Control, which provides attenuation to the send out signal when the echo signal at the microphone (acoustic coupling) is very loud and provides half duplex communication when the coupling crosses the **Howling Suppression Coupling Threshold**. |
| **Howling Suppression Coupling Threshold** | The **Howling Suppression Coupling Threshold** sets the threshold for the minimum acoustic coupling value to force attenuation on the send out signal (half duplex). |

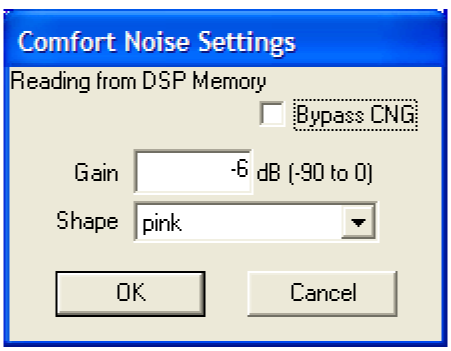
3. Click **OK**.

## 6.5 Comfort Noise

The **Comfort Noise** block mitigates the noise floor modulations introduced by the residual echo reduction, generated by the AEC.

To use the Comfort Noise option:

1. From the **Parameter Manager** window, select the **Comfort Noise** block. The **Comfort Noise Settings** window opens.




2. Select the required settings based on the Comfort Noise options.

**Table 6-5 Comfort Noise options**

| Option | Description |
|---|---|
| **Bypass CNG** | This check box enables or disables the comfort noise gain control, and during the tuning process, to ensure the proper amount of comfort noise is added.<br>By default, the comfort noise gain control is enabled.<br>When Comfort Noise is enabled, it provides smoothness to the background noise during echo removal times and enhances perceptual quality of audio. |
| **Gain** | The Gain setting is available only when the **Bypass CNG** check box is not selected, and controls the amplitude level of the added comfort noise signal. Type the appropriate value in this field. |
| **Shape** | The Shape setting is intended to enable the user to choose the weighting of the comfort noise spectrum. Choices are: Brown, Pink, White, Blue, and Purple. |

3. Click **OK**.

## 6.6 Send EQ

The Send EQ block opens the **Send Equalizer Settings** window. The Send EQ and the Receive EQ windows are almost identical. See Receive EQ for instructions on using the features on the **Send Equalizer Settings** window.

## 6.7 Send AGC

The **Send AGC** block includes a Pre-**Gain** field that can be used as a pre-gain to the AGC when the AGC is not bypassed.

To adjust the Send AGC Settings:

1. From the Parameter Manager window, click the **Send AGC** block. The **Send Automatic Gain Control Settings** window opens.

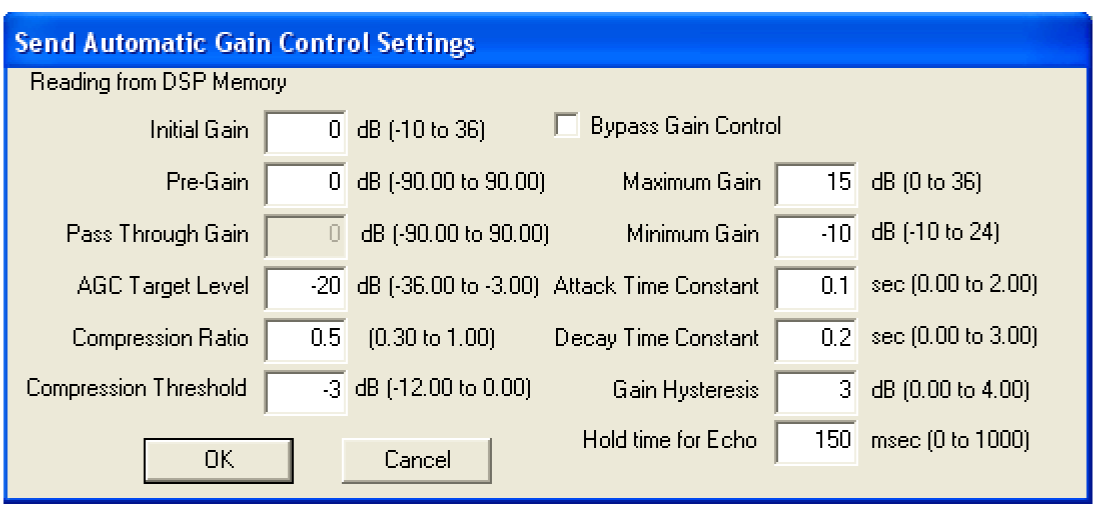



**Figure 6-5 Send Automatic Gain Control settings window**

2. Select the required settings based on the Send Automatic Gain Control options.

**Table 6-6 Send Automatic Gain Control options**

| Option | Description |
|---|---|
| **Bypass Gain Control** | Leave this option unchecked to use the AGC for signal level control. |
| **Initial Gain** | The applied gain of the AGC when cVc is first initialized. |
| **Pre-Gain** | A digital gain applied before the AGC. |
| **Pre-Amp** | A digital gain applied before the AGC. |
| **Pass Through Gain** | When the system mode is changed to Pass-Through (PT), this digital gain enables you to set the **Pass Through Gain** since the other blocks are bypassed. This is mainly to compensate for the loss of the AGC block. Typically used for demonstration when toggling between HFK and PT modes, or used for power-saving operation. |
| **AGC Target Level** | Sets the required signal level of the receive output, below which no compression of the input signal occurs (usually set close to Full Scale level). |
| **Compression Ratio** | The Compression Ratio defines the slope of the compression curve used for applying gain to the input signal above the AGC Target Level. Setting the **Compression Ratio** to 1 results in no compression, because the **Compression Ratio** values decrease while compression increases. |
| **Compression Threshold** | The point at which compression begins (peak from full scale). |
| **Minimum Gain** | Sets the low threshold level for the gain, and the AGC acts to maintain this value as the minimum gain level. No output has less than this gain value applied. |
| **Maximum Gain** | Sets the high threshold level for the gain, and the AGC acts to maintain this value as the maximum gain level. No output has more than this gain value applied. |
| **Attack Time Constant** | Sets the rate of attenuation (decreasing gain). If the AGC gain is too high and must decrease, sett the **Attack Time Constant** lower to increase the rate of change. |
| **Decay Time Constant** | Sets the rate when increasing gain. When the voice is low, the AGC wants to slowly increase the gain. By Setting the **Decay Time Constant** larger the AGC gain increase is slower to react. |
| **Gain Hysteresis** | Sets the upper and lower boundaries for the gain to change.<br>For example, a value of 4 means that the AGC will adjusts only when the speech signal has changed by 4 or more dB above or below the target level. |
| **Hold Time for Echo** | This prevents the AGC from changing gain caused by residual echo and should only adapt during near end speech. This parameter parameter sets the amount of hold adaptation of the Send AGC following an echo event. |

3. Click **OK**.

# 7 Exiting the Parameter Manager application

To exit the **Parameter Manager** application, select **File**/**Exit** from the menu bar. The **Parameter Manager** window closes.

# 8 Matching Parameter Manager and DSP code versions

The version of the Parameter Manager must match the DSP code version. The Parameter Manager headset application is compatible with only one version of cVc.

When the Parameter Manager application is started, the **Universal Parameter Manager** window displays.

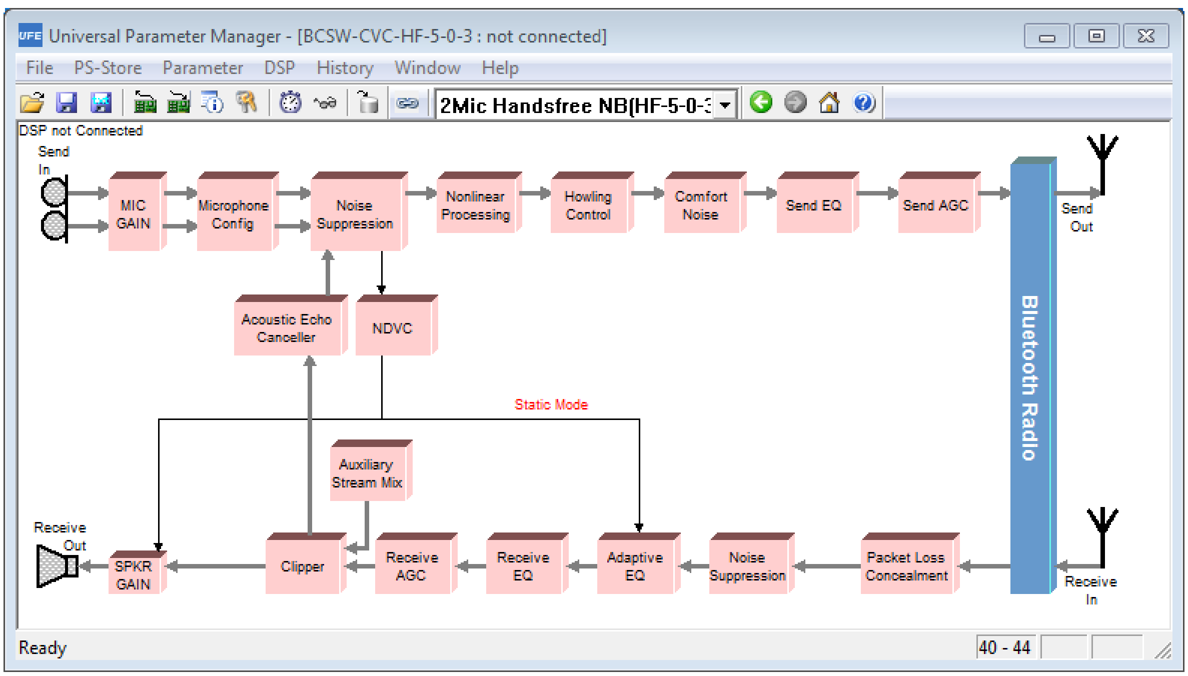



**Figure 8-1 Parameter Manager Window in static mode**

The title bar shows **[BCSW-CVC-HS-6-0-2 : not connected]** because a phone call has not been activated the system is in the Static mode of operation, the cVc slave device is not running and the Monitoring mode cannot be accessed.

To activate cVc initiate a call. When a call is activated, the Parameter Manager the title bar shows **[BCSW-CVC-HS-6-0-2 : SPI->LPT1]**, and the top left corner of the window shows the product code **[BCSW-CVC-HS-6-0-2]**, version number **[vB10E** and build number **[B1]**.

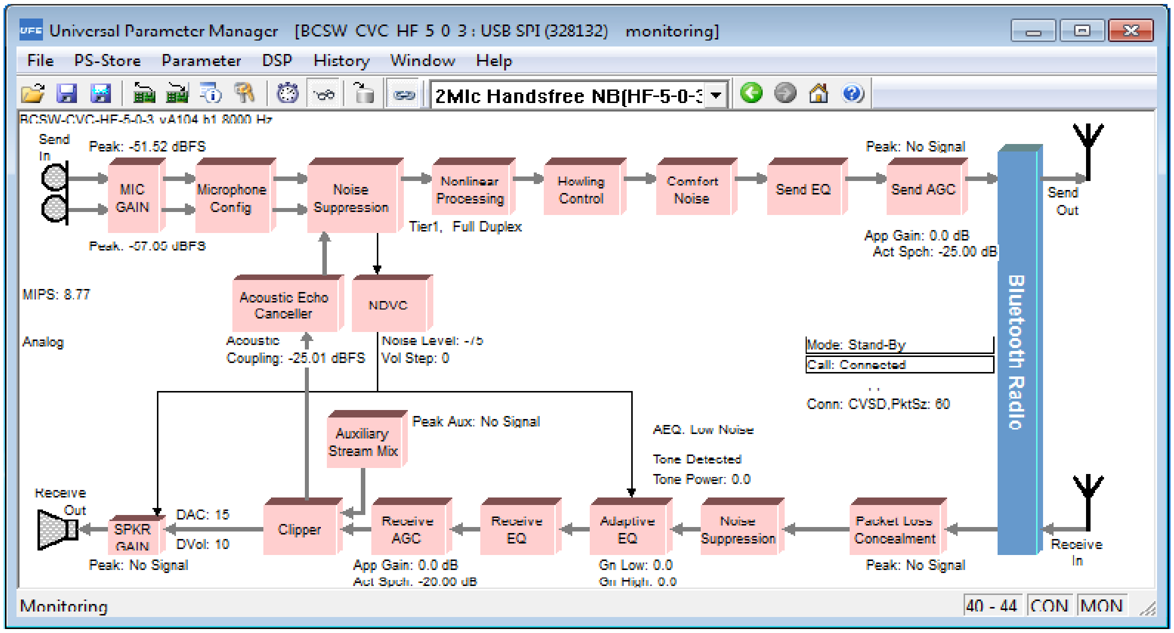
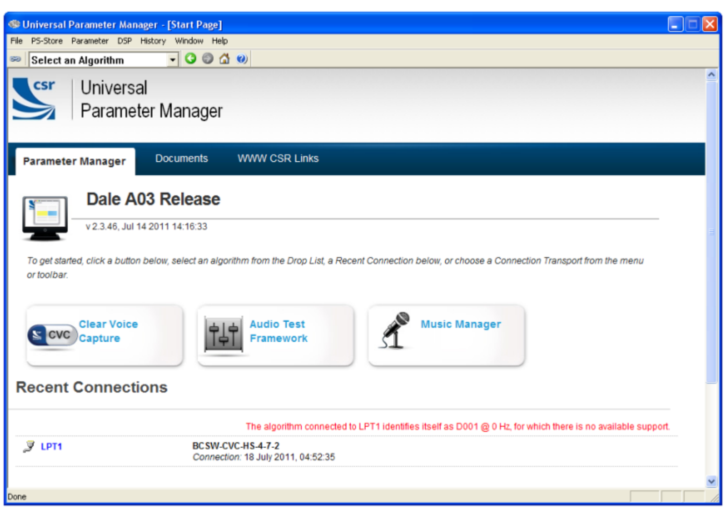



**Figure 8-2 Parameter Manager Window in an active call**

A status message confirms that the Parameter Manager matches the cVc code version. The system remains in a Static mode of operation.

When the Parameter Manager and the cVc code versions do not match, **DSP Not Responding** or **Unknown DSP Software** displays in the **Parameter Manager** window or automatically jumps to the home page.



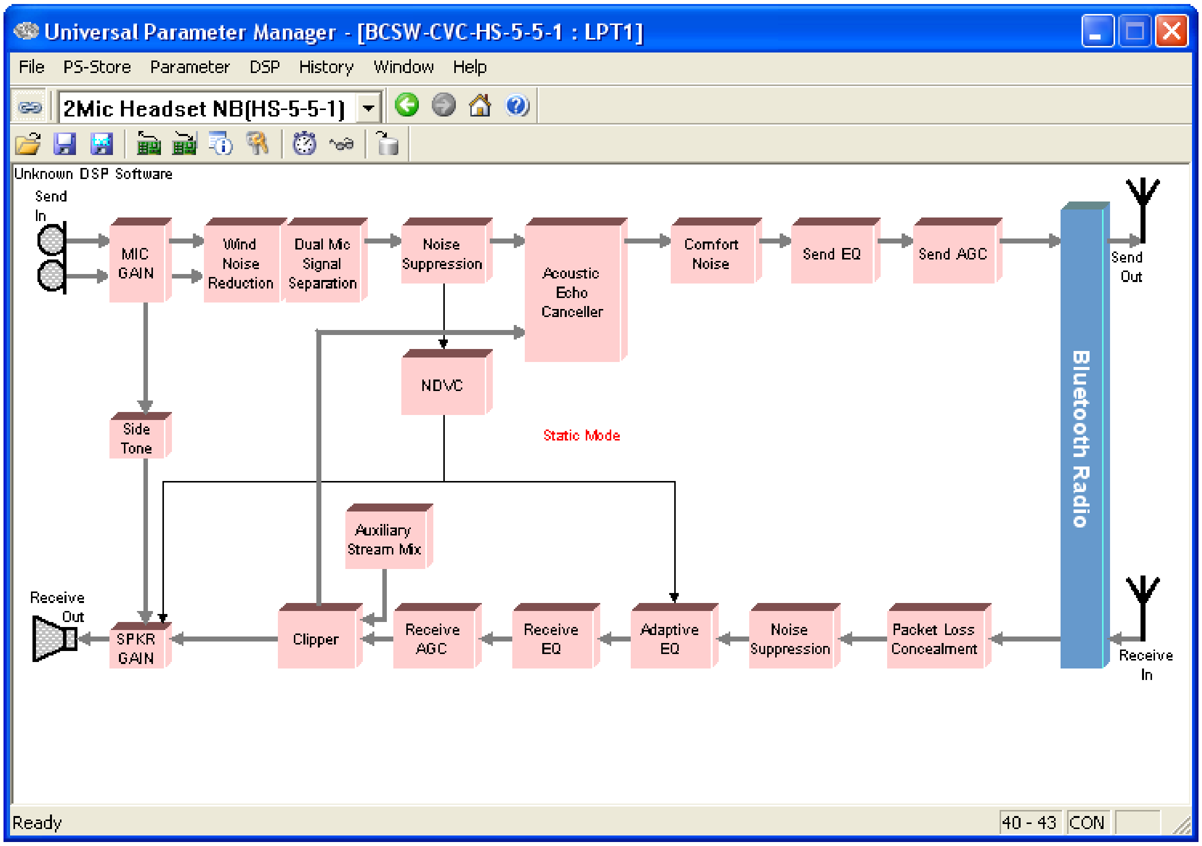



**Figure 8-3 Parameter Manager window showing invalid version message**

In the **Recent Parameter Manager Connections** section details are provided, that attempt to identify the software version found and actions to remedy the problem.

If the DSP is not running (no active call) this text displays:

**NOTE** The device connected to SPI -> LPT1 does not identify itself as a cVc algorithm. Ensure that the device is connected and enabled.

# Document references

| Document | Reference |
|---|---|
| *Enabling cVc for Headset SDKs* | 80-CT409-1/CS-00122720-AN |
| *cVc 8th Generation 2-mic Tuning Guide (BCSW-CVC-HS-6-0-2, B10D)* | 80-CF054-1/ CS-00404394-DC |
| *QCC3001 BGA Mono Headset User Guide* | 80-CF158-1/CS-00405482-UG |
| *QCC3002 WLCSP Mono Headset with Qualcomm aptX User Guide* | 80-CF167-1/CS-00405485-UG |
| *QCC3002 BGA Mono Headset with Qualcomm aptX User Guide* | 80-CF160-1/CS-00405486-UG |
| *QCC3004 Stereo Headset User Guide* | 80-CF073-1/CS-00405040-UG |
| *QCC3005 Stereo Headset with Qualcomm aptX User Guide* | 80-CF072-1/ CS-00405036-UG |
| *QCC3001 WLCSP Mono Headset User Guide* | 80-CF159-1/CS-00405484-UG |

# Terms and definitions

| Term | Definition |
|---|---|
| ADC | Analogue to Digital Converter |
| AEC | Acoustic Echo Canceller |
| AEQ | Adaptive Equaliser |
| AGC | Automatic Gain Control |
| ALT | Automatic Level Tuning |
| ASR | Automatic Speech Recognition |
| BlueCore | Group term for QTIL's range of Bluetooth wireless technology chips. |
| Bluetooth SIG | Bluetooth Special Interest Group |
| Bluetooth | Set of technologies providing audio and data transfer over short-range radio connections. |
| BCSW | BlueCore Software |
| CODEC | Coder Decoder |
| cVc | Clear Voice Capture |
| CVSD | Continuous Variable Slope Delta Modulation |
| DAC | Digital to Analogue Converter |
| DMSS | Dual Microphone Signal Separation |
| DSP | Digital Signal Processor |
| e.g. | *exempli gratia*, for example |
| ENR | Echo and Noise Reduction |
| etc | *et cetera*, and the rest, and so forth |
| EQ | Equalizer |
| GSM | Global System (for) Mobile (communications) |
| GUI | Graphical User Interface |
| HFK | Handsfree Kit |
| HS | Headset |
| HTML | HyperText Markup Language |
| IC | Integrated Circuit |
| I/O | Input/Output |
| ICs | Integrated Circuits |
| MIC | Microphone |

| Term | Definition |
|---|---|
| MIPS | Million Instructions Per Second |
| NB | Narrow Band |
| NDVC | Noise Dependant Volume Control |
| NS | Noise Suppression |
| OMS | One Microphone Solution noise reduction |
| PC | Personal Computer |
| PCM | Pulse Code Modulation |
| PEQ | Parametric Equalisation |
| PLC | Packet Loss Concealment |
| QTIL | Qualcomm Technologies International Ltd |
| RCV | Receive |
| ROM | Read Only Memory |
| SCO | Synchronous Connection-Oriented |
| SDK | Software Development Kit |
| SNR | Signal to Noise Ratio |
| SPI | Serial Peripheral Interface |
| SPKR | Speaker |
| SSR | Simple Speech Recognition |
| ST | Side Tone |
| STMR | Side Tone Masking Rating |
| UFE | Universal Front End |
| VM | Virtual Machine |
| WB | Wide Band |
| WNR | Wind Noise Reduction |