Qualcomm Technologies International, Ltd.

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

User Guide

80-CT413-1 Rev. AH

November 7, 2017

Revision history

Revision	Date	Description
1	December 2013	Original publication of this document. Alternate document number: CS-00309823-UG
2	December 2013	Minor editorial corrections
3	March 2014	Editorial changes
4	May 2014	Editorial changes
5	July 2014	Updated for ADK 3
6	June 2015	Updated for ADK 4.0
7	September 2016	Updated for ADK 4.1
AH	November 2017	Updated for ADK 4.3. Updated to reflect new document and revision numbering.

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1 Introduction

The Parameter Manager is part of the UFE, which is a Windows-based application that enables the handsfree device manufacturer to configure and monitor the Qualcomm Technologies International, Ltd. (QTIL) Handsfree DSP audio application software. The UFE is available with the appropriate version of the Audio Development Kit (ADK). The ADK is available from www.csrsupport.com.

QTIL provides Clear Voice Capture (cVc) algorithms that create voice products. The Parameter Manager application enables you to customize the performance of your handsfree system. This document describes the Parameter Manager application, which is a wizard-like graphical user interface (GUI) that operates in a Windows environment.

Use the Parameter Manager application with the cVc audio application running on a BlueCore DSP. The cVc Two Microphone (2-mic) Handsfree application provides the following modules, which are accessible using the Parameter Manager application:

- Microphone Configuration
- Noise Suppression (NS) that includes:
  - Beam Former (BF)
  - Blind Source Separation (BSS)
  - Adaptive Noise Canceller (NC)
  - Acoustic Events Detection (AED)
  - Stationary Noise Suppression
- Acoustic Echo Cancellation (AEC) that includes:
  - Nonlinear Processing
  - Howling Control
- Automatic Gain Controls (AGC)
- Equalizers (EQ)
- Stream Mixer
- Clipper
- Near End Audio Enhancement that includes:
  - Noise Dependent Volume Control (NDVC)
  - Adaptive EQ (AEQ)
  - Packet Loss Concealment (PLC)
  - Noise suppression

This document describes how to use the Parameter Manager application for basic tuning and monitoring activities. See the *BCSW-CVC-HF-5-0-3 2M-HF Tuning Guide* for information on the tuning process.

1.1 Software versions supported

This document describes the audio controls of cVc BCSW-CVC-HF-5-0-3 algorithm. Table 1.1 describes the audio controls and adjustments that are used on the IC.

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 supports narrow band (8 kHz sample rate) using CVSD, and includes wide band (16 kHz sample rate) using modified sub band coding (mSBC).

Download the QTIL86xx Parameter Manager UFE installer from www.csrsupport.com.

1.2 8th Generation new features

This algorithm is for use in a vehicle's overhead console, where the microphones are configured in a broadside arrangement. Using beam forming technology, the manufacturer can adjust the beam into a fixed direction towards the primary talker, all other talkers and noise sources outside this beam is attenuated.

New and improved 8th generation features include:

- Two microphone beam forming
- Additional noise suppression options after the beam former is provided:
  - BSS (Blind Source Signal Separation)
  - NC (noise cancellation)
- Updated AGC Module that improves tracking of the target level
- Send AGC does not adapt gain during receive speech (echo)
- Updated CNG Module that enables selectable colored noise
- DSP Framework that is updated and optimized using a frame size of 60 samples
- PEQ Master Gain that independently forms bi-quad stages
- Normalized tone volumes between processing modes with and without CVC
- Microphone Configuration
- Reduced tuning complexity for WNR

- Reduced tuning complexity for noise suppression
- Half Duplex Mode

1.3 Assumptions

This document assumes:

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

2 Getting started

To use the Parameter Manager application:

1. Install the QTIL Audio Development Kit or the UFE Installer from www.csrsupport.com.
2. Run the UFE application.
3. Click the **Quick Start** link in the UFE Document section for more information about the Parameter Manager application.
4. Open the 2-mic Handsfree Parameter Manager application.
5. Connect the Parameter Manager application using an active SPI.
6. Enter the security key if required.
7. Perform one of these steps:
   - Pair and connect the handsfree device with a Bluetooth source device, such as a Bluetooth phone, as the audio gateway.
   - Connect USB to PC if USB wired mode is used.
8. Use parameters and metrics information for tuning and/or monitoring.

2.1 Installing the standalone universal front end application

For the standalone UFE Installer, see www.csrsupport.com.

2.1.1 For flash-based ICs using the ADK

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

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

The installation process creates a corresponding Start Menu link:

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

2.1.2 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>

2.2 Accessing the UFE application

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

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

Access the UFE application from the Windows **Start** menu. Figure 2-1 shows an example UFE Installer.

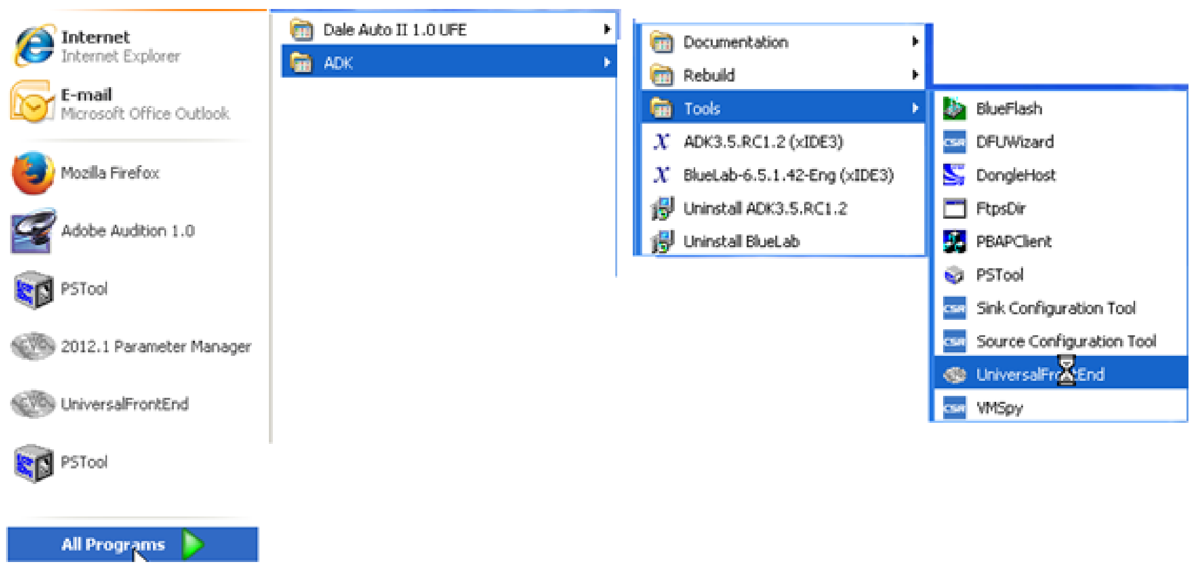


Figure 2-1 Accessing UFE from the Windows start menu

Click **QTIL ADK Parameter Manager** to open an HTML page. .

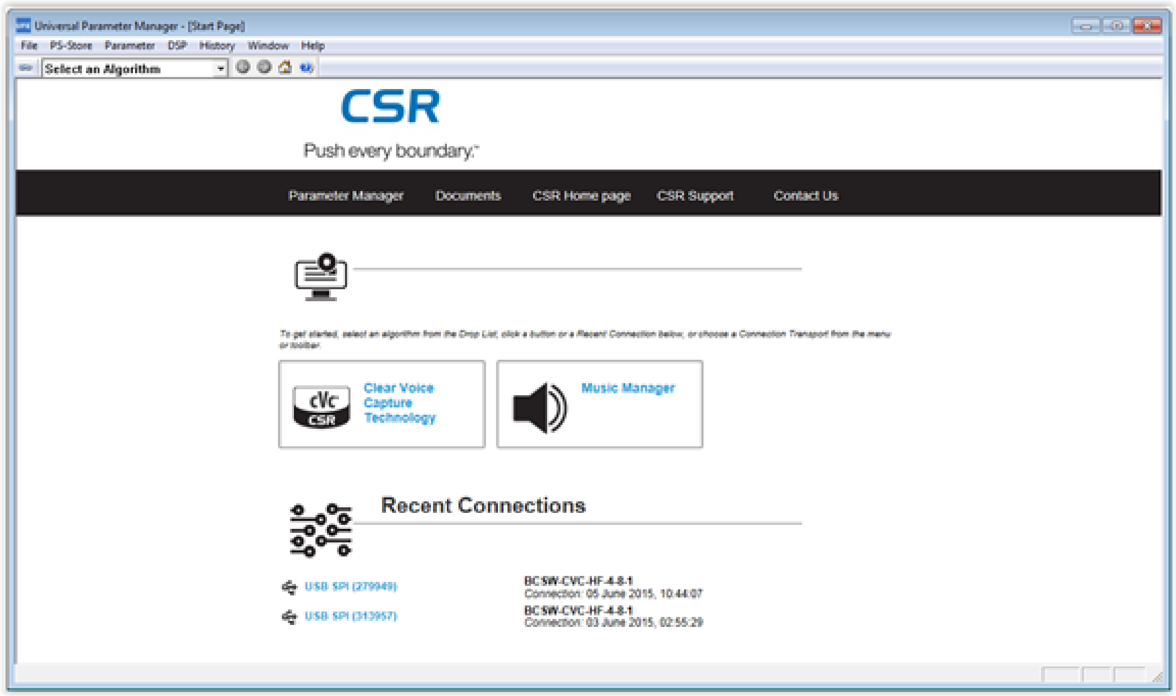


Figure 2-2 UFE application opening HTML page showing quick start link

2.3 Viewing the UFE Quick Start Guide

To view the UFE Quick Start Guide:

1. Click the **Documents** link on the opening HTML page.
2. Select **Quick Start**.
3. See Figure 2-2 for the location of the hyperlink. After you select it, the document appears.
4. Click the Home icon to return to the opening HTML page.

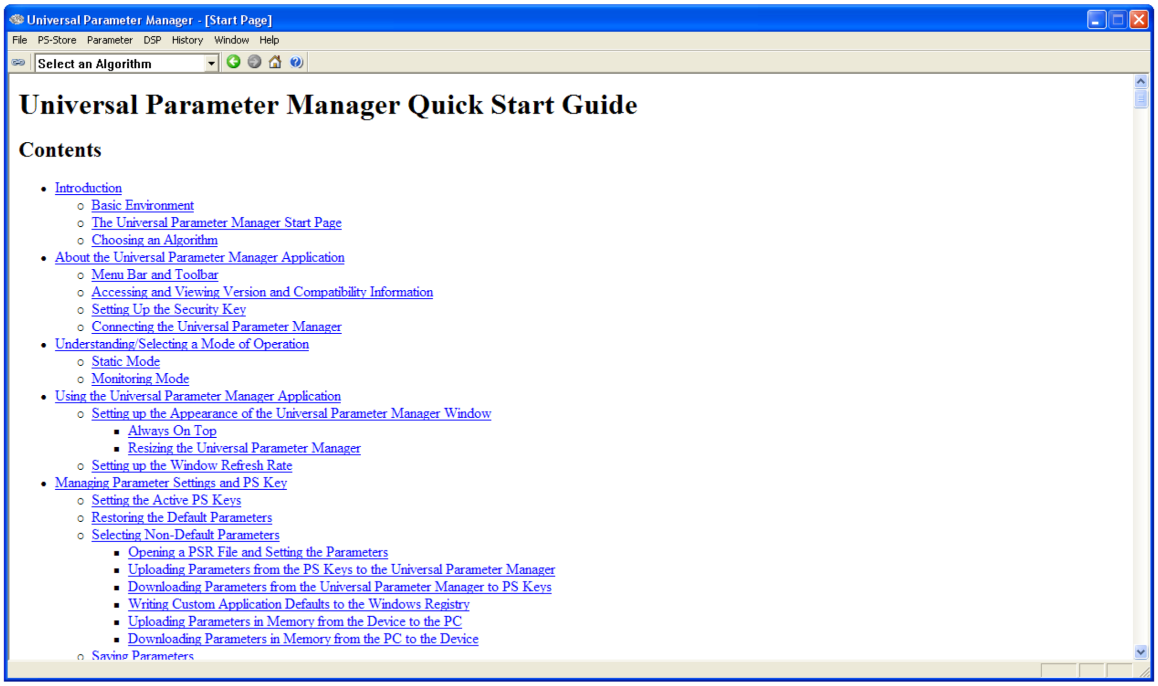


Figure 2-3 UFE quick start embedded documentation

2.4 Accessing the Handsfree Parameter Manager

To access the Handsfree Parameter Manager from the UFE Opening HTML page, select the **Select an Algorithm** from the dropdown list in the menu bar.

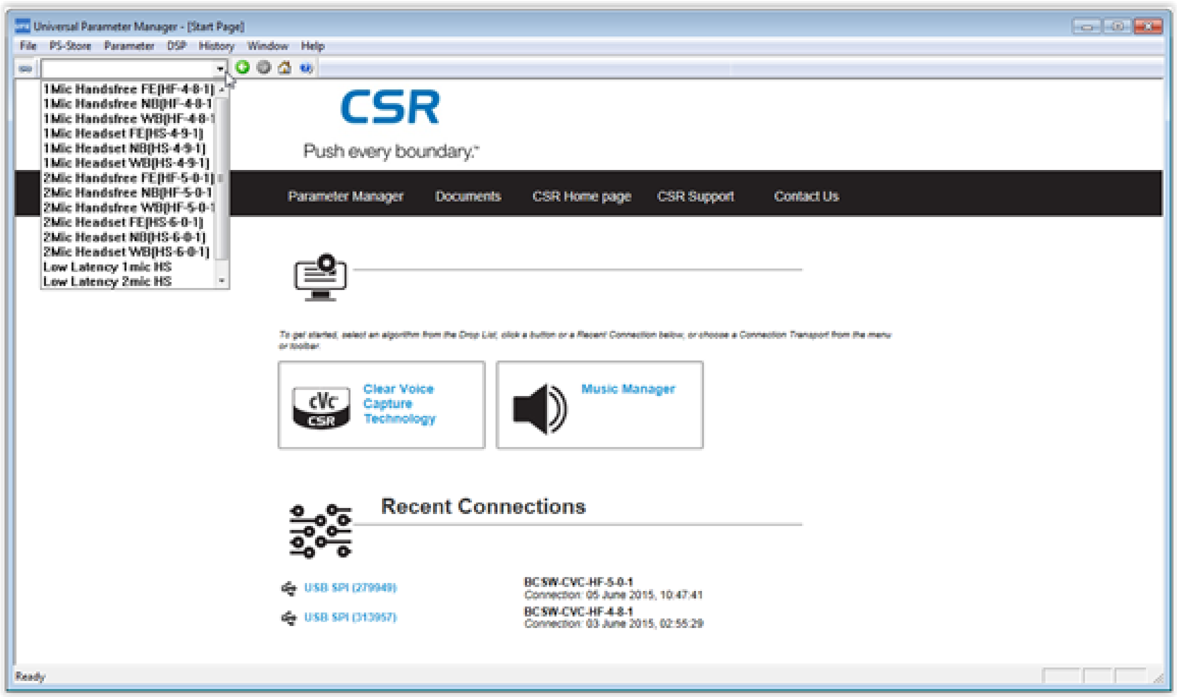


Figure 2-4 UFE application showing select an algorithm dropdown list (ADK)

The dropdown list contains the following options for CSR86xx devices:

- 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(HF-5-0-3)
- 2Mic Handsfree NB(HF-5-0-3)
- 2Mic Handsfree WB(HF-5-0-3)
- 2Mic Headset FE(HS-6-0-2)
- 2Mic Headset NB(HS-6-0-2)
- 2Mic Headset WB(HS-6-0-2)
- Low Latency 1mic HS

- Low Latency 2mic HS
- Music Manager

To select the 2-mic Handsfree Parameter Manager application, click **2Mic-Handsfree NB(HF-5-0-3)** algorithm from the dropdown list in Figure 2-4.

2.5 Connecting Parameter Manager via SPI

See the **Quick Start Guide** for descriptions on how to connect the Parameter Manager via a Serial Peripheral Interface (SPI) and to setup the security key, if required.

2.6 Security key setup

A security key protects the cVc library. When the ADK application is used, the BCSW-CVC-HF-5-0-3 mutes the audio until a valid security key is stored in the appropriate PS Key location. Contact a QTIL sales representative to learn more about obtaining valid keys for use in production devices.

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, *Enabling cVc for Headsets ADKs* describes how to activate the cVc algorithms for development purposes. When the Headset ADK is installed, navigate from the Windows **Start** menu to: **<ADK Name>\Documentation\ Support Documentation**

This opens the Support Document Index page from which the documents accompanying the ADK can be opened. Open the *Enabling cVc for Headset ADKs* document..

3 Handsfree Parameter Manager application

Table 3-1 Gain parameters and metrics

Option	Description
Gains: Adjustable Tuning Parameters	
MIC Gain	Analogue and Digital gain stage. Determines the gain applied to an incoming microphone signal. See MIC Gain.
SPKR Gain	Sets the overall gain of the DAC during the tuning process. This value is then placed into the volume table. See SPKR Gain

Table 3-2 Receive Path parameters and metrics

Option	Description
Receive Path Processing Parameters	
Packet Loss Concealment	This block improves the receive path audio quality in the presence of bit and packet errors within the Bluetooth link by using a variety of techniques such as pitch based waveform substitution. The user can bypass the processing block. See Packet Loss Concealment.
Noise Suppression	See the description for Send Noise Suppression. This block removes unwanted noise during hands-free conversation, cleaning the audio for the near end listener. See Noise Suppression.
Adaptive EQ	The Adaptive EQ improves speech intelligibility and loudness in quiet and in noisy environments. Three AEQ curves can be defined, and are dynamically transitioned depending on the level of near end noise. See Adaptive Equalizer.
Receive AGC	Automatic Gain Control (AGC) combined with audio dynamic range compression is provided on the Receive channel. 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. The AGC has multiple parameters that can be adjusted to obtain the required signal level with required response times. See Receive AGC.

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

Option	Description
Receive EQ	See the description for Send EQ. See Receive EQ.
Clipper	Pre-clips the reference signal before the echo canceller. This feature offsets any non-linearaties that occur after the echo canceller. The developer can control the amount of clipping. See Clipper.
Aux (Auxiliary) **Stream Mix**	Digital audio streams can be mixed into the receive path from other sources to enable ring tone and voice prompt playback, without interrupting the receive voice audio. The developer can control the mix of SCO audio and Auxiliary tones. See Auxiliary Stream Mix.
NDVC (Noise Dependent Volume Control)	The Speaker Gain can be controlled in 3 dB analogue steps so that the near-end volume adjusts according to the current noise level on the near-end. This block sets fixed gains according to standards and then programs a range that the volume increases further if near-end noise is present. This relies on the overall system having adequate speaker sensitivity. See NDVC.

Table 3-3 Send Path parameters and metrics

Option	Description
Send Path Processing Parameters	
Microphone Configuration	This block tells cVc how the microphones are placed in the vehicle. This enables the Noise Suppression algorithms to determine what is intended speech and what is unwanted noise. See Microphone Configuration.
Acoustic Echo Canceller	This block accesses the echo cancellation settings. It includes a sub-band adaptive linear filter that models the acoustic path from the receive reference point to the microphone input. It also provides a non-linear processing function that applies narrowband and wideband attenuation. This is done adaptively as a result of residual echo present after the linear filter. See Acoustic Echo Canceller.
Noise Suppression **(includes Beam Former, Blind Source Separation, Acoustic Event Detection, and Stationary Noise Suppression)**	Reduces noise with temporal characteristics uncorrelated with speech. The amount of noise suppression can be controlled to achieve optimum suppression versus voice distortion levels for the intended application. This block can remove unwanted noise during a hands-free conversation, cleaning the audio for the far end listener. See Noise Suppression.
Non-linear Processing	Non-linear processing helps to mitigate the echo distortion due to non-linearities of the loudspeaker and volume control. See Non-linear Processing.

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

Option	Description
Howling Control	Howling Control provides attenuation of the Send Out signal if the echo signal is extremely high. This switched the system to Half-Duplex when the acoustic echo exceeds a configurable threshold level. See Howling Control.
Comfort Noise	The Comfort Noise block mitigates the noise floor modulations introduced by the residual echo reduction, generated by the AEC. This feature can be parametrically bypassed and the gain controlled. See Comfort Noise.
Send EQ	Five-stage parametric and graphic equalization is provided for both the send and receive channels, which can be independently enabled and programmed to achieve the required frequency response. See Send EQ.
Send AGC	Maintains consistent listener experience regardless of the user speech level. This AGC has multiple parameters that can be adjusted to obtain the required signal level with required response times. See Send AGC.

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

NOTE Various statistics are displayed and updated during an active call, such as Peak and Noise Level.

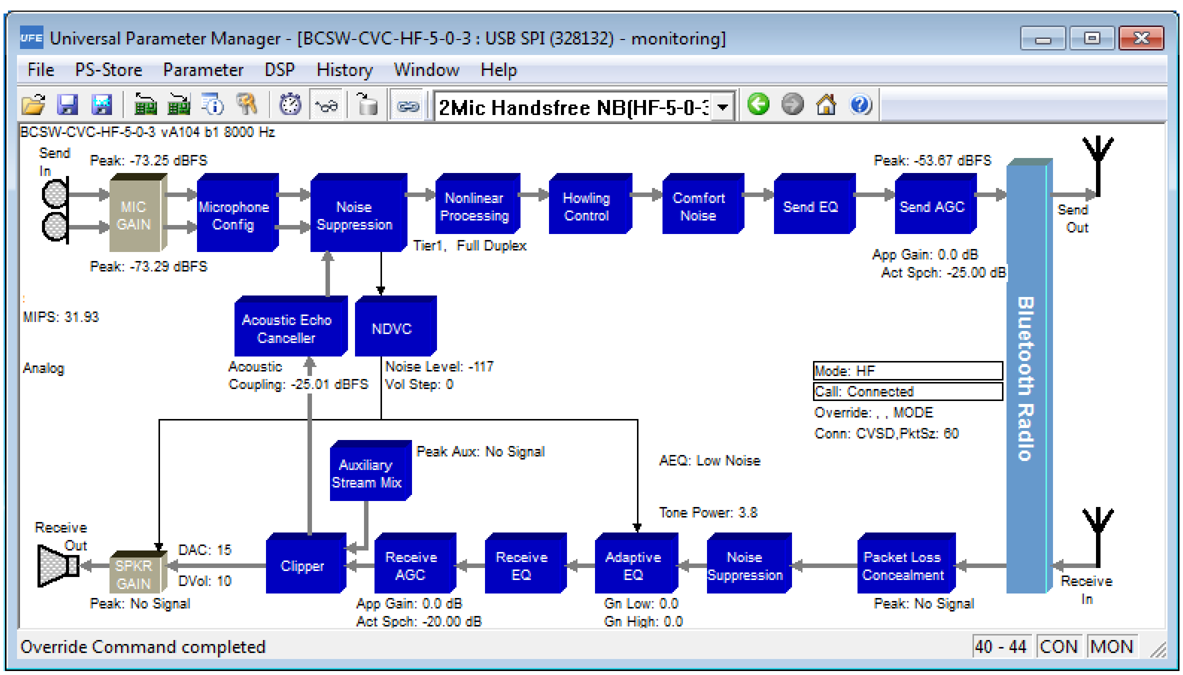


Figure 3-1 Parameter Manager window showing all active modules

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 greyed out.

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

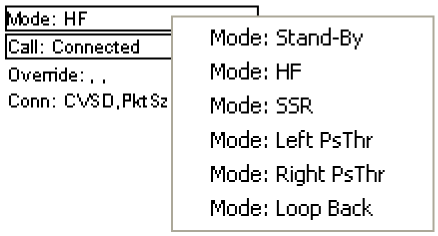


Figure 3-2 Selecting an override mode

Overrides modes are:

- Stand-by: Processing in standby, no audio flow
- HF: Full handsfree processing
- SSR: Simple Speech Recognition
- Left PsThr: Left microphone pass-through
- Right PsThr: Right microphone pass-through
- Loop Back: Loop microphone to speaker, ADC to DAC

Figure 3-3 shows example views of the Parameter Manager in different monitoring override modes. From left to right, top to bottom the modes are Stand-By, SSR, Pass Through, and Loop Back.

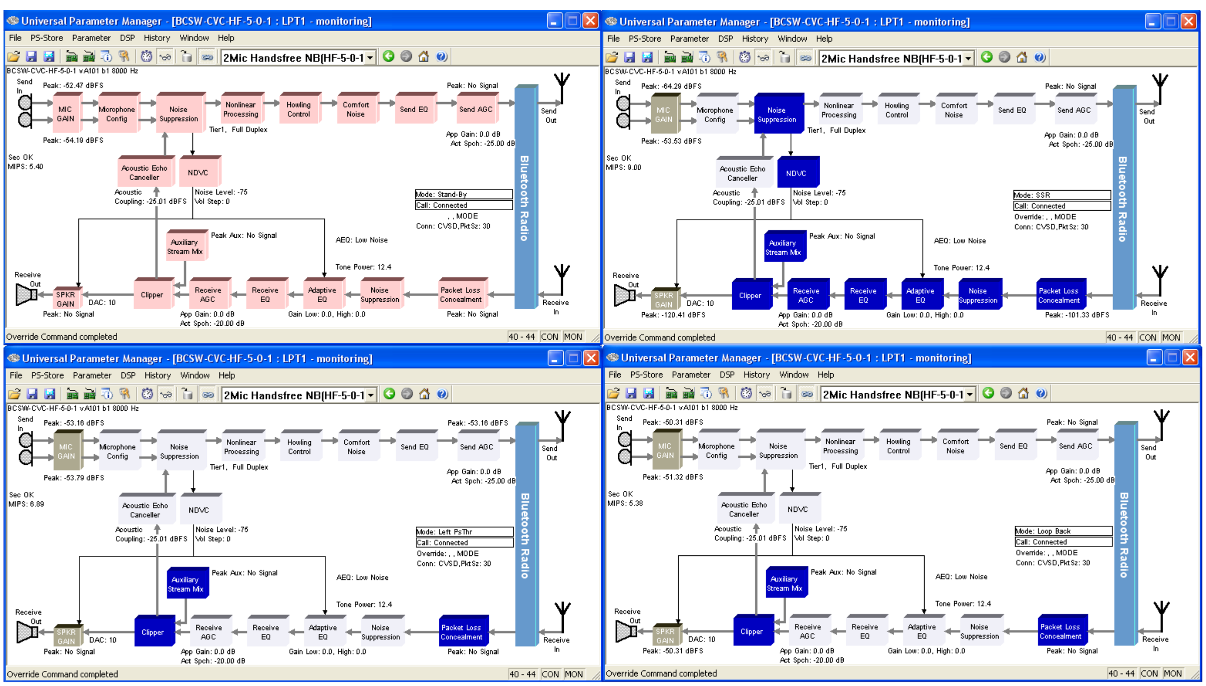


Figure 3-3 Sample override mode examples during monitoring

4 Using the gain 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.

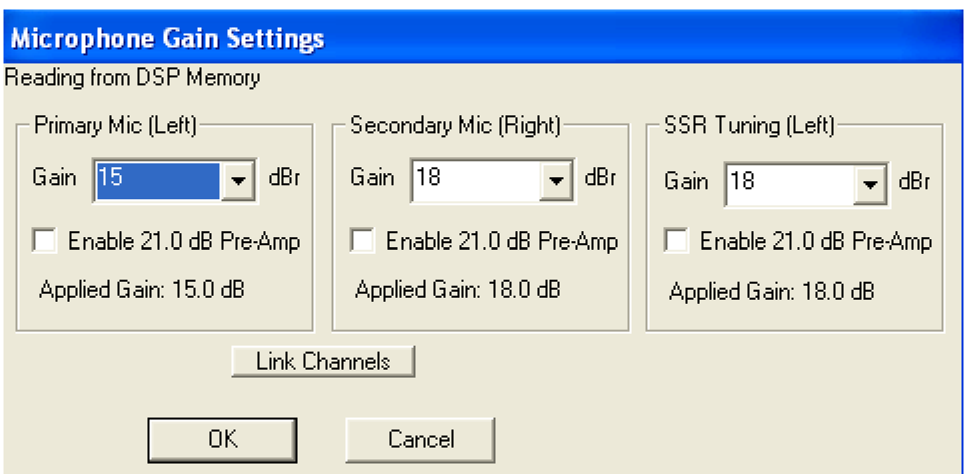


Figure 4-1 Microphone Gain Settings window

2. This window controls the ADC gains of the Left and Right Microphones and the SSR gain.
3. The **Microphone Gain Settings** window displays with the current **Gain dBr** value and an **Enable 21.0dB Pre-Amp** option. The sum of these values displays in the **Applied Gain: xx.x dB** read-only area, which the software calculates and uses.
4. Select the required settings.

Table 4-1 Microphone Gain Settings options

Option	Description
Gain	Use the drop-down arrow to select the required gain setting for HFK mode. This setting reflects a combination of analogue gains (black text) and digital gains (red text).
Enable 21.0dB Pre-Amp	When this check box is unchecked (default), this means that the microphone pre-amplifier is not enabled. Check the boxes to enable the microphone pre-amplifier to apply analogue 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.

5. After selecting the required settings, click **OK**.

4.2 SPKR Gain

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

NOTE To set the Speaker Gain in the final configuration, the volume table in the Headset VM application must be edited.

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.

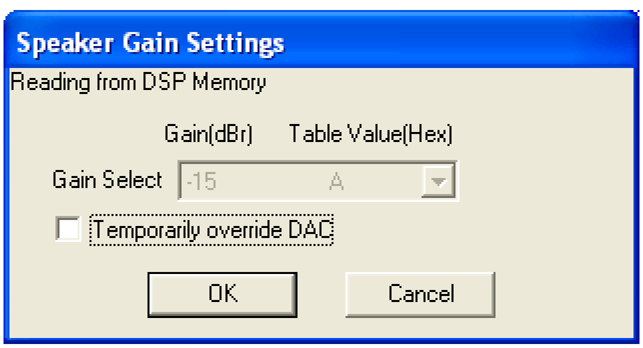


Figure 4-2 Speaker Gain Settings window

2. Select the required settings.

Table 4-2 Speaker Gain Settings options

Option	Description
Gain Select	The default Gain Select value is a read-only value based on the phone volume index. After the phone volume is changed, closing and reopening the SPKR GAIN block updates the read-only Gain Select value. To manually enter a Gain Select value: 1. Check the Temporarily override DAC option box 2. Select the appropriate value in the Gain Select field. A volume change can override the value in the Gain Select field.
Temporarily override DAC	Enables the Gain Select field, which enables a manual entry. A volume change can override the value in the Gain Select field.

5 Using the Receive Path audio tuning controls

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

- Packet Loss Concealment
- Noise Suppression
- Adaptive EQ with Frequency Emphasis or Frequency Expansion
- Receive AGC
- Receive EQ (Parametric Equalization for loudspeaker frequency correction)
- Clipper (includes Boost)
- Aux Stream Mix
- NDVC

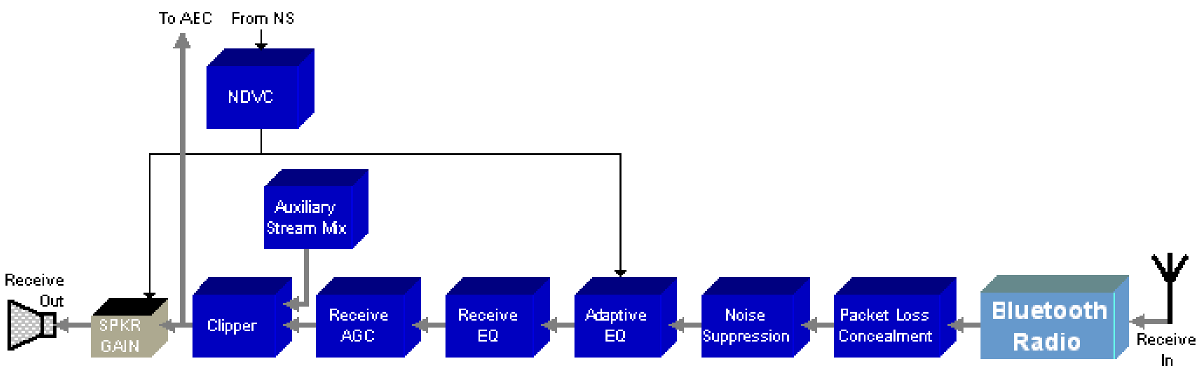


Figure 5-1 Receive Audio path processing blocks

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 a variety of techniques such as pitch based waveform substitution.

To Adjust the Packet Loss Concealment Settings, no tuning is provided. Check the **Bypass Packet Loss Concealment** to disable the module.

NOTE To achieve the best quality audio, always enable the Packet Loss Concealment.

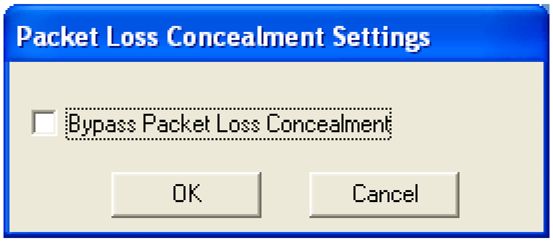


Figure 5-2 Packet Loss Concealment Settings window

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.

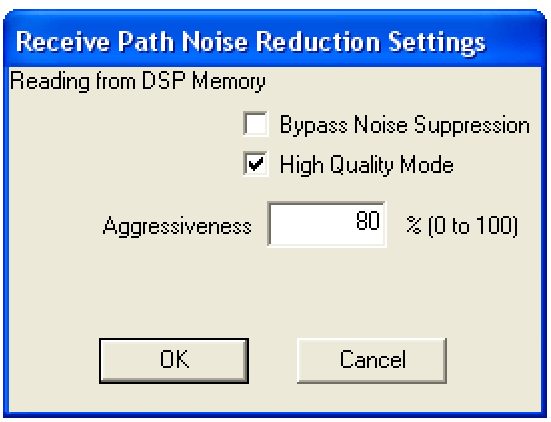


Figure 5-3 Noise Suppression Settings window

2. Select the required settings based on the options.

Table 5-1 Noise Suppression Settings options

Option	Description
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. The receive signal has been processed by the cellular network, and transmitted over Bluetooth so to avoid over processing the voice, our recommendation is to set the Aggressiveness ≤ 80%.
Bypass Noise Suppression	If checked, bypasses the NS feature, reducing processor cycles.
High Quality Mode	If checked, invokes additional algorithm processing improving the quality of the voice.

3. Click **OK**.

5.3 Adaptive Equalizer

The Adaptive EQ improves speech intelligibility and loudness in quiet and, more important, noisy environments. When the Adaptive EQ block is enabled it 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 window, the AEQ speech signal is divided into three different frequency regions:

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

The AEQ has a fixed power ratio for the ref band. The user sets the low and high band thresholds 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 the receive speech. The curves transition based on the near end noise level as measured by the NDVC.

NOTE For the Handsfree device to benefit from this feature, the loudspeaker must provide adequate fidelity delivered to the user’s ear.

To adjust the Adaptive EQ Settings:

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

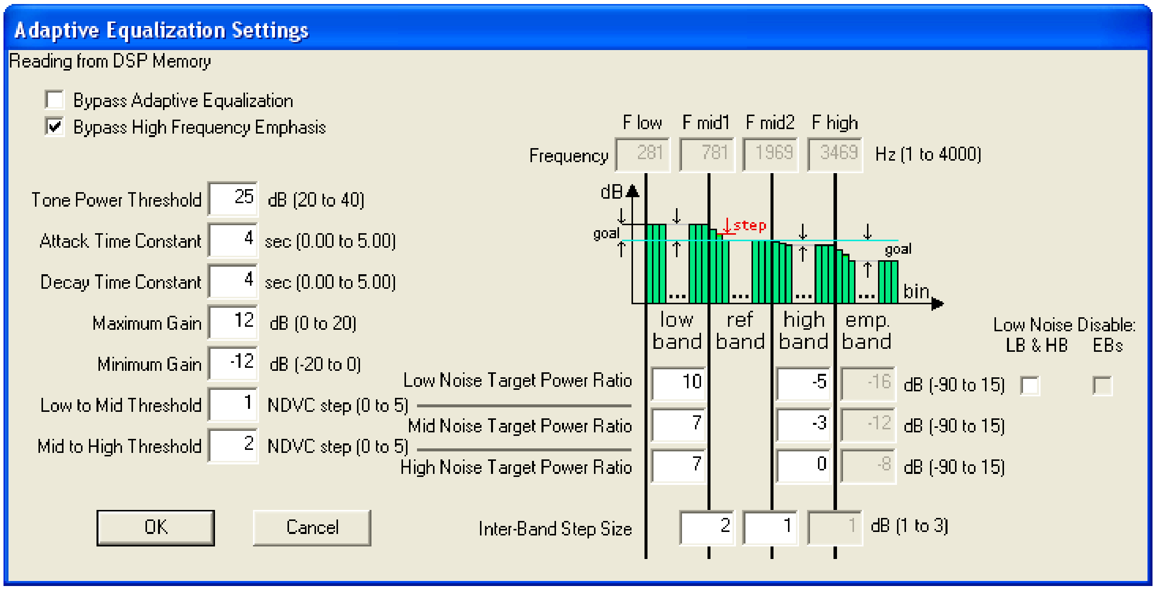


Figure 5-4 Adaptive Equalization Settings window for narrow band

2. Select the required settings.

Table 5-2 Adaptive Equalization Settings options

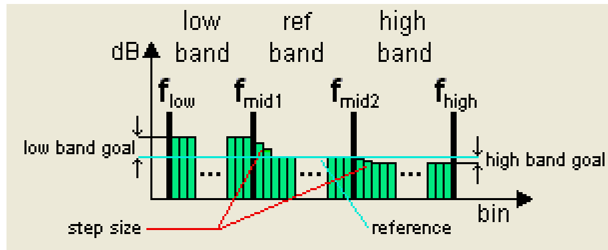
Option	Description
Tone Power Threshold	If a tone appears in the receive audio path, the AEQ or the NS should not adapt during the tone. This causes distortions in the tone. Based on the Tone Power Statistic, set the threshold to identify a tone. When the Tone Power Threshold has been exceeded, the Tone Detected statistic appears. Setting the threshold to low may cause some speech to be falsely detected. The AEQ and NS does not operate which may cause speech distortion. Setting the threshold to high may cause some tones to be 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 has more than this gain value applied.
Minimum Gain	Sets the low threshold limit for the gain applied to any frequency bin. No output has less than this gain value applied.
Low to Mid Threshold	Sets 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 This field is only effective if the NDVC is enabled and the DAC has available headroom.

Table 5-2 Adaptive Equalization Settings options (cont.)

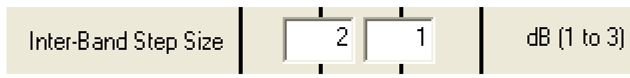
Option	Description
Mid to High Threshold	Sets the NDVC step at which the AEQ switches from the Mid to High Noise Target Power Ratio curve. The switch point is based on the Vol Step statistic. When switched the AEQ statistic reads AEQ: High Noise This field is only effective if the NDVC is enabled and 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.  Two values are user defined the low band and high band goals. Enter the value in dB, 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.  Two values are user defined, the low band and high band goals. Enter the value in dB and 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.  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 Equalization Settings options (cont.)

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



Two values are user defined. The first value defines the step size used to smooth 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, you can disable the AEC frequency shaping for the low noise condition only, to preserve speech quality. If near end noise is present the AEC 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 compliment the AEQ, High Frequency Emphasis or Frequency Expansion can be added to improve the intelligibility of the far end caller. See the *BCSW-CVC-HF-5-0-3 2M-HF Tuning Guide Guide* for details.

This section defines the relative additions of the base AEQ shown in Figure 5-4.

5.3.1.1 Adjusting the Adaptive EQ with High Frequency Emphasis Settings

High Frequency Emphasis can be used with any standard narrow band 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 do this:

1. Select **2Mic Handsfree NB[HS-5-0-3)** from the UFE drop list or make a narrow band call, and place the UFE into monitoring mode.
2. From the **Parameter Manager** window, click the **Adaptive EQ** block. The **Adaptive Equalizer Settings** window opens.

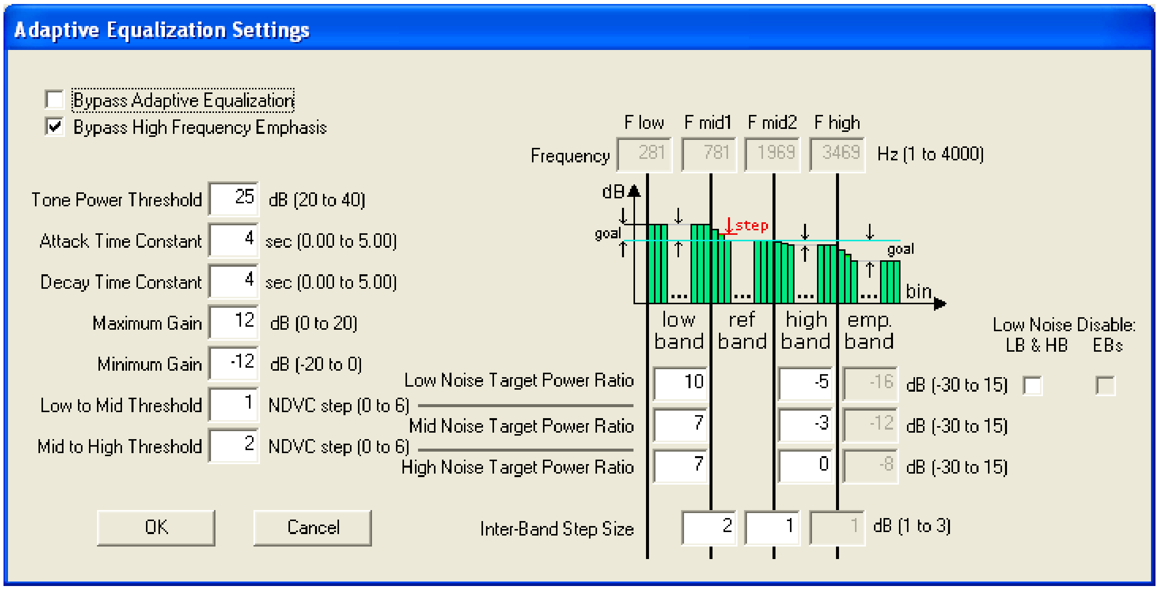


Figure 5-5 Adaptive EQ with High Frequency Emphasis Settings window

3. Select the required settings.

Table 5-3 Adaptive EQ with High Frequency Emphasis Settings 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, in an effort 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 **Med Noise Target Power Ratio** **High Noise Target Power Ratio** **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.

5.3.1.2 Adjusting the adaptive EQ with Frequency Expansion settings

Frequency Expansion can be used with any standard narrow band call, but a special mode is invoked where 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 do this:

1. Choose **2Mic Handsfree Freq. Exp.** from the UFE drop list or make a narrow band 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.

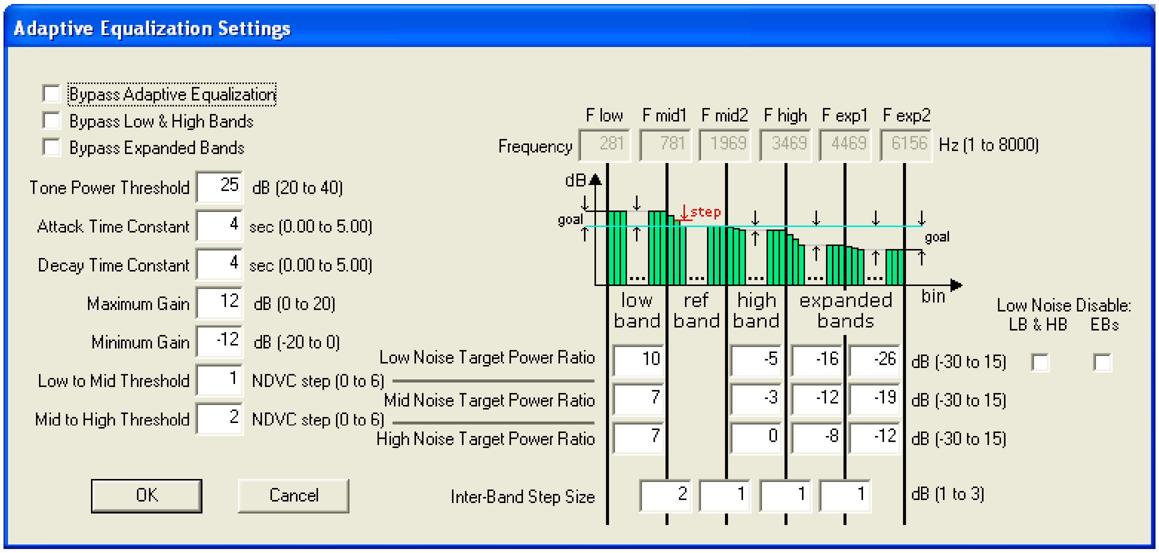


Figure 5-6 Adaptive EQ with Frequency Expansion Settings window

3. Select the required settings.

Table 5-4 Adaptive EQ with High Frequency Emphasis Settings 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, in an effort 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, in an effort 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 **Med Noise Target Power Ratio** **High Noise Target Power Ratio** **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 AGC

To adjust Receive AGC Settings:

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

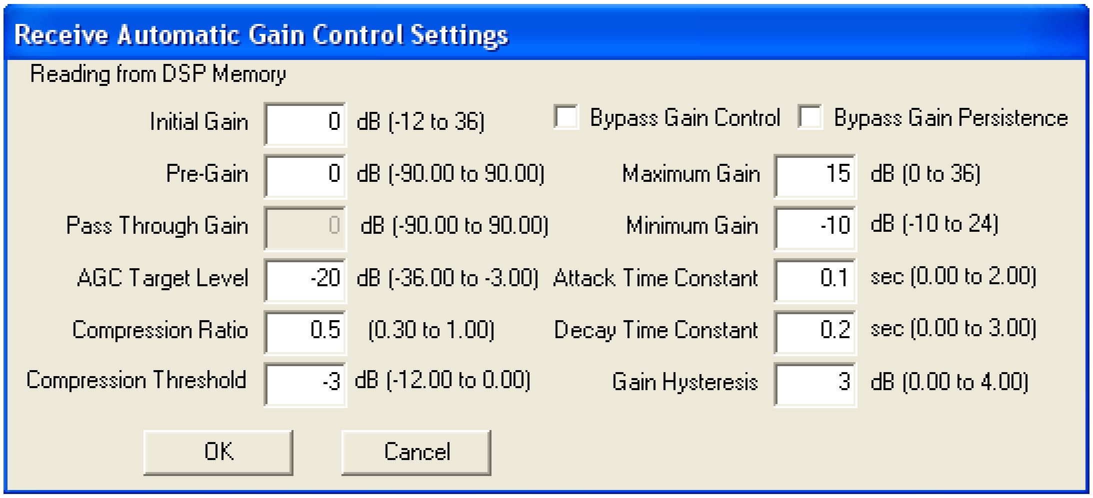


Figure 5-7 Receive Automatic Gain Control Settings windows

2. Select the required settings.

Table 5-5 Receive AGC Settings 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 PS Key to be used at the initiation of a new call, helping the Receive AGC adapt quicker at start up. Check this option to disable the Calibration Persistence option.
Initial Gain	The applied gain of the AGC when cVc is first initialised.
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 because the other blocks are bypassed. This compensates for the loss of the AGC block. It is also used when toggling between HFK and PT modes, or used for the LBIPM (Low Battery Intelligent Power Management) 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 decreases compression increases.
Compression Threshold	The point at which compression begins (peak from full scale).
Maximum Gain	Sets the high threshold level for the gain. The AGC acts to maintain this value as the maximum gain level. No output has more than this gain value applied.
Minimum Gain	Sets the low threshold level for the gain. The AGC acts to maintain this value as the minimum gain level. No output has less than this gain value applied.
Attack Time Constant	Sets the rate of decreasing gain (attenuation). If the AGC gain is too high and needs to decrease faster, set the Attack Time Constant lower, which speeds up the rate of change.

Table 5-5 Receive AGC Settings options (cont.)

Option	Description
Decay Time Constant	Sets the rate of 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 reacts more slowly.
Gain Hysteresis	Sets the upper and lower boundaries for the gain to change. For example a value of 4 means the AGC adjusts only when the speech signal has changed by 4 or more dB above or below the target level.

3. Click **OK**.

5.5 Receive EQ

The Receive EQ parameter is a graphical user interface that alters the frequency response by configuring up to five bi-quad 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.

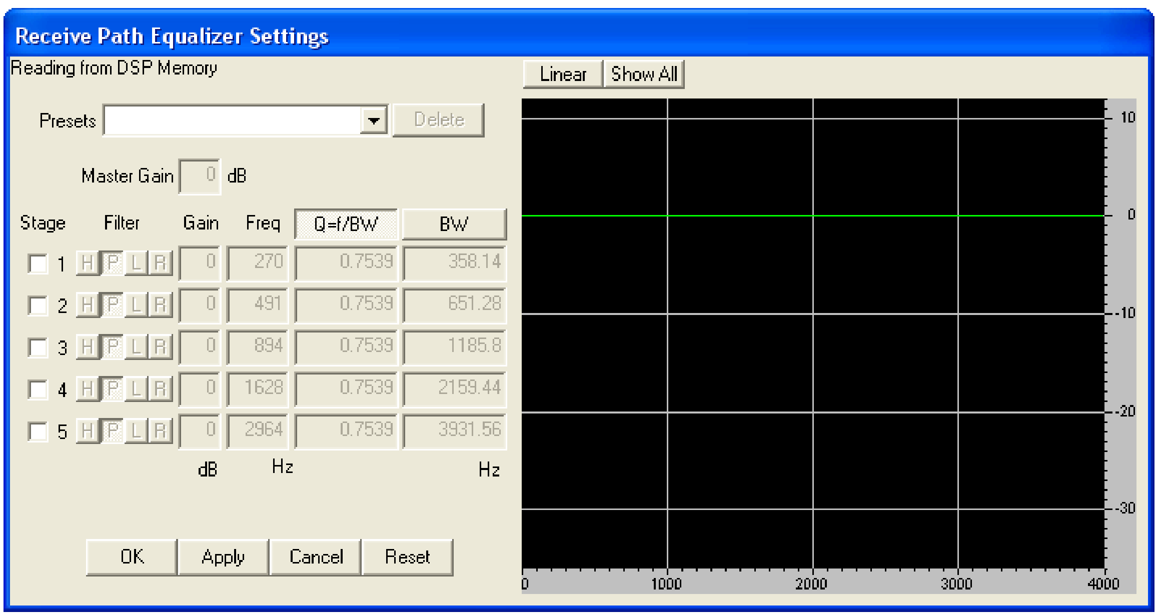


Figure 5-8 Receive Equalizer Settings window

2. The current state of the screen is the default state before presets are selected (the **Presets** field is blank by default). The EQ interface supports multiple views of the equalizer without needing to close the window.
3. Select the required settings.

Table 5-6 Equalization Settings options

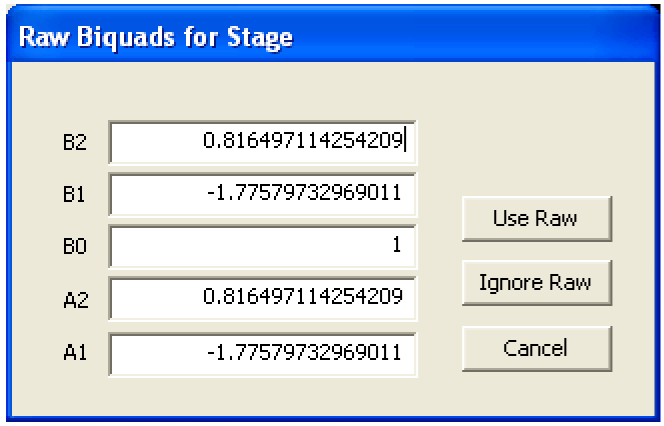
Option	Description
Presets	 This dropdown list provides a collection of sample equaliser filters. Available options include: ■ Default ■ Low Boost ■ Mid Softener ■ Gentle High Boost ■ V curve ■ 5 State Graph EQ Each Preset option can be altered using the five stages listed in the Graphic EQ/Parametric 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. User-defined Presets can be selected from the Presets dropdown list. Once selected, the Delete button is activated. If Delete is pressed, the Preset is deleted from the dropdown list and the registry. It is not recoverable.
Linear / Log	This option controls the appearance of the plot area on the screen. Click to toggle the curve plot between the Linear (default) and Log (Logarithmic) views.
Show All	This toggle option enables the plotting of the individual stages as well as the combined filter. Select **Show All** to display 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. The range that can be entered in this field is -90 to 12 dB.
Stage	The Stage checkbox enables the user to define the number of bi-quad stages to use in the equalizer filter configuration.

Table 5-6 Equalization Settings options (cont.)

Option	Description
Filter	The Filter option enables the Stage Filter to be set to H, P, L or R i.e. High Pass, Parametric, Low Pass and Raw. A filter type can be set for each stage enables creation of complex curves. ■ High Pass ■ Parametric ■ Low Pass ■ Raw If Use Raw is selected you can enter the stage coefficients directly using the GUI provided. Parametric mode: The fields in the Gain, Freq, Q=f/BW and BW columns are editable. Data can be entered or selected 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 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. ■ Q: Sharpness of curve/quality ■ f: Frequency ■ BW: Bandwidth
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.
Reset	Click this button to create a filter curve from the last saved state i.e. the state last saved by clicking **OK**. 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. Changes made in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory. The **Apply** button can be used to write unlimited parameters to the DSP memory without closing the Receive Equalizer Settings window. In the Static mode, this button is not available (greyed 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. Changes made in the Monitoring mode only affect the DSP and PC memory, not the Persistent Store memory. In the Static mode, this button is not available (greyed 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	

Table 5-6 Equalization Settings options (cont.)

Option	Description
Zoom In	Use the mouse on the grey Scale bar in the plot area to select a starting zoom point. 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. Release the mouse button. The horizontal scale only drags if the scale is zoomed.

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 pre-clipped 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, causing 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.

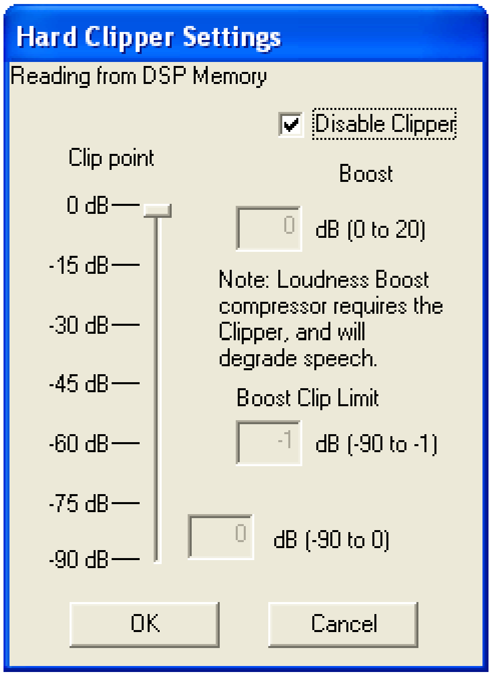


Figure 5-9 Hard Clipper Settings window

The available Clipper variables are:

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

2. Select the required settings.

Table 5-7 Clipper Settings options

Option	Description
Disable Clipper	By default, this option is checked, and the Clipper is disabled. Uncheck to enable the Clipper. 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. All other parameters are disabled unless the Disable Clipper check box is unchecked.
Clip point (slider) (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. 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. 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. Important Note: 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.

Table 5-7 Clipper Settings options (cont.)

Option	Description
Boost (Loudness Boost)	The Loudness Boost is a pure digital gain applied to the receive signal, above the saturation point. This setting raises the loudness at the loudspeaker without overdriving the saturation point for the loudspeakers. Increasing the gain in the loudness boost can further degrade the audio quality. To set the loudness boost, type the appropriate setting in this field (the range is from 0 dB to 20 dB).
Boost Clip Limit	The Boost Clip Limit sets the maximum scale a receive signal is allowed to achieve, while avoiding the saturation (hard limiter). 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. Lowering the Boost Clip Limit further degrades the audio quality. 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. Important Note: 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

e **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 opens.

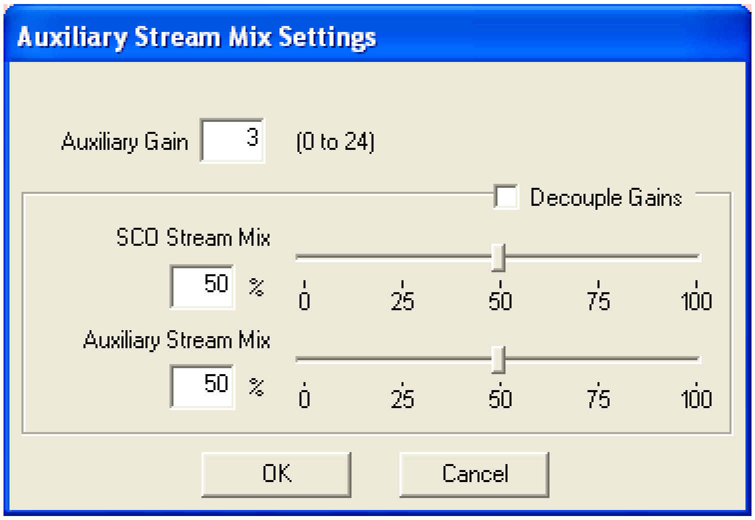


Figure 5-10 Auxiliary Stream Mix Settings window

2. Select the required settings.

Table 5-8 Auxiliary Stream Mix Settings options

Option	Description
Auxiliary Gain	Sets the amount of gain that is applied to the Auxiliary Signal. To set the Auxiliary Gain, type the appropriate setting in this field. 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. 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 NDVC

The **Noise Dependent Volume Control** 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.

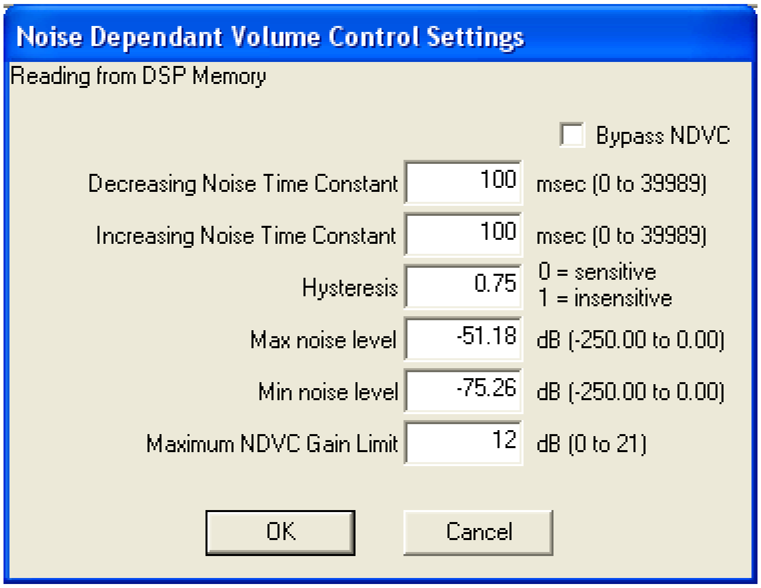


Figure 5-11 Noise Dependent Volume Control Settings window

2. Select the required settings.. See the associated Tuning Guide for alternative settings.

Table 5-9 Dependent Volume Control Settings options

Option	Description
Bypass NDVC	By default, the Bypass NDVC option is checked. The NDVC feature is disabled and the fields on this screen are read-only (greyed out). When this option is unchecked the NDVC feature is enabled and the fields on this screen are available.
Decreasing Noise Decay Time Constant	Sets the decay time constant used to decrease the volume steps based on the noise level.
Increasing Noise Attack Time Constant	Sets the attack time constant used to increase the volume steps based on the noise level.
Hysteresis	Sets the sensitivity when switching between adjacent volume states (Range 0.00 to 1.00). Lowering the value gives higher sensitivity. For example a value of 0.75 means the hysteresis is less sensitive than when set to 1.00.
Max noise level	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	Sets the noise level threshold at which the NDVC adds the minimum gain i.e. no Gain is added.
Maximum NDVC Gain Limit	Limits the maximum gain that the NDVC applies. The system code must be set up 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**.

6 Using the Send Path tuning controls

The Parameter Manager provides access to these send path tuning controls:

- Microphone Configuration
- Acoustic Echo Canceller
- Noise Suppression
- Non Linear Processing
- Howling Control
- Comfort Noise
- Send EQ
- Send AGC

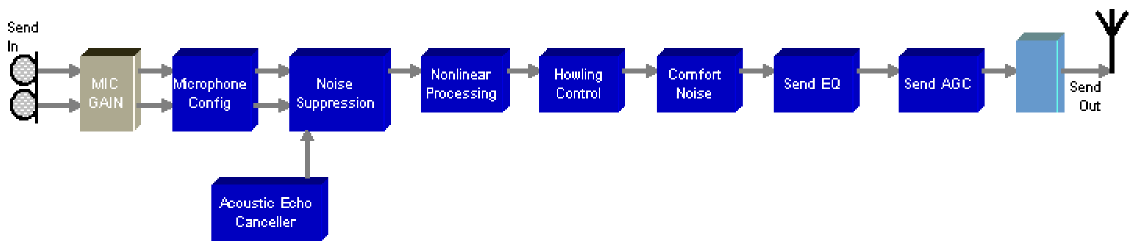


Figure 6-1 Send Path tuning controls

6.1 Microphone Configuration

The Microphone Configuration block defines how the microphones are configured in the system.

To access The Microphone Configuration Settings:

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

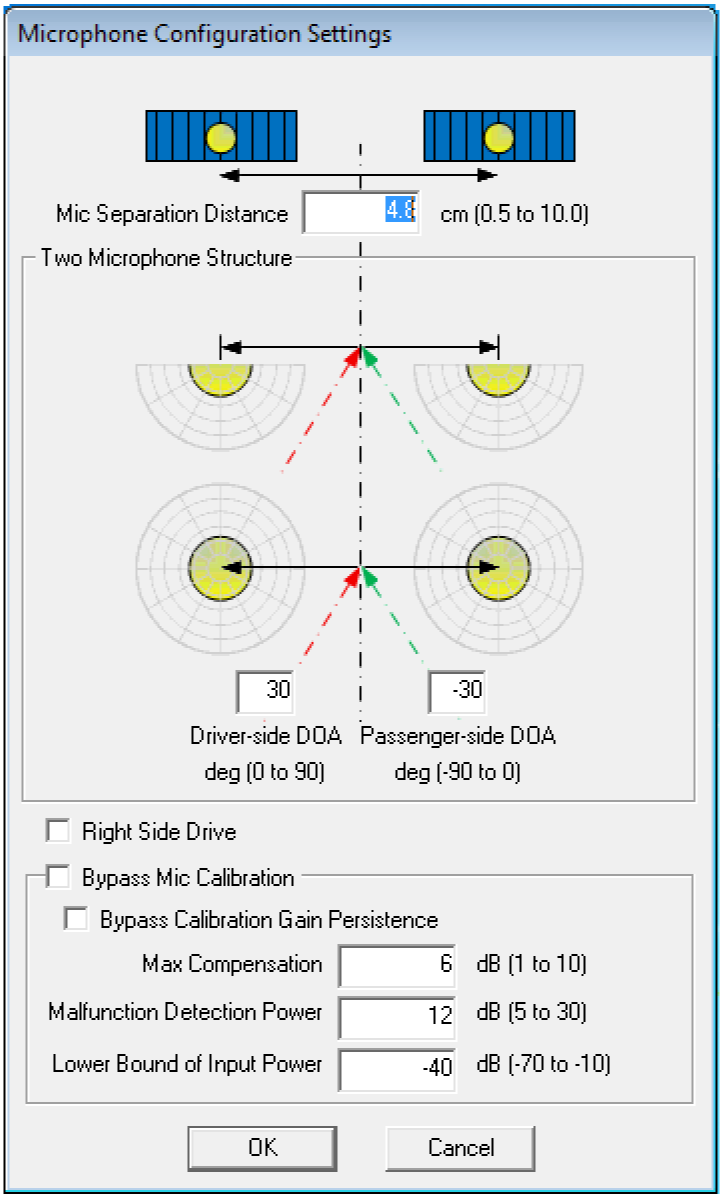


Figure 6-2 Microphone Configuration Settings window

2. Select the required settings.

Table 6-1 Microphone Configuration Settings options

Option	Description
Mic Separation Distance	Distance between microphones as measured from their centre points.
Driver-side DOA	Direction of arrival of driver speech to the centre point between the two microphones.
Passenger-side DOA	Direction of arrival of passenger speech to centre point between the two microphones.
Right Side Drive	Check box if vehicle is right hand drive.

Table 6-1 Microphone Configuration Settings options (cont.)

Option	Description
Bypass Mic Calibration	The Mic Calibration adjusts the gain of the secondary microphone if there is a difference between the left and right microphones sensitivity. Check this option to disable the Microphone Calibration
Bypass Calibration Gain Persistence	The Calibration Gain Persistence stores the last known microphone gain compensation in a PS Key to be used at the initiation of a new call thus helping the 2-mic algorithm adapt quicker at start up. Check this option to disable the Calibration Persistence option.
Max Compensation	Maximum gain which can be applied to secondary microphone to adjust for difference in microphone sensitivity between left and right microphones.
Malfunction Detection Power	Malfunction Detection Power is the threshold which if exceeded the application assumes one of the microphones has completely malfunctioned and disables the 2-mic portion of the algorithm.
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.

6.2 Acoustic Echo Canceller

The Acoustic Echo Canceller reduces echo 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.

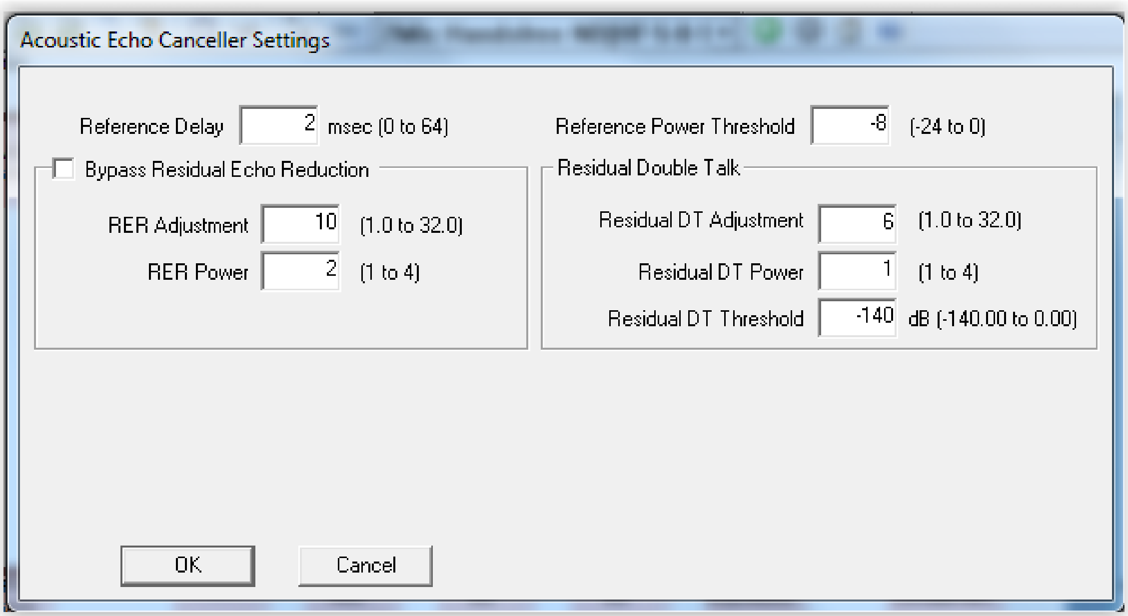


Figure 6-3 Acoustic Echo Cancellation Settings window

2. Select the required settings.

Table 6-2 Acoustic Echo Cancellation Settings options

Option	Description
Bypass Residual Echo Reduction	By default, this option is unchecked, which means the Residual Echo Reduction is enabled. This feature is most beneficial when there are no non-linearities in a system and the acoustic coupling of the system is minimal. Check this option to disable the residual echo reduction. Disabling the residual echo reduction saves ~1.4 MIPS.
Reference Power Threshold	Once the receive energy exceeds this level the AEC adapts right away in any case. Otherwise the AEC only adapts when the receive signal is not stationary. This helps the AEC from diverging.
Reference Delay	Reference Delay is a delay buffer that compensates for the latency in the signal as it travels: □ From the DAC to the loudspeaker □ Over the acoustic enclosure □ To the ADC □ Send path data buffering before AEC filter This delay in milliseconds is presented to the AEC as the echo component in the Microphone signal. Type the appropriate reference delay setting in this field if other than the default is required. 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. If the entered delay is shorter than the actual latency, the algorithm's effective echo tail length is reduced.
RER Adjust	The RER Adjustment controls the amount of attenuation during receive only speech. Increasing this value increases double talk performance but can degrade single talk echo cancellation performance.
RER Power	The RER Power controls the number of times the RER Attenuation is applied during receive only speech. Increasing this parameter raises the number of times RER attenuation is applied.
Residual DT Adjustment	The RER Adjustment is used to control the amount of attenuation during double talk. Increasing this value increases double talk performance but can degrade single talk echo cancellation performance.
Residual DT Power	The RER Power controls the number of times the RER Attenuation is applied during double talk. Increasing this parameter raises the raises the number of times 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 are used.

3. Click **OK**.

6.3 Noise Suppression

The Noise Suppression block defines both the aggressiveness of the 2-mic Handsfree noise suppression algorithm and stationary noise reduction algorithm.

To set Noise Suppression Options:

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

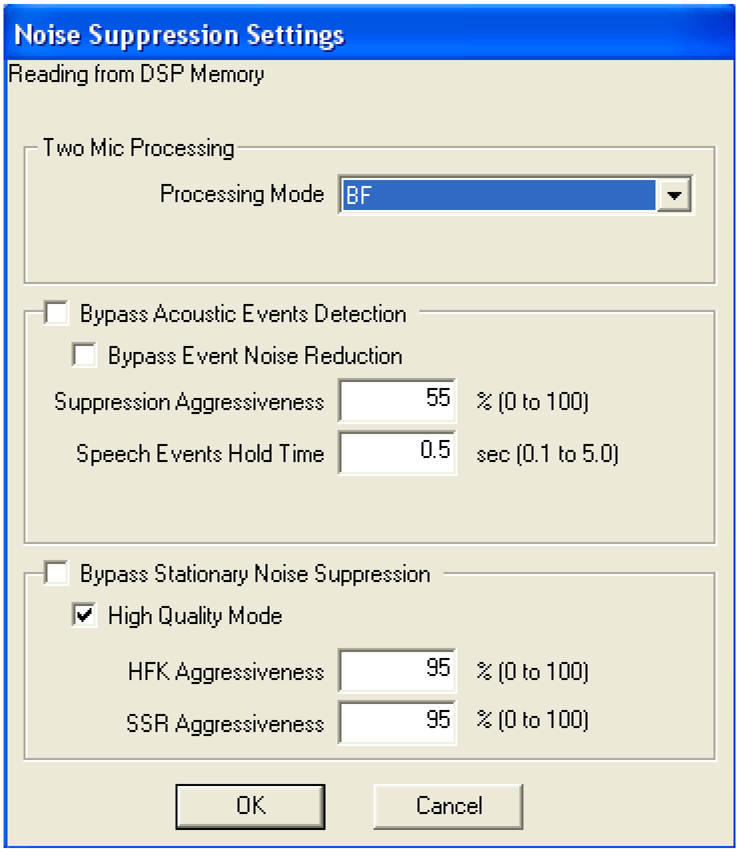


Figure 6-4 Noise Suppression Settings window

2. Select the required Noise Suppression settings.

Table 6-3 Noise Suppression Settings window options

Option	Description
Two Mic Processing Mode	Selects which parts of the 2-mic processing modes is used. Beam Former (BF) is always enabled. Blind Source Separation (BSS) and Noise Canceller (NC) can be added.
Bypass Acoustic Events Detection (AED)	The Acoustic Event Detection performs additional noise reduction only when there is no target speech detected. If checked, the AED is bypassed.
Bypass Event Noise Reduction	If checked, bypasses the noise reduction feature of the AED.
Suppression Aggressiveness	Controls the amount of suppression applied by the AED.
Speech Events Hold Time	Transition time from target speech event to noise event. The larger this value is, the slower of transition from target speech to noise.

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

Option	Description
Bypass Noise Suppression	If checked, bypasses the NS feature reducing processor cycles. Important Note: Other modules are dependent on intelligence within the Send Noise Suppression bloc. Bypassing may cause system performance to degrade.
High Quality Mode	If checked, invokes additional algorithm processing improving the quality of the voice. Unchecked results in minor reduction in processor cycles.
HFK Aggressiveness	Controls the amount of noise suppression applied to the send signal during main processing HFK mode. A good balance between voice quality and high noise suppression is achieved by setting the HFK Aggressiveness to 95% which yields >17 dB of SNR improvement (using pink noise). Setting this parameter to 100% maximizes noise suppression.
SSR Aggressiveness	Controls the amount of noise suppression applied to the send signal during the SSR mode of operation. Setting this parameter to 100% maximizes noise suppression.

3. Click **OK**.

6.4 Non-linear Processing

The Non-linear Processing block compensates for non-linearities introduced into the receive path. Changes in the volume that introduce non-linear effects into the acoustic path are compensated for when either the Tier 1 or Tier 2 sets of heuristic non-linear processing are used.

To use the Non-linear Processing Option:

1. From the **Parameter Manager** window, select the **Nonlinear Processing** block. The **Nonlinear Processing Settings** window opens.

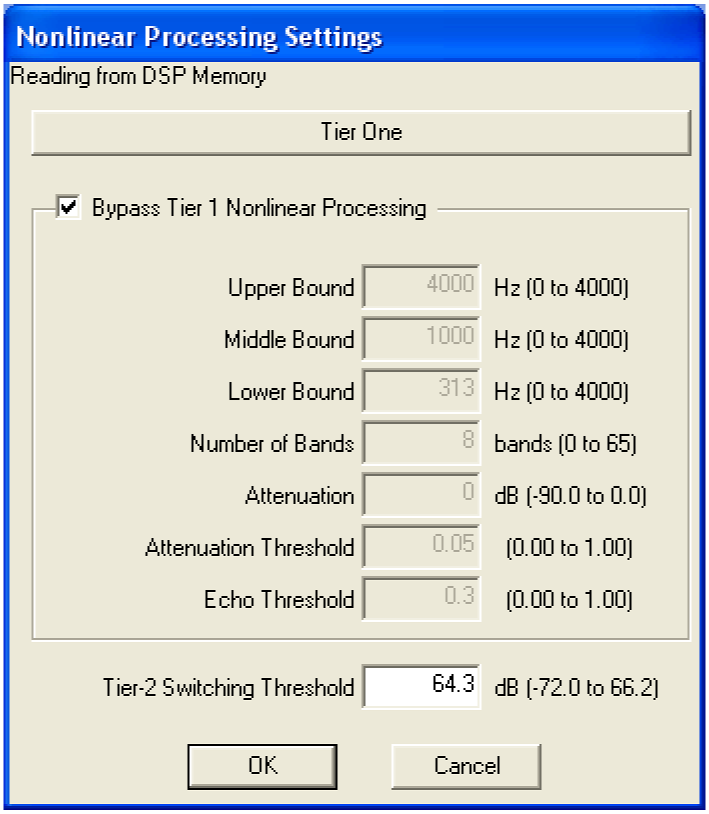


Figure 6-5 Nonlinear processing Settings window

Table 6-4 Nonlinear Processing Settings options

Option	Description
Tier 1/Tier 2	This option toggles between the Tier 1 and Tier 2 non-linear processing features
Bypass Tier 1 Nonlinear Processing Bypass Tier 2 Nonlinear Processing	By default, the **Bypass Tier 1 Nonlinear Processing** option is checked and non-linear processing is disabled. In this state, all the fields in this area on the screen are read only (i.e. greyed out) To choose non-linear processing, uncheck the **Bypass Tier 1 Nonlinear Processing** option. The user can: ■ Adjust the bandwidths of the three bounds: □ Lower Bound □ Middle Bound □ Upper Bound ■ Define the number of bins over which to average residual attenuation (limited by Middle Bound- Lower Bound + 1) ■ Select the maximum attenuation to be applied to the **Send Out** signal when echo is present.

Table 6-4 Nonlinear Processing Settings options (cont.)

Option	Description
Lower Bound, Middle Bound and Upper Bound	Identification of the parts of the spectrum that contain residual echo and require attenuation is beneficial when adjusting these parameters. The **Nonlinear Processing** algorithm compares the **Lower Bound** and the **Middle Bound** parameters for the average value and compares it to the internal **Attenuation Threshold**. Based on this average, the value in the **Attenuation** field is applied between the **Middle Bound** and the **Upper Bound**.
Number of Bands	The **Number of Bands** computes the average value between the **Lower Bound** and the **Middle Bound**. Adjusting the **Number of Bands** is a trade-off of sensitivity versus robustness. Lowering the **Number of Bands** makes it more sensitive and less robust.
Attenuation	This sets the maximum amount of attenuation applied between the Middle Bound and the Upper Bound when the **Nonlinear Tier(s)** have been triggered.
Attenuation Threshold	The **Attenuation Threshold** is compared against average values for residual attenuation between **Lower Bound** and **Middle Bound**. If the average residual attenuation is less than this threshold then attenuation is applied between the **Middle Bound** and **Upper Bound**. Increasing the **Attenuation Threshold** makes it more likely for attenuation to be applied.
Echo Threshold	The **Echo Threshold** is decides whether or not there is an echo event. Increasing the **Echo Threshold** makes it less likely to recognize an echo event. **Nonlinear Processing** is only triggered if and echo event is recognized.
Tier-2 Switching Threshold	The transition between the Tier 1 module and the Tier 2 module is set by the value set in the **Tier-2 Switching Threshold** field. Tier 2 is typically used for higher speaker volume levels. Tier 1 is used for lower volume levels. Both Tier 1 and Tier 2 can be selected.

6.5 Howling Control

The **Howling Control** block enables the howling control suppression parameters to be set.

To use the Howling Control:

1. From the **Parameter Manager** Window, select the **Howling Control** block. The Howling Control Settings window opens.

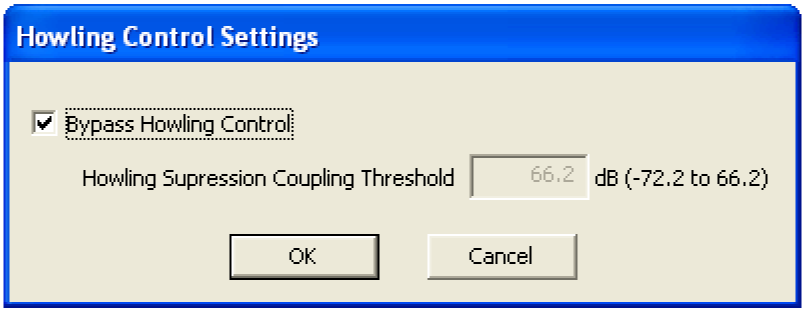


Figure 6-6 Howling Control Settings window

2. Select the required settings.

Table 6-5 Howling Control Settings options

Option	Description
Bypass Howling Control	By default, this option is checked and the feature is disabled. 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).

6.6 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.

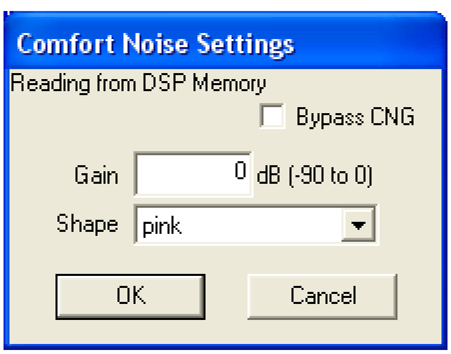


Figure 6-7 Comfort Noise Settings window

2. Select the required settings.

Table 6-6 Comfort Noise Settings options

Option	Description
Bypass CNG	This check box enables or disable the comfort noise gain control, and during the tuning process, to make sure the proper amount of comfort noise is added. By default, the comfort noise gain control is enabled. 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 enables the user to choose the weighting of the comfort noise spectrum. Choices are: Brown, Pink, White, Blue, and Purple.

3. Click **OK**.

6.7 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.

6.8 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. This opens the **Send Automatic Gain Control Settings** window.

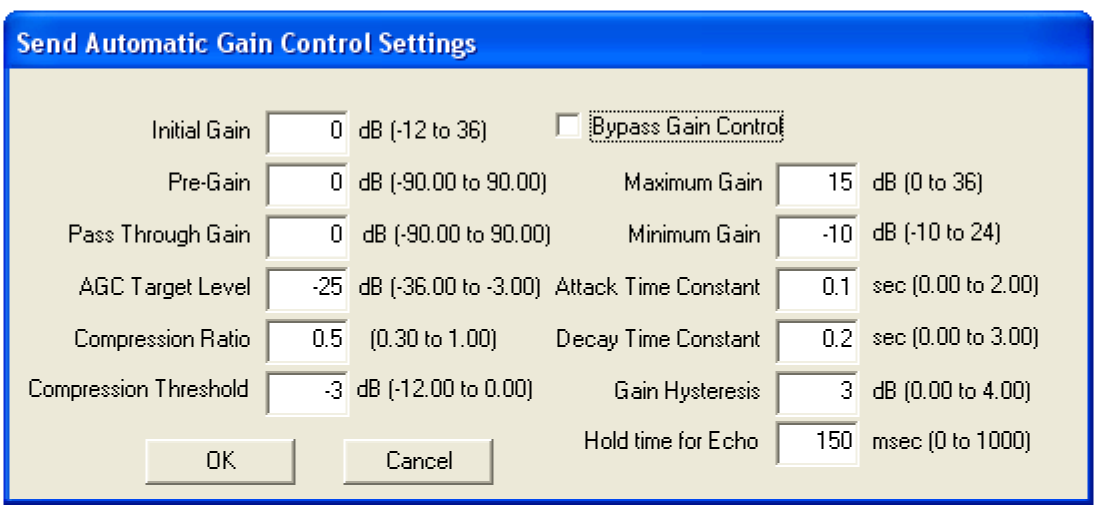


Figure 6-8 : Send Automatic Gain Control Settings window

2. Select the required settings.

Table 6-7 Send Automatic Gain Control Settings options

Option	Description
Bypass Gain Control	Leave this option unchecked to use the AGC for signal level control.
Initial Gain	Gain of the AGC at the initialization of cVc
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 primarily used 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 send 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, as the Compression Ratio values decreases compression increases.
Compression Threshold	The point at which compression begins (peak from full scale).
Maximum Gain	Sets the high threshold level for the gain. i.e. the AGC acts to maintain this value as the maximum gain level. No output has more than this gain value applied.
Minimum Gain	Sets the low threshold level for the gain. The AGC acts to maintain this value as the minimum 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 needs to decrease, set the **Attack Time Constant** lower, which speeds the rate of change.
Decay Time Constant	Sets the rate of 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 are slower to react.
Gain Hysteresis	Sets the upper and lower boundaries for the gain to change. For example a value of 4 means the AGC 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 parameters sets the amount of time to 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 2-mic Handsfree application is compatible with only one version of cVc.

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

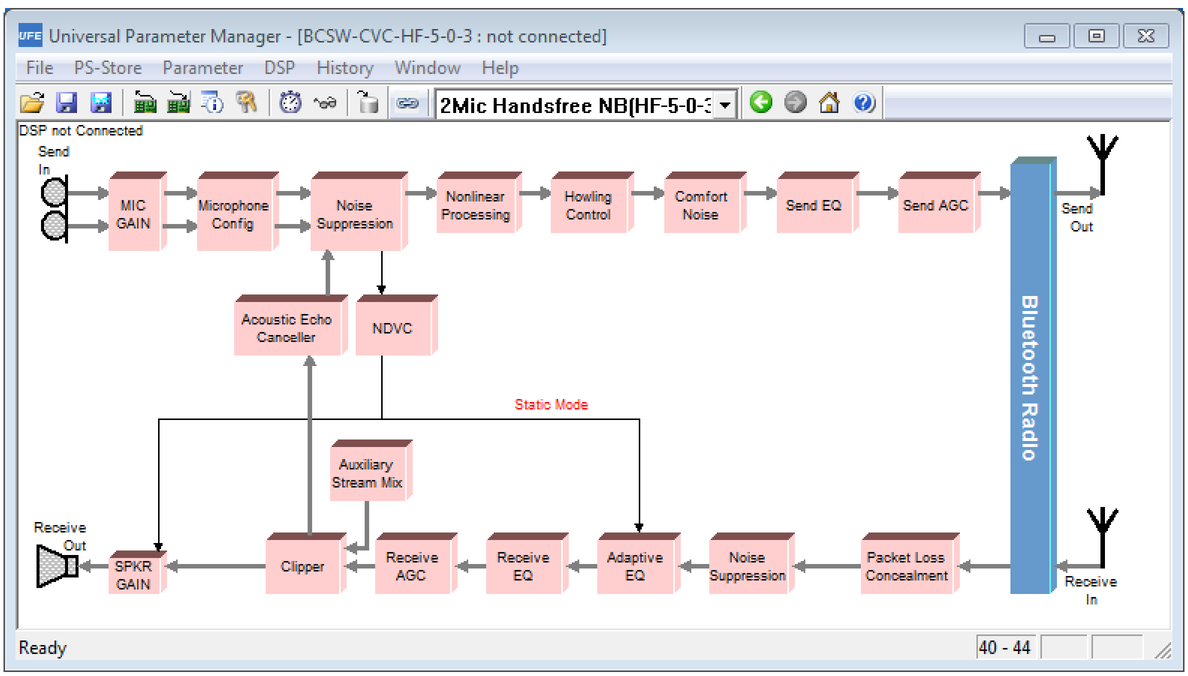
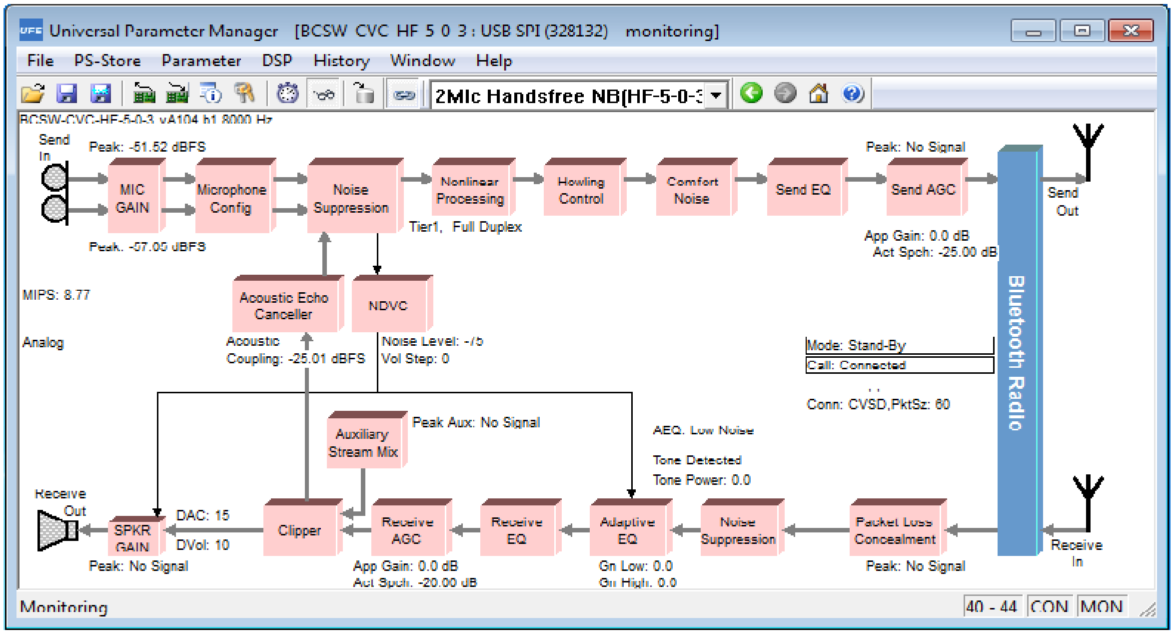


Figure 8-1 Parameter Manager window in static mode

The title bar shows **[BCSW-CVC-HF-5-0-3 : > not connected]** because a phone call has not been activated in 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-HF-5-0-3 : SPI->LPT1]**, and the top left corner of the window shows the product code **[BCSW-CVC-HF-5-0-3]**, build number **[B1]** and the sample rate of **[8000 Hz]**.



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** appears in the **Parameter Manager** window.

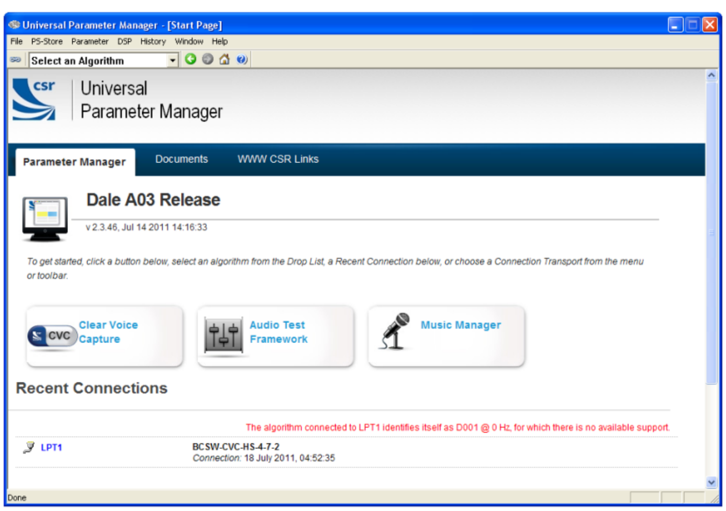


Figure 8-2 Parameter Manager DSP Not Responding message

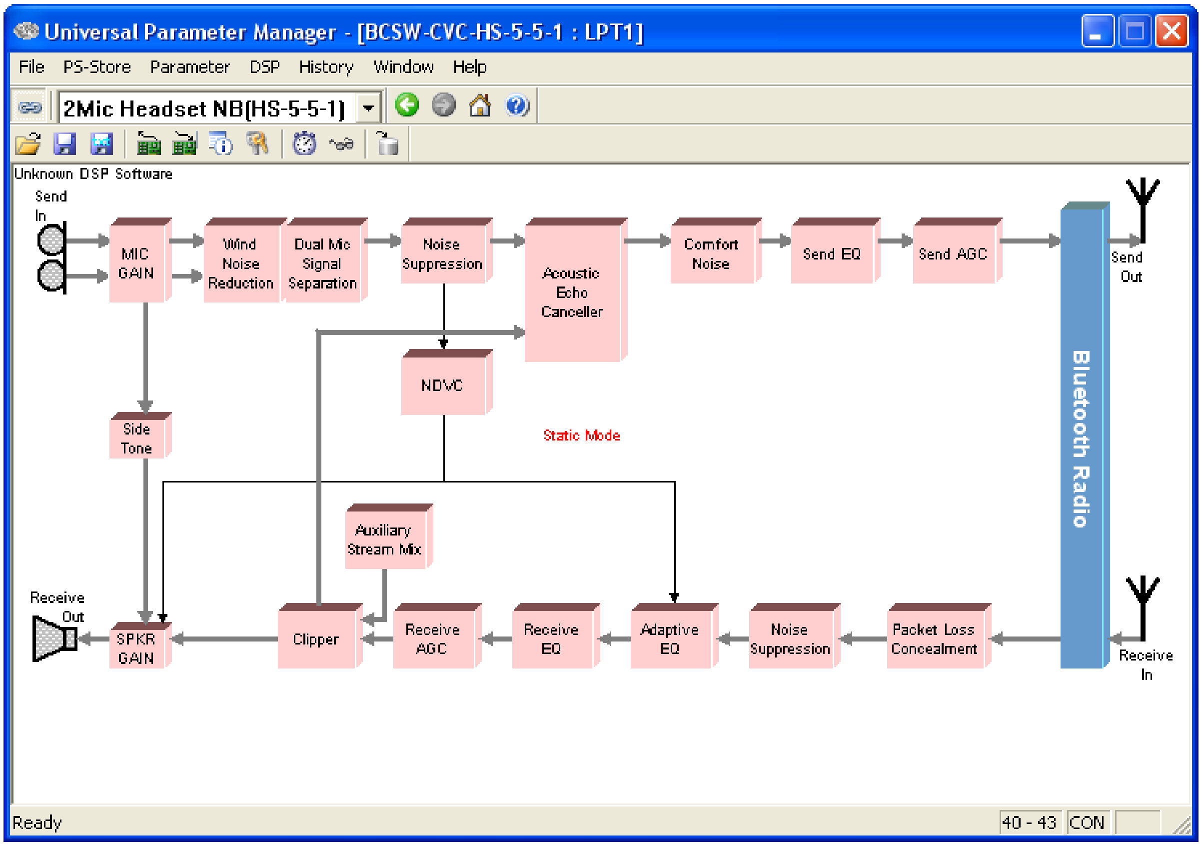


Figure 8-3 Parameter Manager window showing invalid version message

The **Recent Parameter Manager Connections** section provides details that attempt to identify the software version found and actions to remedy the problem.

If the DSP is not running (no active call) the following text is displayed: **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
BCSW-CVC-HF-5-0-3 2M-HF Tuning Guide	80-CT412-1/CS-00309822-UG
Enabling cVc for Headset SDKs	80-CT409-1 /CS-00122720-AN

Terms and definitions

3GPP	3rd Generation Partnership Project
ADC	Analogue to Digital Converter
ADK	Audio or Application Development Kit
AEC	Acoustic Echo Cancellation
AED	Acoustic Events Detection
AEQ	Adaptive Equalizer
AGC	Automatic Gain Control
AT	Attention (modem command prefix)
B&K	Bruel & Kjaer
BCSW	BlueCore Software
BF	Beam Former
BlueCore®	Group term for QTIL's range of Bluetooth wireless technology chips.
Bluetooth®	Set of technologies providing audio and data transfer over short-range radio connections.
BSS	Blind Source Separation
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
e.g.	*exempli gratia*, for example
EQ	Equalizer
ERLE	Echo Return Loss Enhancement
etc	*et cetera*, and so on
GSM	Global System of Mobile Communications
GUI	Graphical User Interface
HATS	Head and Torso Simulator

HF	Handsfree
HFK	Headset Mode
HS	Headset
HSP	Headset Profile
HTML	HyperText Markup Language
IC	Integrated Circuit
IEC	International Electrotechnical Commission
ITU	International Telecommunication Union
ITU-T	International Telecommunication Union-Telecommunication
MIC	Microphone
MIPS	Million Instructions Per Second
mSBC	Modified Sub Band Coding
NB	Narrow Band
NDVC	Noise Dependent Volume Control
NS	Noise Suppression
PC	Personal Computer
PCM	Pulse Code Modulation
RCV	Receive
PEQ	Parametric Equalization
PLC	Packet Loss Concealment
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
SSR	Simple Speech Recognition
ST	Side Tone
STMR	Side Tone Masking Rating
THD+N	Total Harmonic Distortion + Noise
UFE	Universal Front End
VM	Virtual Machine
WB	Wide Band
WNR	Wind Noise Reduction