Qualcomm Technologies International, Ltd.

BCSW-CVC-HS-4-9-2 1M-HS Parameter Manager

User Guide

80-CT420-1 Rev. AJ

November 7, 2017

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

Revision history

Revision	Date	Description
1	December 2013	Original publication of this document. Alternate document number CS-00309816-UG.
2	December 2013	Minor editorial corrections
3	March 2014	Edit to the content into the current template
4	May 2014	Editorial updates
5	July 2014	Updated for ADK 3.5
6	September 2016	Updated for ADK 4.1
7	April 2017	Updated for ADK 4.2
AH	November 2017	Updated to reflect new document and revision numbering scheme
AJ	November 2017	Editorial updates

Contents

Tables

Figures

1 Parameter Manager overview

The Parameter Manager is part of the UFE, which is a Windows-based application that enables the 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 chip.

QTIL provides Clear Voice Capture (CVC) algorithms used to create voice products. The One Microphone (1-mic) Headset Parameter Manager application enables you to customize the performance of your 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 Qualcomm® BlueCore™ digital signal processor (DSP).

The CVC Headset application provides these major modules accessible using the Parameter Manager application:

- Noise Suppression (NS) that includes Wind Noise Reduction (WNR)
- Acoustic Echo Cancellation (AEC)
- Automatic Gain Controls (AGC)
- Equalizers (EQ)
- Stream Mixer
- Clipper
- Near End Audio Enhancement includes Noise Dependent 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 *1-mic Headset Tuning Guide* for information on the tuning process.

1.1 Software versions supported

This document covers the audio controls of CVC BCSW-CVC-HS-4-9-2 algorithm. The audio controls and adjustments are used on the IC listed in Table 1.1.

Table 1-1 Part number matrix

IC Supported	CVC Product Code	Version SysID	NB (8k)	WB (16k)	CVC License Key Part Number
CSR8670 (Flash)	BCSW-CVC-HS-4-9-2	B012	Yes	Yes	BCSW-CVC-HS-1M-Fx
CSR8675 (Flash)	BCSW-CVC-HS-4-9-2	B012	Yes	Yes	BCSW-CVC-HS-1M-Fx

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

Download the CSR86xx UFE installer from www.csrsupport.com.

1.2 8th Generation new features

QTIL is constantly developing its CVC 1-mic solution and making improvements driven by the market place. This section lists improvements made since the previous release (BCSW-CVC-HS-4-8-1) that improve performance and/or affect the tuning process.

New/improved features include:

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

1.3 Assumptions

The Parameter Manager application is designed for developers of Bluetooth voice-enabled products.

This document assumes:

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

2 Using the Parameter Manager application

The basic steps for using the Parameter Manager application are:

- Install the QTIL Audio Development Kit or the UFE Installer from www.csrsupport.com.
- Run the UFE application
- Use the **Quick Start** link in the UFE Documents section page that displays when UFE application is accessed to familiarize yourself with the Parameter manager application
- Access the Headset Parameter Manager application
- Connect the Parameter Manager application using an active SPI
- Enter the security key, if required
- Pair and connect the device (usually a Bluetooth headset) with a Bluetooth source device (usually a Bluetooth phone) as the audio gateway or connect USB to PC if USB wired mode is to be used
- Use parameters and metrics information for tuning and/or monitoring

2.1 Installing the standalone universal front end application

QTIL has designated a default location for the installation of the Parameter Manager. Ensure that location is available. If it is not available, it is created for you.

2.1.1 Installing the QTIL installer

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

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>

2.1.3 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

2.2 Accessing the universal front end application

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

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

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

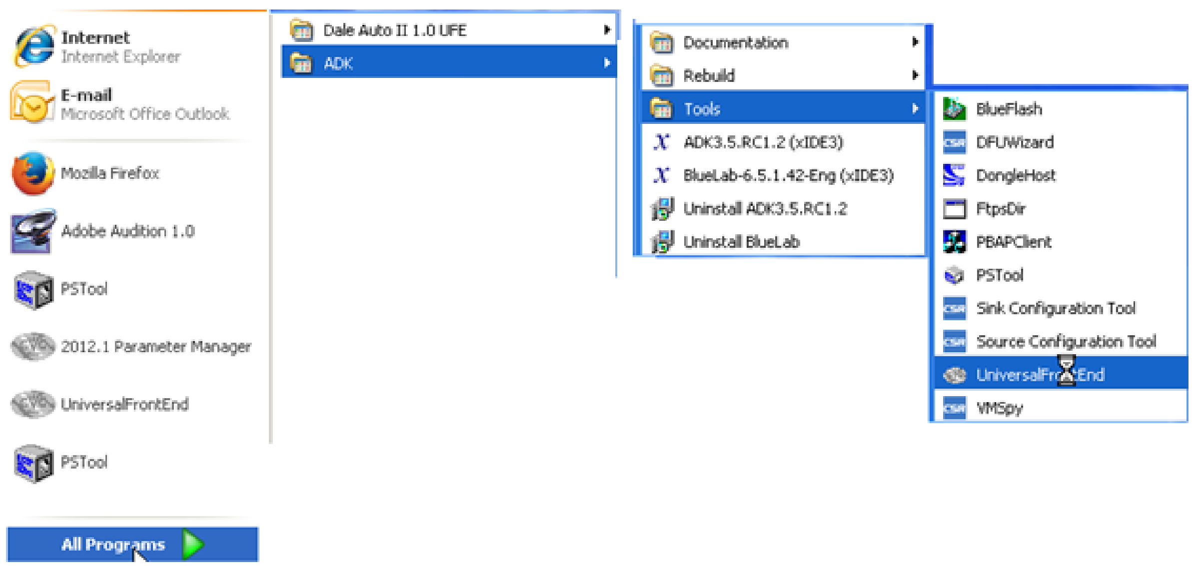


Figure 2-1 Accessing UFE from Windows Start menu

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

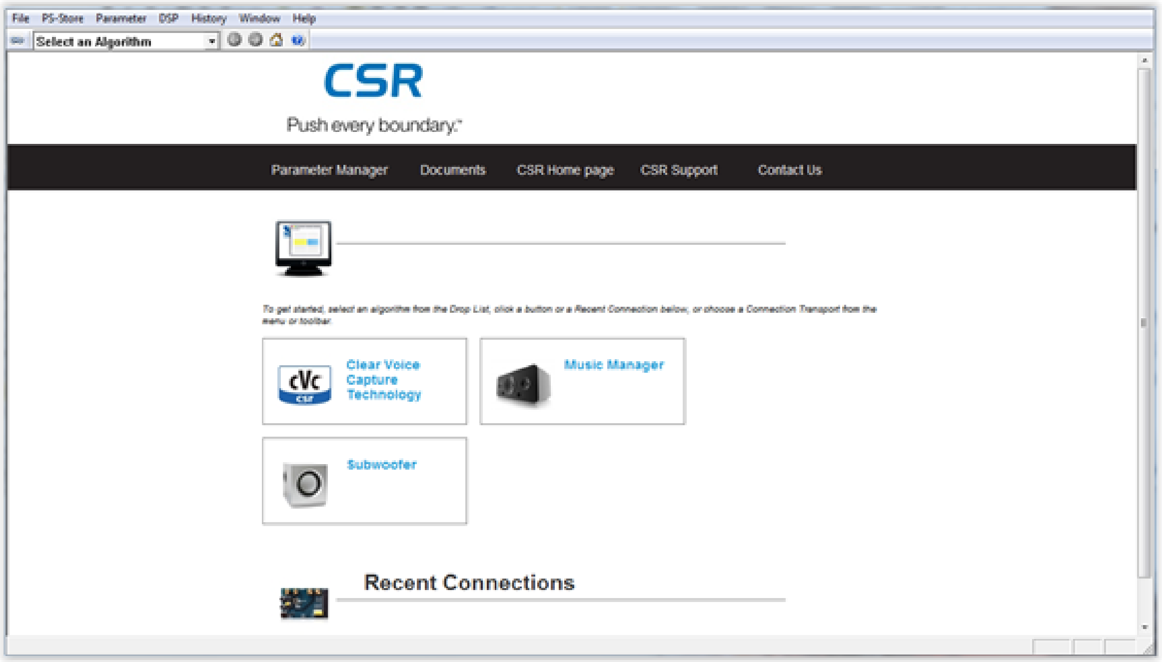


Figure 2-2 UFE application opening HTML page showing Quick Start link

2.3 Viewing the UFE Quick Start

Click the **Documents** link on the opening HTML page. This displays the Quick Start Link.

Click the Home icon to return to the opening HTML page.

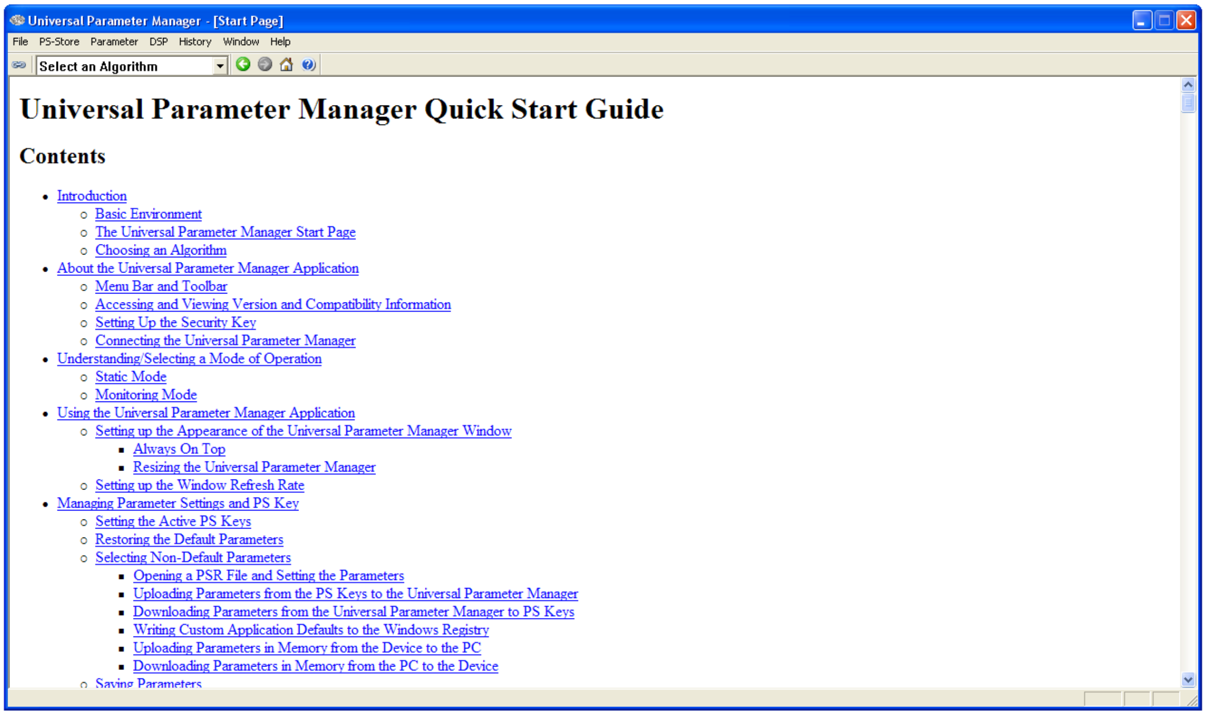


Figure 2-3 UFE Quick Start embedded documentation accessing the Headset Parameter Manager

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

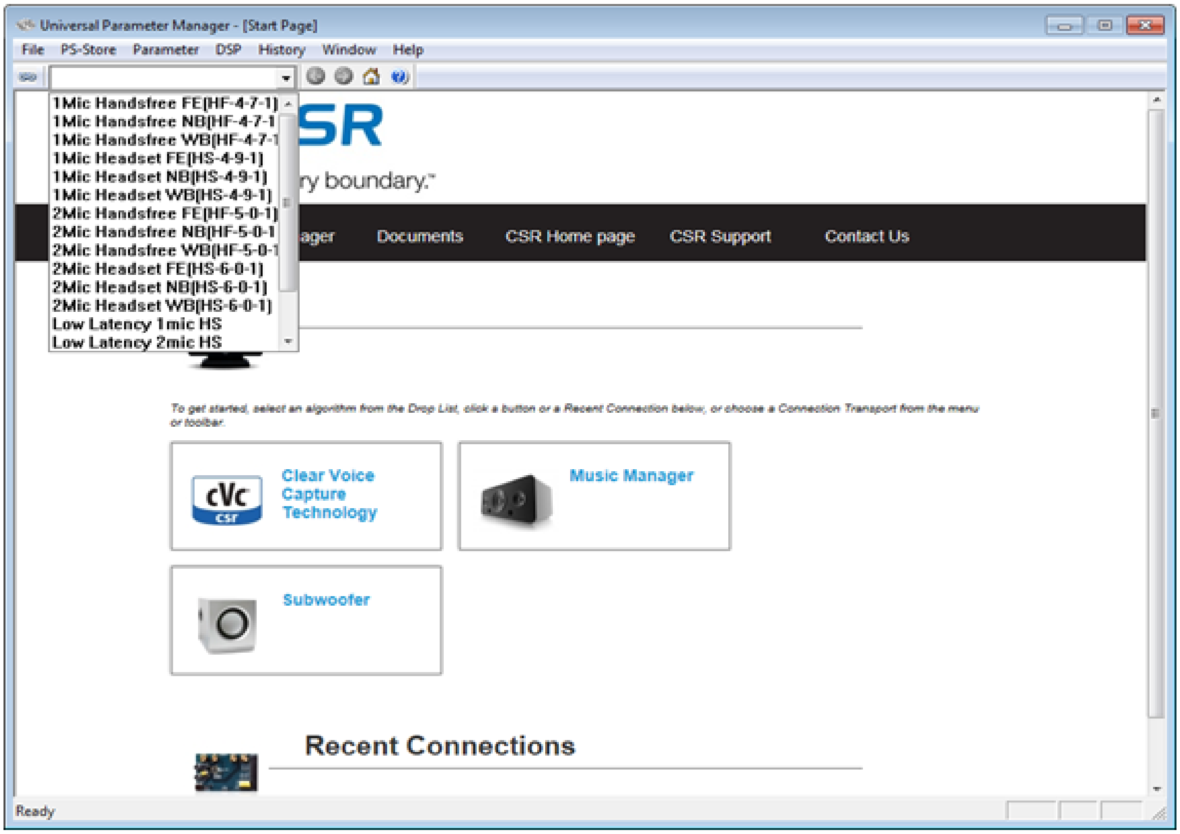


Figure 2-4 UFE application showing Select an Algorithm dropdown list (ADK)

The dropdown list contains the following options for the CSR86xx:

- 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 44.1 kHz
- Music Manager 48 kHz

To select the 1-mic Headset Parameter Manager application, click **1Mic-Headset NB(HS-4-9-2)** algorithm from the dropdown list in Figure 2-4.

2.4 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 setup the security key (if required) and tune the algorithms.

2.5 Security key setup

A security key protects the CVC library. When the ADK application is used, the BCSW-CVC-HS-4-9-2 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.

The CSR86xx ROM-based chips do not require a security key.

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 Headset ADKs* describes how to activate the cVc algorithms for development purposes. When the Mono Headset ADK is installed, navigate from the windows **Start** menu to:

<ADK Name>\Documentation\Mono Headset ADK Support Documentation

This displays the Support Document Index page from which the documents accompanying the ADK can be opened. Open the *Enabling cVc for Headset ADKs* document, and see section 3.

3 Headset Parameter Manager application

The BCSW-CVC-HS-4-9-2 B010 represents the 1-mic Headset Parameter Manager.

This section describes 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	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 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	The Packet Loss Concealment 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 Noise Suppression (Including Wind Noise Reduction). This block removes unwanted noise during hands-free conversation, cleaning the audio for the near end listener. See Figure 5-2.
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 four 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-linearity's that would occur after the echo canceller. The amount of clipping can be controlled by the developer. 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 interruption of 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 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 adequate speaker sensitivity being designed into the overall system. See NDVC.
Side Tone	This block is provided in the digital domain to enable side tone capability in the near-end speaker. The side tone is automatically levelled by cVc to deliver a consistent loudness, independent of the Bluetooth volume. The developer may enable or disable this function. The amount of injected side tone is programmable. See Side Tone.

Table 3-3 Send Path parameters and metrics

Option	Description
Send Path Processing Parameters	
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 (includes Half-duplex control).
Noise Suppression **(includes Wind Noise Reduction)**	This block educes noise with temporal characteristics uncorrelated with speech. The Noise Suppression function is most effective in reducing noise with constant statistics. This algorithm is not intended to cancel instantaneous noise. The amount of noise suppression can be controlled to achieve optimum suppression versus voice distortion levels for the intended application. This block also contains a wind noise reduction feature (send path only) that cleans the speech when wind is detected. This block can remove unwanted noise during a hands-free conversation, cleaning the audio for the far end listener. See Noise Suppression (Including Wind Noise Reduction).

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

Option	Description
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	This block 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 Several statistics are displayed and updated during an active call, such as Peak and Noise Level.

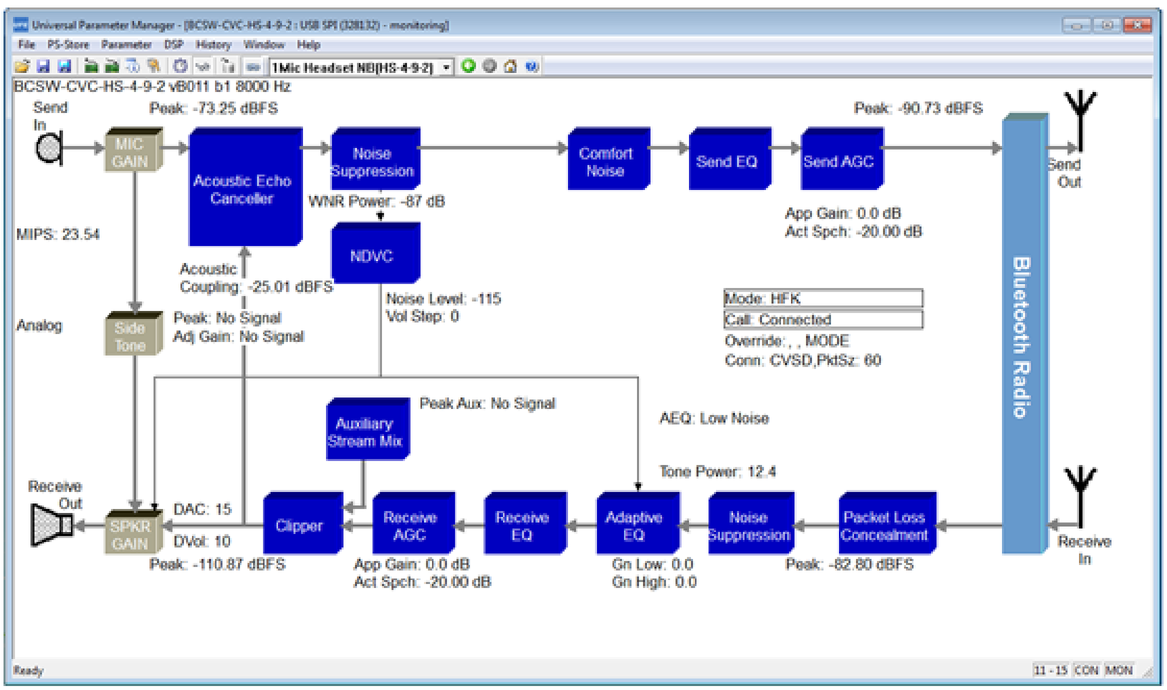


Figure 3-1 Parameter Manager window showing all active modules

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, left-click the **Mode:** field and select the required override mode.

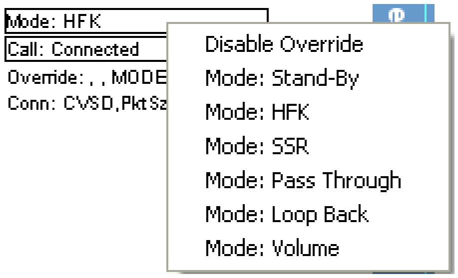
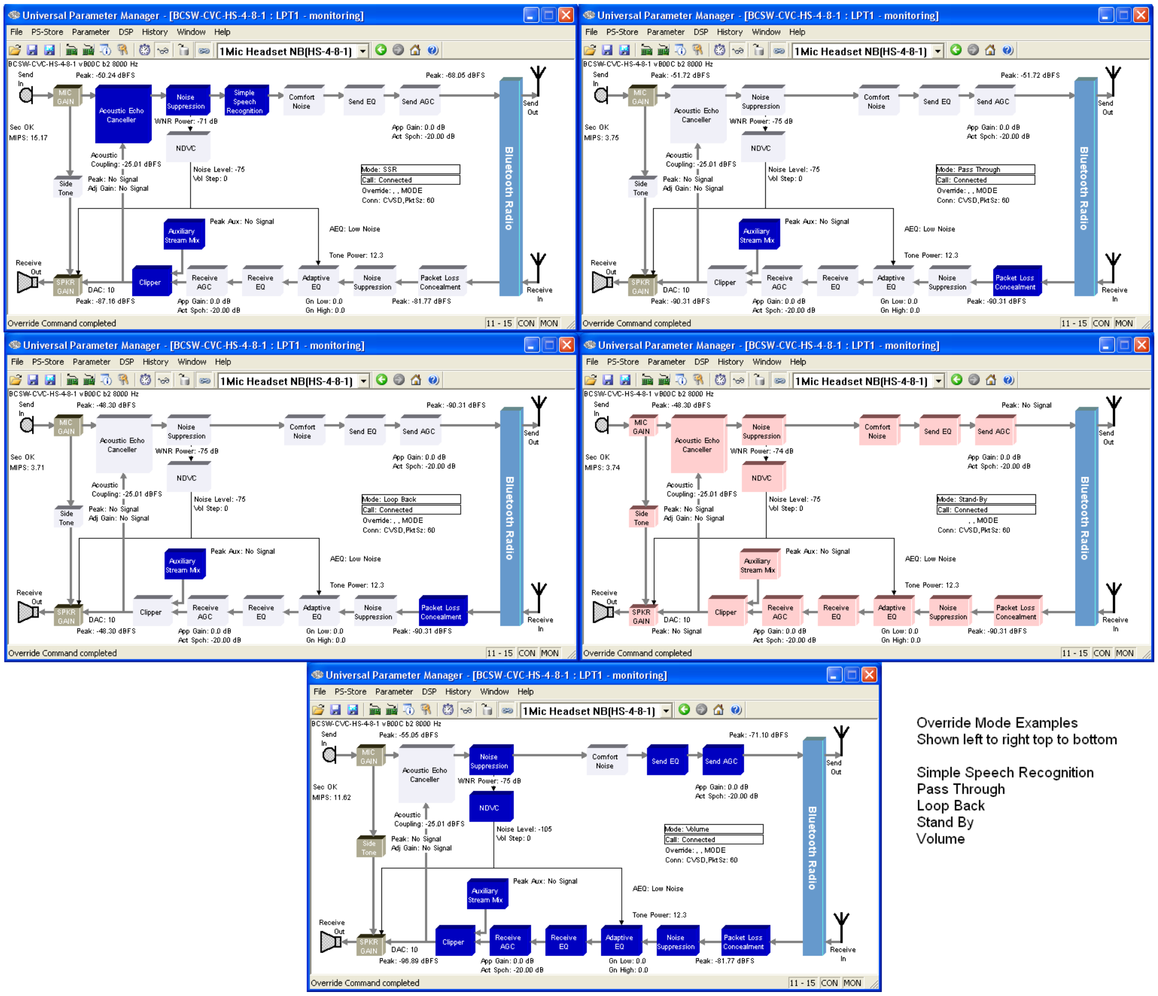


Figure 3-2 Selecting an override mode

Overrides modes are:

- Stand-by: Processing in standby, no audio flow
- HFK: Full processing
- SSR: Simple Speech Recognition
- Pass Through: Microphone pass-through

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



Override Mode Examples
Shown left to right top to bottom

Simple Speech Recognition
Pass Through
Loop Back
Stand By
Volume

Figure 3-3 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 displays. This window controls the ADC gain.

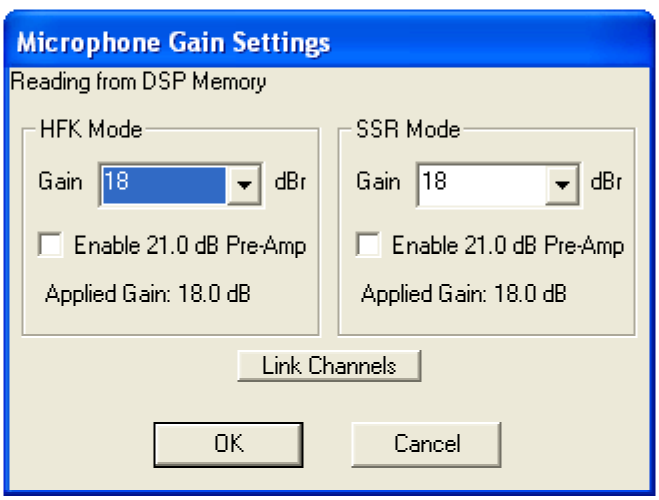


Figure 4-1 Microphone Gain Settings Window

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.

2. 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 these check boxes are unchecked (default), 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.

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

4.2 SPKR gain

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

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 displays.

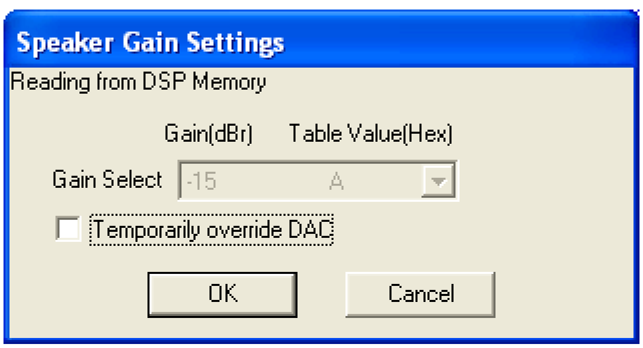


Figure 4-2 Speaker Gain Settings window

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 a 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: a. Check the Temporarily override DAC option box b. Type the appropriate value in the Gain Select field. 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 to be made. The value in the Gain Select field can be overridden by a volume change.

3. After selected the required settings, click **OK**.

5 Using the Receive Path audio tuning controls

Parameter Manager provides access to these receive path audio tuning blocks. Figure 5-1 shows these blocks.

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

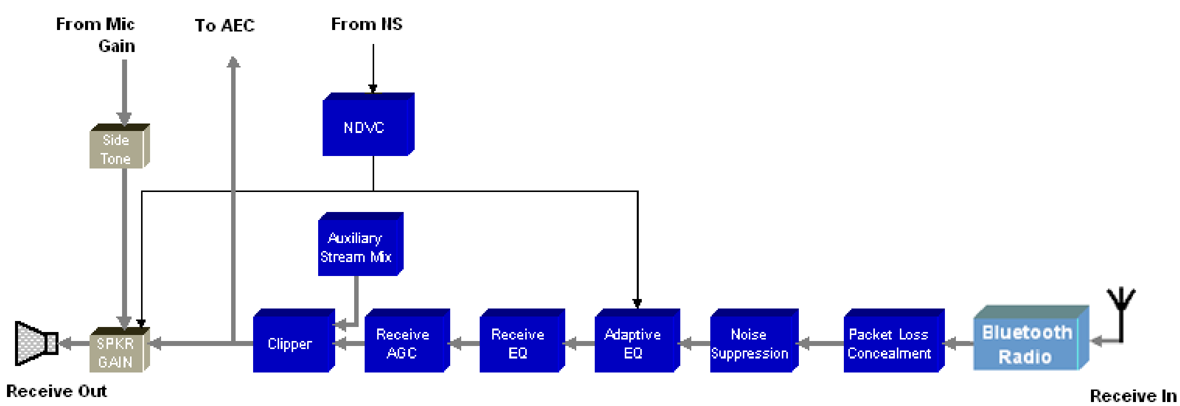


Figure 5-1 Receive Audio Path processing blocks

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 a variety of techniques such as pitch based waveform substitution.

To adjust the Packet Loss Concealment (PLC) settings, no tuning is provided. Check the **Bypass Packet Loss Concealment** to disable module if required.

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

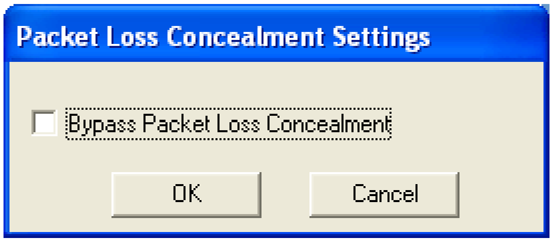


Figure 5-2 Packet Loss Concealment Settings window

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 displays.

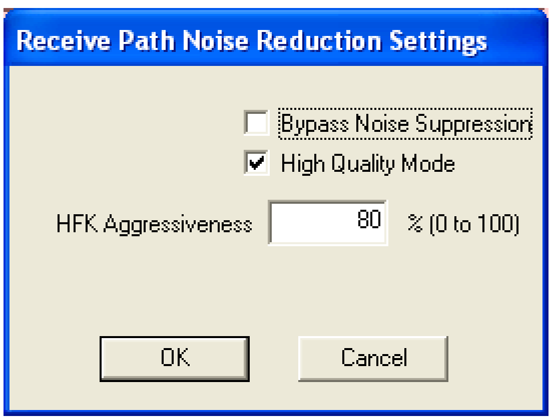


Figure 5-3 Noise Suppression Settings window

2. Select the required settings.

Table 5-1 Noise Suppression Settings 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. The receive signal has been processed by the cellular network, and transmitted over Bluetooth to avoid over processing the voice. QTIL recommends that you set the HFK 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 most 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 diagram, the AEQ speech signal is divided into three different frequency regions. They are defined as:

- 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 in 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 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 Equalization Settings** window displays.

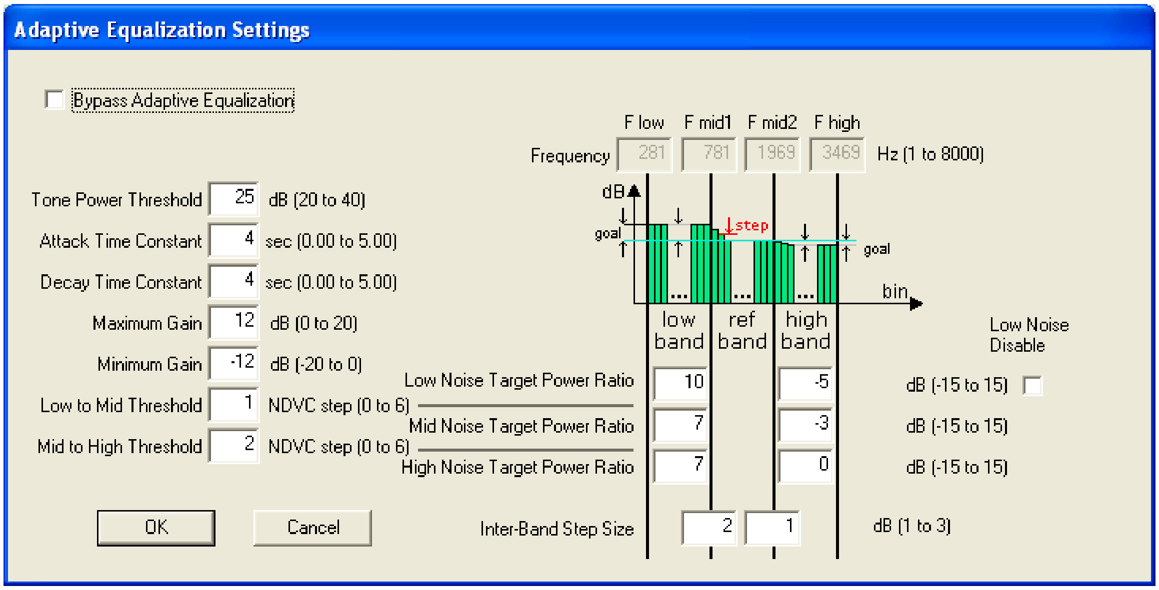


Figure 5-4 Adaptive Equalization Settings window (for wide band)

2. Select the required settings.

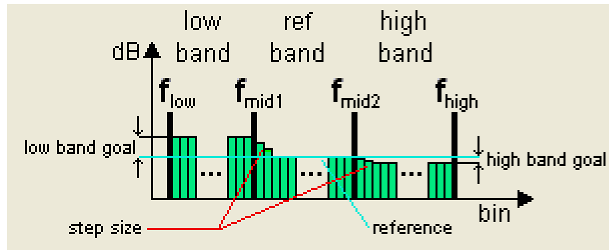
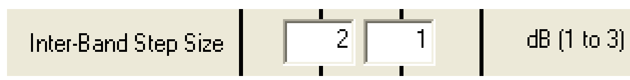
Table 5-2 Adaptive Equalization options

Option	Description
Tone Power Threshold	If a tone displays in the receive audio path, the AEQ or the NS should not adapt during the tone. This causes a distortion 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 displays. Setting the threshold to low could result in some speech to be falsely detected. The AEQ and NS is operating and speech distortion could result. Setting the threshold to high could 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, which increases 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, which increases 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	Use this field to 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. This field is only effective if the NDVC is enabled and the DAC has available headroom.

Table 5-2 Adaptive Equalization 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. Low Noise Target Power Ratio 10 -5 dB (-15 to 15) 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. Mid Noise Target Power Ratio 7 -3 dB (-15 to 15) 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. High Noise Target Power Ratio 7 0 dB (-15 to 15) 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 options (cont.)

Option	Description
Inter-Band Step Size	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. 
Low Noise Disable	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.

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 *CSR86xx 1-Mic Headset Tuning 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 adjust the Adaptive EQ with High Frequency Emphasis Settings:

1. Select **1Mic Headset NB[HS-4-9-2** 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 displays.

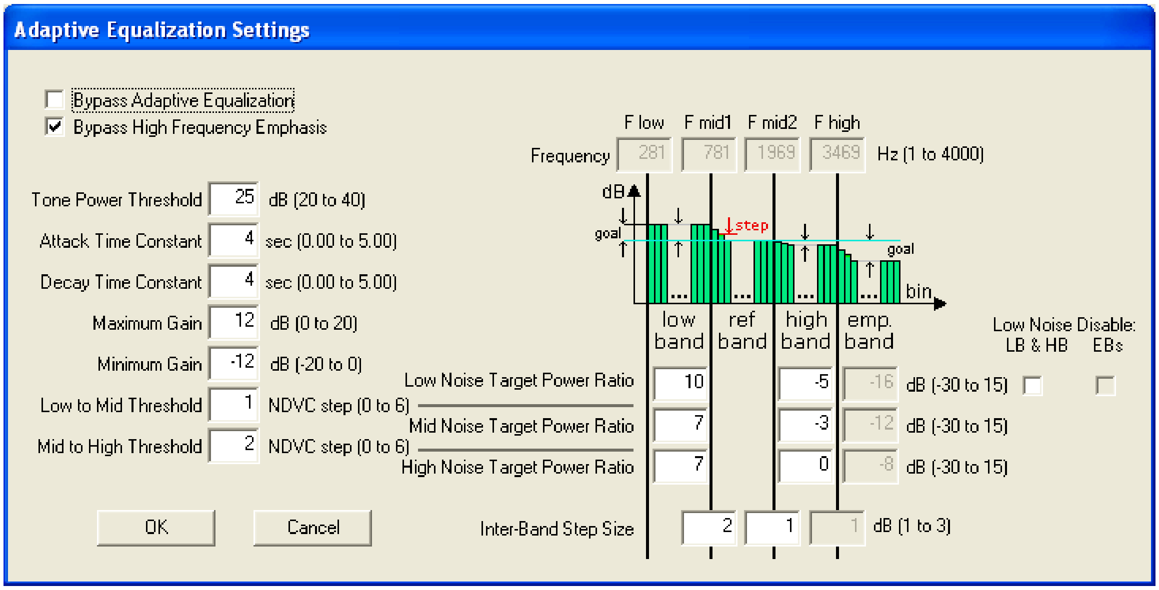


Figure 5-5 Adaptive EQ with High Frequency Emphasis Settings window

3. Select the required settings..

Table 5-3 Adaptive EQ with High Frequency Emphasis 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 **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 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.

1. Choose **1Mic Headset 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 displays.

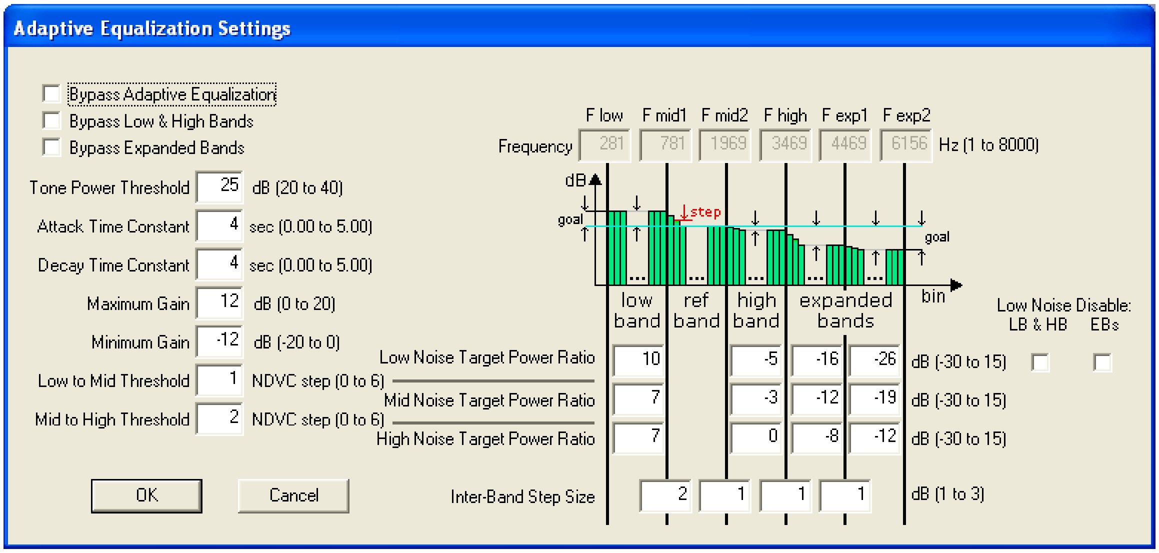


Figure 5-6 Adaptive EQ with Frequency Expansion Settings window

3. Select the required settings.

Table 5-4 Adaptive EQ 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 **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 displays.

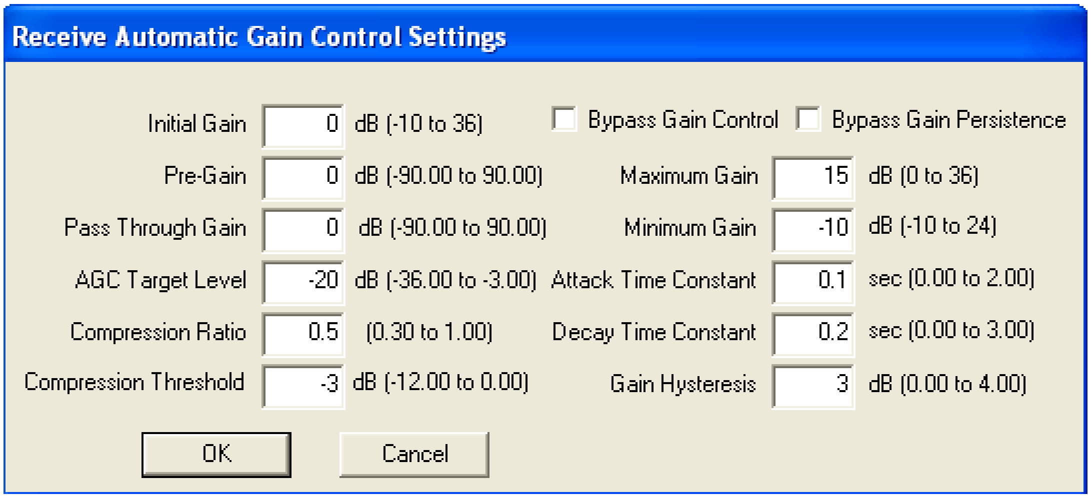


Figure 5-7 Receive Automatic Gain Control Settings windows

2. Select the required settings.

Table 5-5 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 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 is primarily to compensate for the loss of the AGC block. It is typically used for demonstration when toggling between HFK and PT modes, or 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.
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 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. For example, 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 the rate of change.

Table 5-5 Receive AGC 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 is 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.

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 displays.

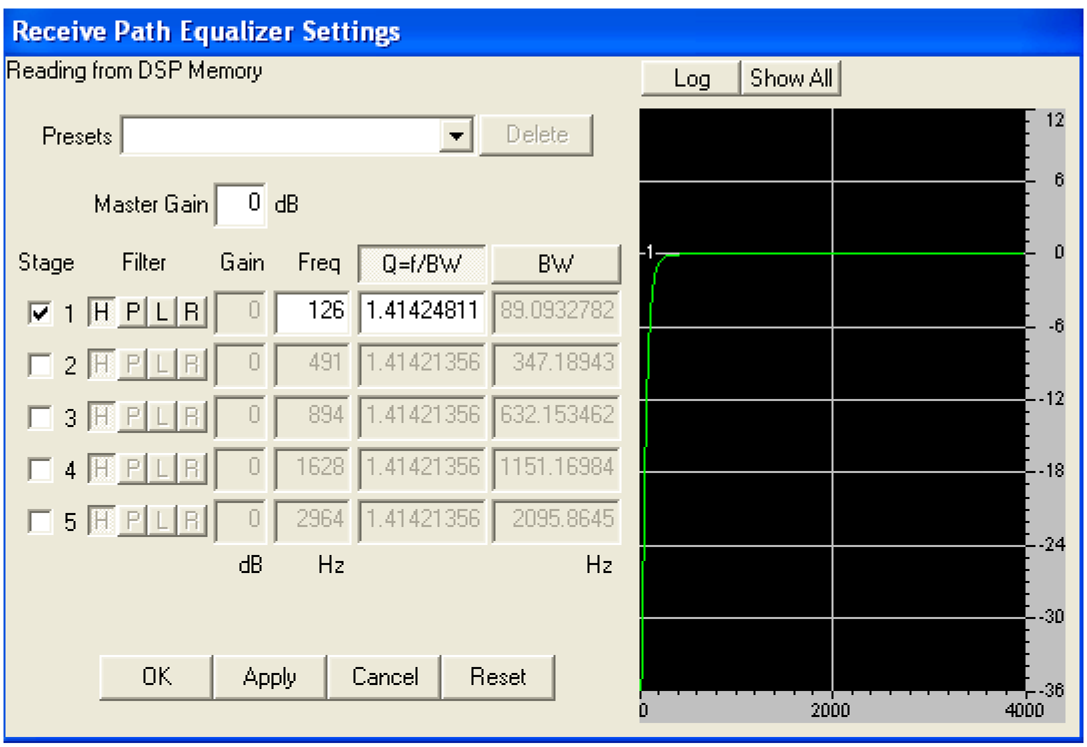


Figure 5-8 Receive Equalizer Settings window

2. The current state of the screen is the default state before presets are selected (the **Presets** field is blank by default). The Red/Green line 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.

3. Select the required settings.

Table 5-6 Equalization options

Option	Description
Presets	This dropdown list provides a collection of sample equalizer 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	To create an equalizer filter type a preset name. The Add button highlights it. 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 the 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. 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. 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 options (cont.)

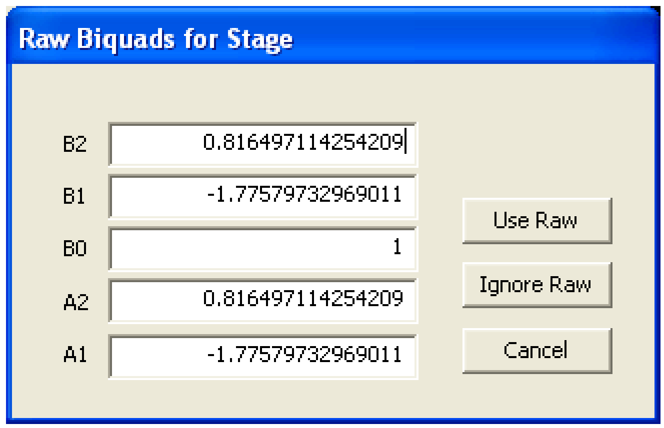
Option	Description
Filter	The Filter option enables the Stage Filter to be set to H, P, L or R (High Pass, Parametric, Low Pass and Raw). A filter type can be set for each stage enables creation of complex curves: ■ High Pass ■ Parametric ■ Low Pass ■ Raw  If Use Raw is selected, enter the stage coefficients directly using the GUI provided. In 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 stat, which is 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).

Table 5-6 Equalization options (cont.)

Option	Description
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	
Zoom In	On the grey Scale bar in the plot area, 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 to perform optimal echo cancellation.

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 displays.

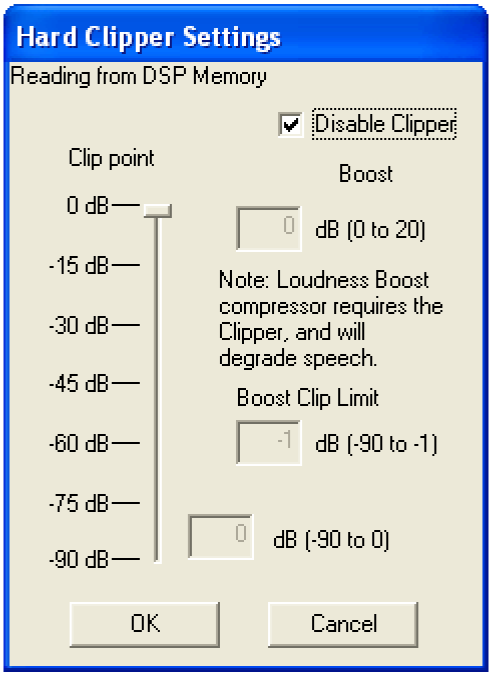


Figure 5-9 Hard Clipper Settings window

The available Clipper variables are:

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

2. Select the required settings.

Table 5-7 Clipper 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 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

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.

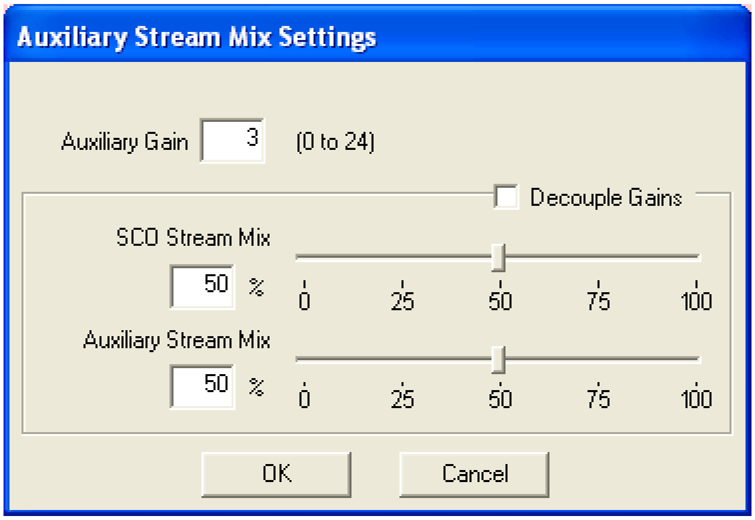


Figure 5-10 Auxiliary Stream Mix Settings window

2. Select the required settings.

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. 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 Dependent Volume Control Settings** window displays.

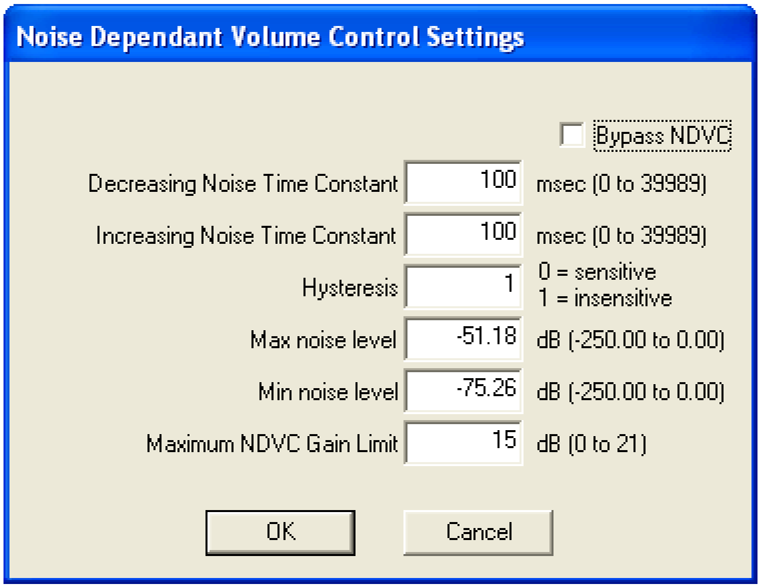


Figure 5-11 Noise Dependent Volume Control Settings window

2. Select the required settings based on the options in Table 5.9. 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, which means 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 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. No gain is added.
Maximum NDVC Gain Limit	This setting 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**.

5.9 Side Tone

The Side Tone option determines the gain applied to the side tone 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 levelling mechanism (**Gain Adjustment Limit**) adds proportional side tone as the Bluetooth volume drops to maintain a "levelled" side tone. 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 displays.

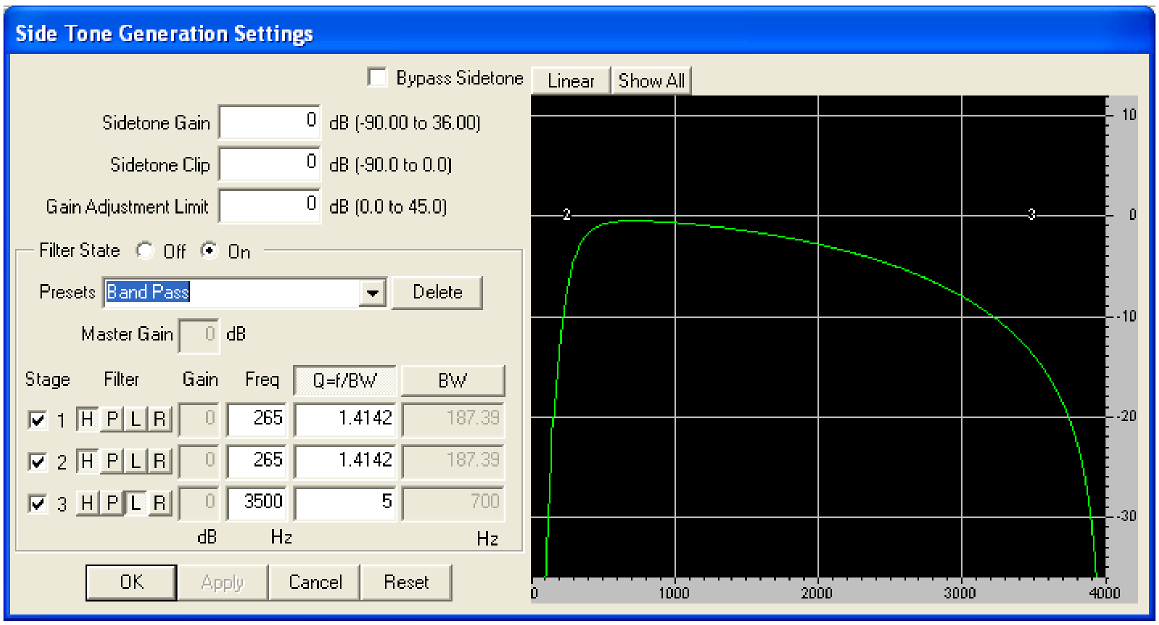


Figure 5-12 Side Tone Generation Settings window

2. Select the required settings.

Table 5-10 Side Tone Generation, gain option

Option	Description
	If checked, bypasses the side tone feature reducing processor cycles.
Sidetone Gain	Selects the required Sidetone Gain. The higher the gain the more side tone is applied. The Default is 0 dB and the allowable range is from -90 to 36 dB.
Sidetone Clip	If you require a large amount of side tone gain and the DAC could saturate, adjust Sidetone Clip to avoid saturation. Bypass Sidetone The Default is 0 dB (no clip or DAC full scale). As the clip value becomes more negative, more clipping occurs. The allowable range is from -90 to 0 dB. BypassThe Sidetone Clip invokes a hard clipper that causes some audio distortion.

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

Option	Description
Gain Adjustment Limit	Adjust the Gain Adjustment Limit parameter to adjust the amount of side tone gain based on Bluetooth volume. The Gain Adjustment Limit parameter controls the amount of levelling. The allowable range is from 0 (no levelling) to 45 dB. The Gain Adjustment Limit levelling mechanism adds proportional side-tone as the Bluetooth volume drops to maintain a "levelled" side tone.
Filter State	The Filter State can be set Off or On. If On, then a programmable three stage filter is applied to the side tone audio. If set Off, than the raw side-tone audio is passed.
Presets **Master Gain** **Stage** **Filter** **Gain** **Freq.** **Q=f/BW** **BW**	If the Filter State is On, then a programmable filter is applied to the side-tone audio. The operation of the filter is the same as described in Receive EQ, except only 3 stages are available. See Figure 5-8 for an example filter and the GUI description and Figure 5-12 for a typical side-tone filter.

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

6 Using the Send Path tuning controls

Parameter Manager provides access to these send path tuning controls:

- Acoustic Echo Canceller
- Noise Suppression
- Comfort Noise
- Send EQ
- Send AGC

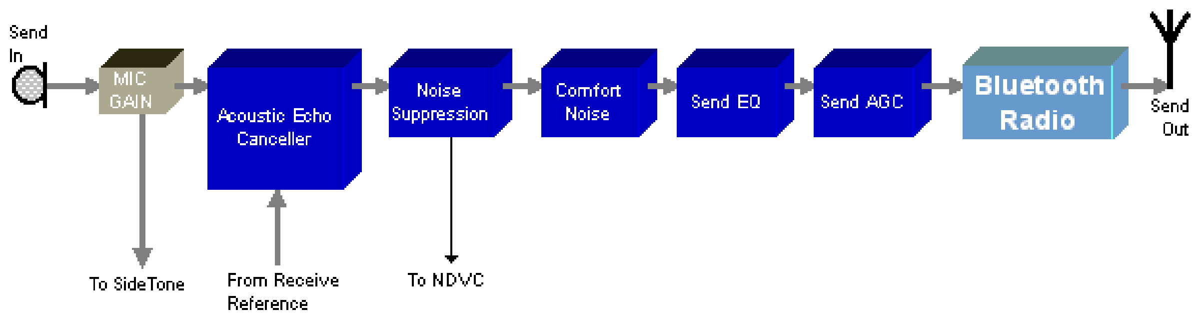


Figure 6-1 Send Path tuning controls

6.1 Acoustic Echo Canceller (includes Half-duplex control)

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 displays.

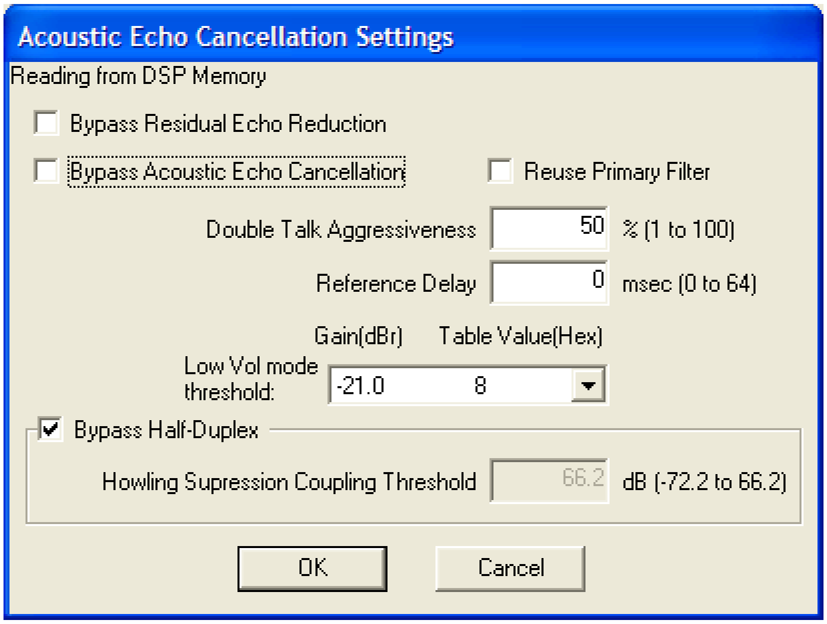


Figure 6-2 Accoustic Echo Cancellation Settings window

2. Select the required settings based on the options in Table 6.1.

Table 6-1 Acoustic Echo Cancellation options

Option	Description
Bypass Residual Echo Reduction	By default, this option is unchecked and the Residual Echo Reduction is enabled. This feature is best used 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.
Bypass Acoustic Echo Cancellation	By default, this option is unchecked and the AEC module is included in the system. To disable AEC processing check this option.
Reuse Primary Filter	Check this option when the echo coupling is high. Selecting this option increases the DSP processing load because the audio is re-cycled through the primary AEC filter to further reduce echo. Under certain conditions where processing power is not a concern, then this option is enabled to ensure 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. The closer this field is set to 100%, the less echo attenuation is applied.

Table 6-1 Acoustic Echo Cancellation options (cont.)

Option	Description
Reference Delay	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.
Low Vol mode threshold	This is a switch threshold used to disable the AEC and Comfort Noise when they are no longer required. This feature disables the AEC when the speakers acoustic loudness is decreased due to a decrease in the headset volume. The Low Vol mode threshold is set to the Bluetooth step where the echo is tolerable without the software AEC. When the Bluetooth volume step is at or above the threshold, the AEC and Clipper are enabled. This mechanism is a power saving technique.
Bypass Half-Duplex	By default, this option is checked and the AEC Half-Duplex module is bypassed from the system. To enable Half-Duplex processing uncheck this option. Half-Duplex is required if the device has excessive or distorted echo that cannot be removed by the AEC alone.
Howling Suppression Coupling threshold	The Howling Suppression Coupling Threshold sets the threshold to invoke full band attenuation in the presence of far end speech to eliminate echo. The threshold is based on the acoustic coupling statistic displayed during monitoring mode. This is a switch threshold used to insert send path attenuation when the far end speech gets coupled back into the microphone, and the audio threshold has been exceeded. The attenuation is removed when below the acoustic coupling threshold.

3. Click **OK**.

6.2 Noise Suppression (Including Wind Noise Reduction)

The Noise Suppression block defines both the aggressiveness of the noise suppression and wind noise reduction algorithm.

To set Noise Suppression options:

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

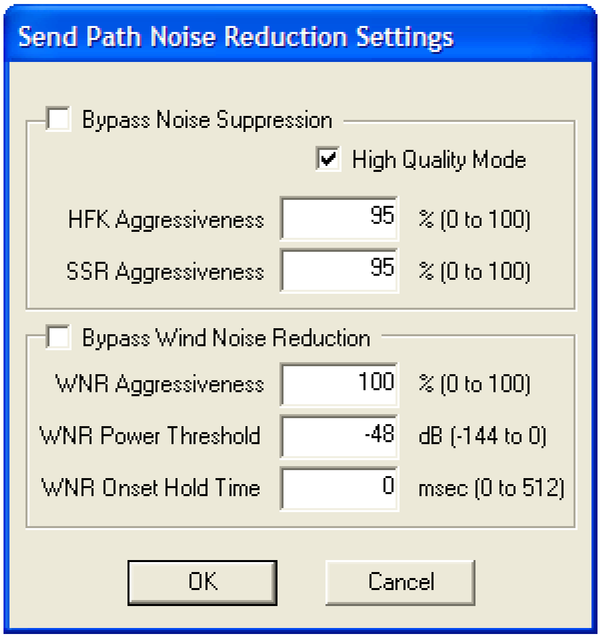


Figure 6-3 Noise Suppression Settings window

2. Select the required Noise Suppression settings.
3. If wind noise reduction is required, select the settings based on the WNR options.

Table 6-2 Noise Suppression settings options

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 block. If it is bypassed, system performance could 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% maximises noise suppression.
NS Aggressiveness	Controls the amount of noise suppression applied to the send signal during the noise suppression mode of operation. Setting this parameter to 100% maximises noise suppression.
Bypass Wind Noise Reduction	If checked, bypasses the WNR feature.

Table 6-2 Noise Suppression settings options (cont.)

WNR Aggressiveness	Controls the amount of wind noise reduction applied to the send signal during main processing HFK mode. Once wind has been detected, a good balance between voice quality and wind noise reduction is achieved by setting the WNR Aggressiveness from 90 to 100%. Setting this parameter to 100% maximises wind noise reduction.
WNR Power Threshold	This detection parameter indicates no wind or a low wind condition. No WNR is performed when the signal energy falls below this WNR Power Threshold. Set this parameter to `0` to never detect wind nor apply WNR.
WNR Onset Hold Time	The WNR Onset Hold Time parameter helps reduce the occurrences of false detects so voice quality is least affected. WNR is applied only when the detected wind duration is longer than the WNR Onset Hold Time. A short hold time makes the algorithm more sensitive to short, sporadic wind. A longer hold time makes the algorithm more conservative in detecting wind while preserving voice quality. Set this parameter to `0` to rapidly apply WNR when the power threshold is crossed.

4. Click **OK**.

6.3 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 displays.

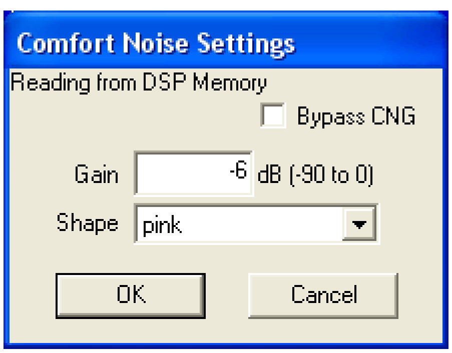


Figure 6-4 Comfort Noise Settings window

2. Select the required settings.

Table 6-3 Comfort Noise options

Option	Description
Bypass CNG	This check box enables or disables 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 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**.

6.4 Send EQ

The Send EQ block displays 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.

6.5 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 displays.

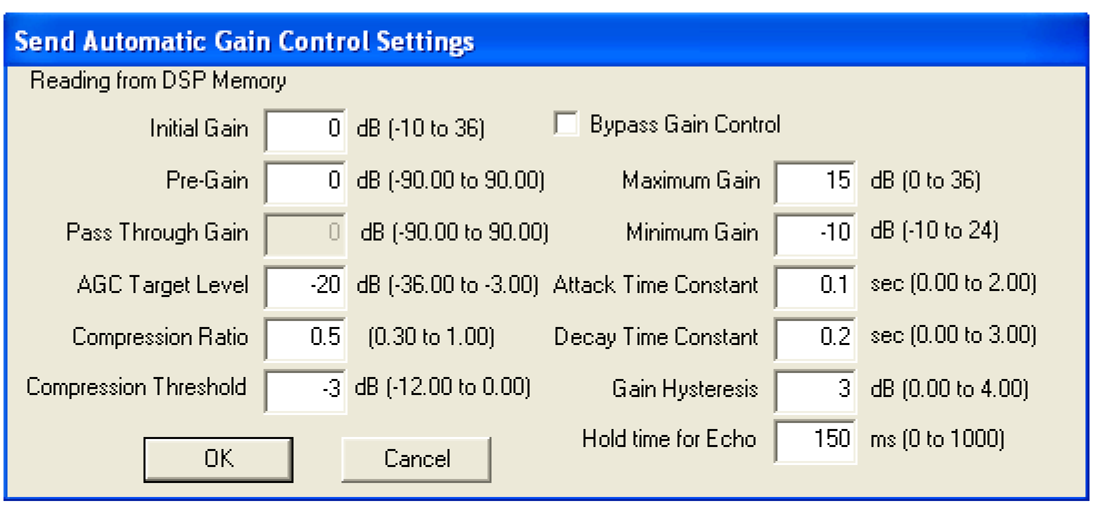


Figure 6-5 Send Automatic Gain Control Settings window

2. Select the required settings.

Table 6-4 Send Automatic Gain Control 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 receive output, below which no compression of the input signal occurs (usually set close to Full Scale level).
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.
Maximum Gain	Sets the high threshold level for the gain. For example, 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 For example, 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, setting 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 is 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	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. This closes the **Parameter Manager** window.

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.

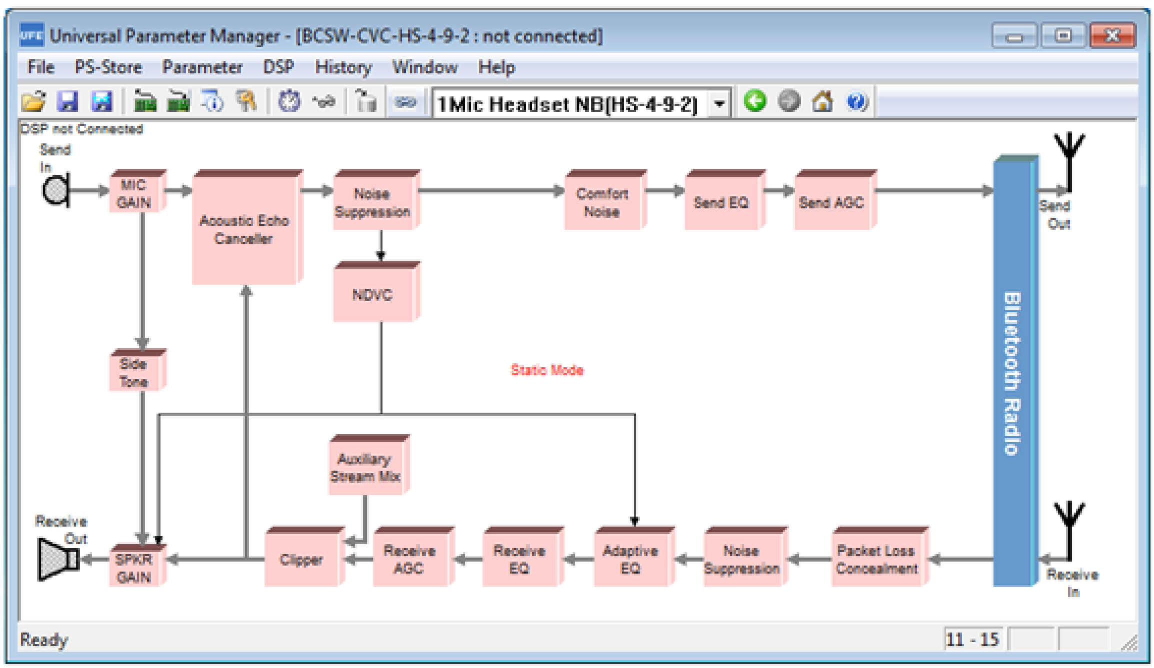


Figure 8-1 Parameter Manager window in Static mode

The title bar shows **[BCSW-CVC-HS-4-9-2 :** > **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-4-9-2 : SPI->LPT1]**, and the top left corner of the window shows the product code

[BCSW-CVC-HS-4-9-2], version number **[vB010]**, build number **[B1]** and the sample rate of **[8000 Hz]**.

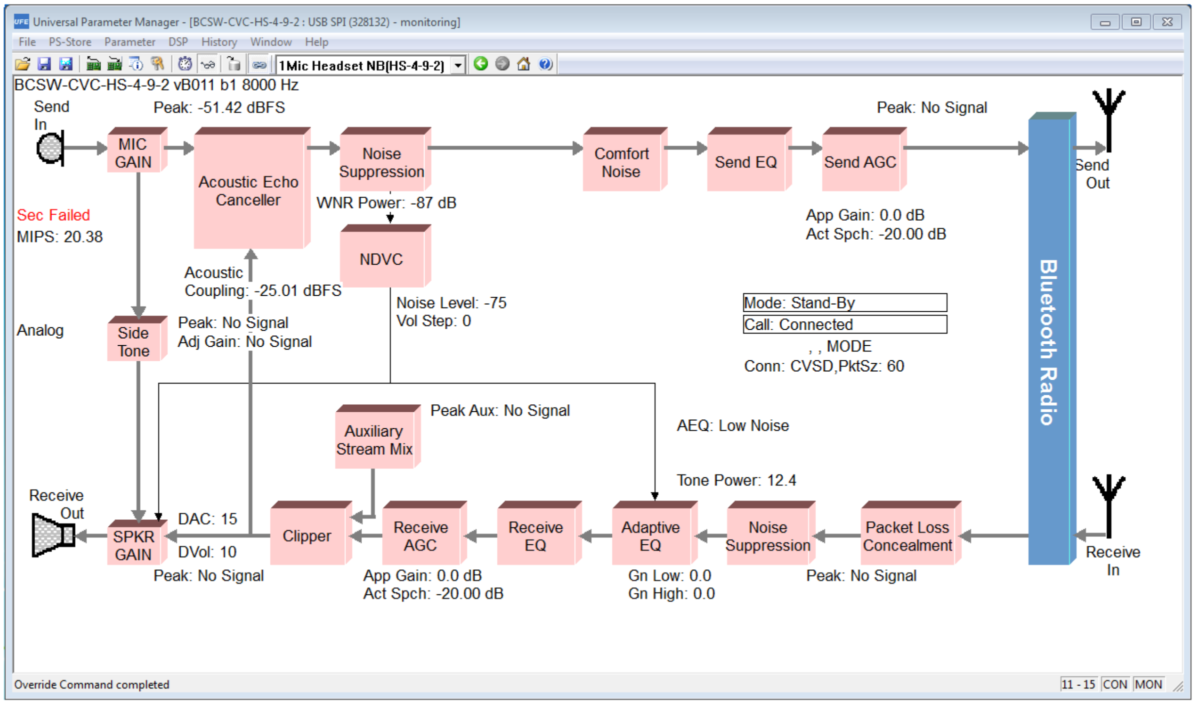


Figure 8-2 Parameter Manager window in an active call

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.

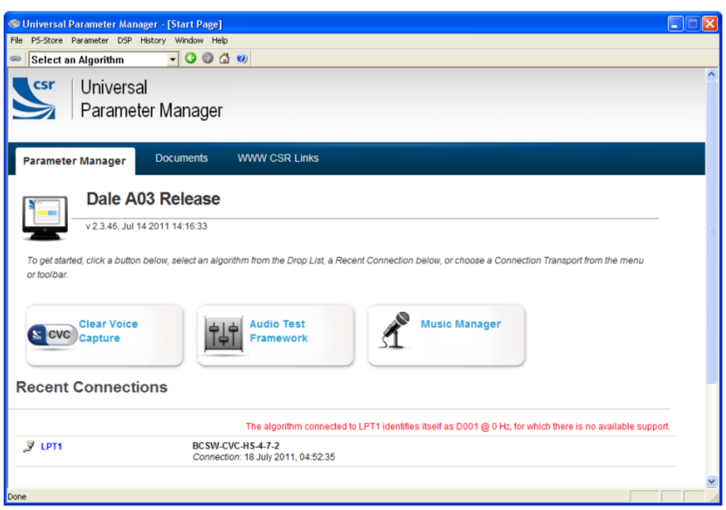


Figure 8-3 Parameter Manager window showing invalid version message

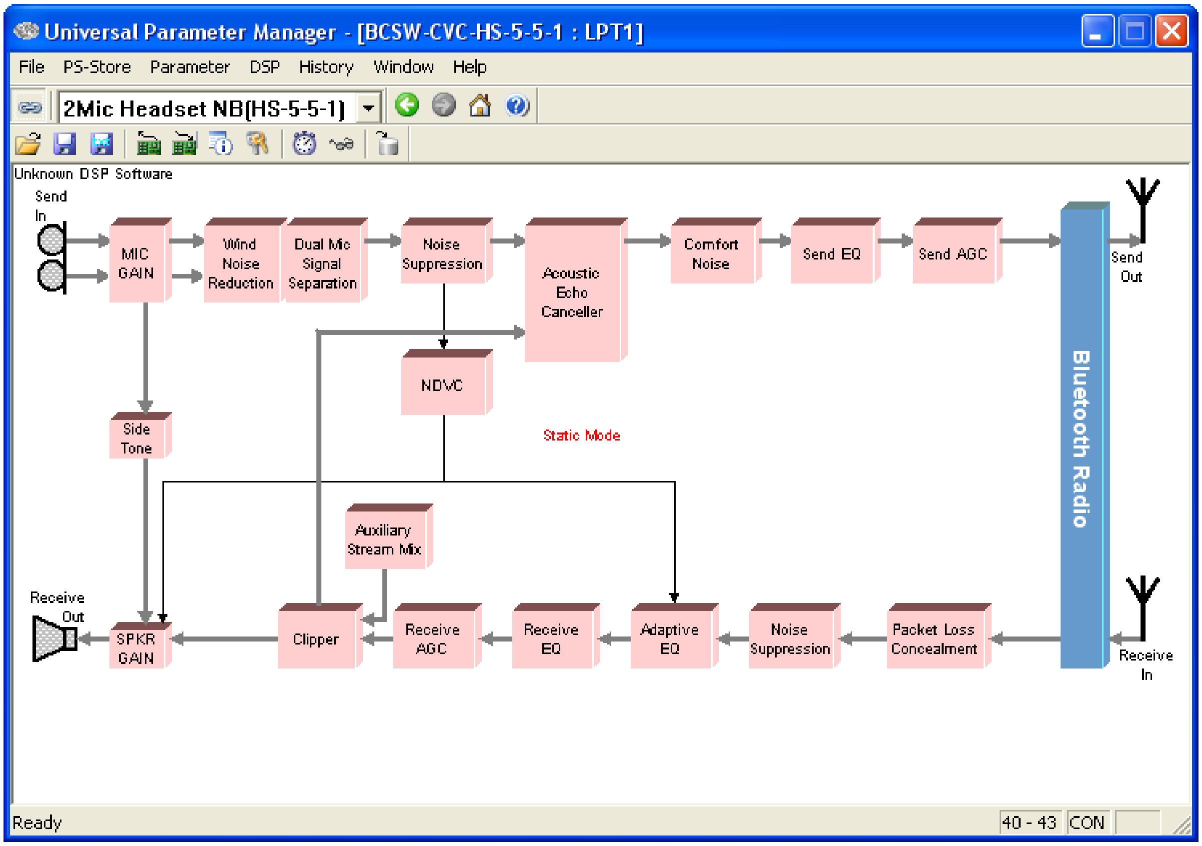


Figure 8-4 Parameter Manager window in Static Mode

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 displays: 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-HS-4-9-2 1M-HS Tuning Guide	80-CT419-1 /CS-00309814-UG
Enabling cVc for Headset SDKs	80-CT409-1 /CS-00122720-AN

Terms and definitions

ADC	Analog to Digital Converter
AEC	Acoustic Echo Canceller
AEQ	Adaptive Equalizer
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
ENR	Echo and Noise Reduction
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
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 Equalization
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