Qualcomm Technologies International, Ltd.

ANC: ADK 4.3 with CSR8675

User Guide

80-CF386-1 Rev. AA

November 15, 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
AA	November 2017	Initial release Alternate document number: CS-00406548-UG

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1 ANC overview

Qualcomm® Ambient Noise Cancellation (ANC) is released as an add-on feature to the Audio Development Kit (ADK). ANC is only supported on CSR8675.

The ANC 1.0 release consists of:

- ANC Tuning Tool, which is a stand-alone PC application for designing ANC filters and calibrating gains
- VM code and libraries to enable ANC features in a CSR8675 headset sink application
- Example VM application for tuning and debugging

These tools enable a user to design a Bluetooth headset, which includes ANC. Features included with ADK, such as high-quality Bluetooth music streaming and voice calls, co-exist with ANC. ANC can be used in a stand-alone mode when Bluetooth streaming is not required.

2 ANC architecture

ANC 1.0 release uses a feed-forward architecture, which requires two microphones, one mounted on the outside of each ear cup. These microphones pick up external noise, which is inverted, filtered, and mixed into the headset output. This creates anti-noise in the ear cup which, when mixed with ambient noise, eliminates noise for the end user.

ANC uses the sidetone path, which is a low latency path between the ADC and the DAC that contains an integrated IIR filter. The ANC signal does not pass through the Qualcomm® Kalimba™ DSP. However, voice and music processing done by the Qualcomm Kalimba DSP DSP is mixed at the DAC with ANC audio.

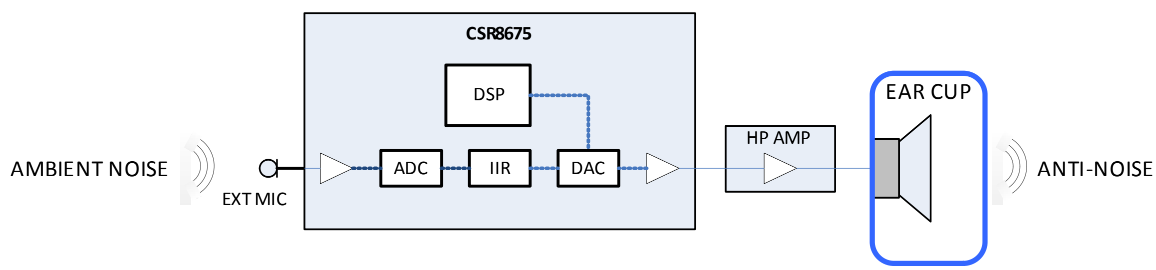

Figure 2-1 ANC feed-forward system diagram

The ANC Tuning Tool designs the IIR filter coefficients. The ANC Tuning Tool simplifies customization and tuning of the ANC filters for specific headset acoustics. The output of the tuning process is a set of gains and coefficients for each channel, which can be downloaded to the device.

To design the IIR filter, the ANC system is modeled into two blocks called *plant* and *controller*.

2.1 Plant

The plant represents the transfer function between the headset speaker and the internal reference microphone.

This transfer function includes the signal delay introduced by the CSR8675 IC and acoustic path inside the ear cup. The ANC Tuning Tool calculates the plant using a recording made by playing noise to the headset speaker and recording it on an internal microphone.

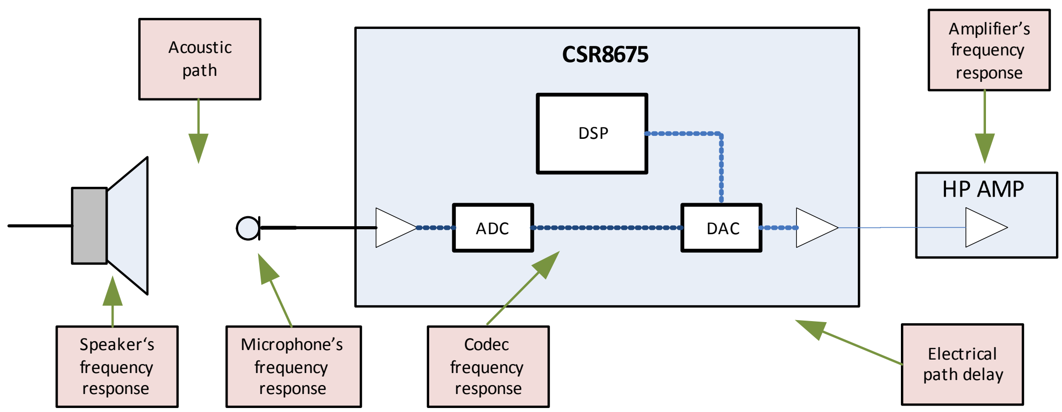

Figure 2-2 Plant

2.2 Controller

The controller is a calculated filter that is based on the transfer function from an external microphone to an internal microphone and the plant.

The controller recording is captured by playing noise into the test chamber and simultaneously recording from internal and external microphones. The ANC tuning tool models the controller as a FIR filter, which is later converted to an IIR filter in the appropriate format to download to a CSR8675 device.

NOTE Recoding between the two microphones must be time-aligned.

2.3 Sidetone

The CSR8675 has a low-latency sidetone path between the ADC and the DAC with an integrated IIR filter. In addition to the IIR filter, there are a series of gain stages available.

The programmability of the IIR and gains enables the ANC feature to be tuned and customized to the customer headset acoustics.

See the *CSR8675 Data Sheet* for details on the sidetone hardware.

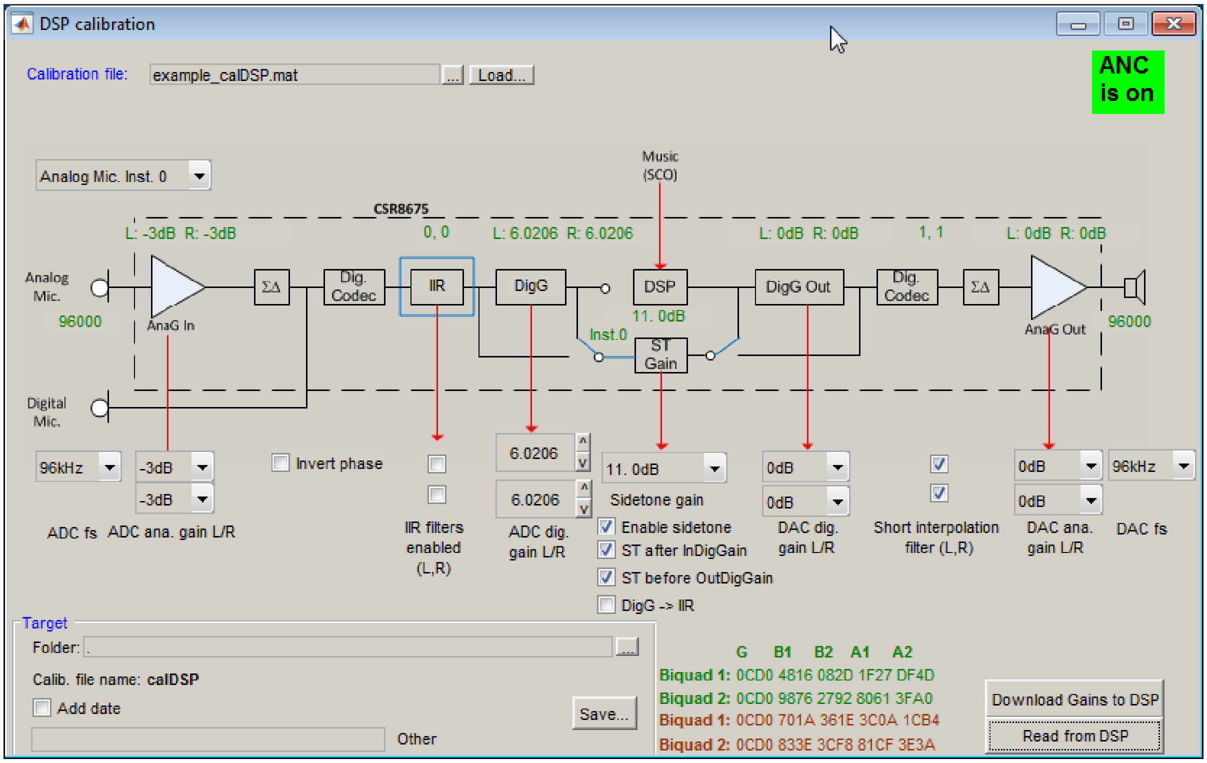

Figure 2-3 Calibration panel available gains

2.4 ANC Latency

For ANC to be effective, the latency of the ANC signal path must be as low as possible.

For the best latency, use the CSR8675's maximum sampling rate of 96 kHz for the ADC and 192 kHz for the DAC. This latency is measured using a 1 kHz sine tone with the IIR filter disabled and DAC in noninverting mode.

Table 2-1 CSR8675 latency specification

Sampling rate	Typical latency (ADC ro DAC)
96 kHz ADC, 96 kHz DAC	110 µs
96 kHz ADC, 192 kHz DAC	55 µs

2.5 Microphones

The CSR8675 supports both analog and digital microphones.

The Sink Configuration Tool can configure microphone connections. A microphone can be used in a shared configuration where it can serve both as an ANC microphone and voice microphone.

NOTE CSR8675 can support a total of 6 mic inputs (6 digital or 2 analog and 4 digital), which can support systems that implement 2-mic voice processing (such as Qualcomm cVc noise cancellation technology) simultaneously with ANC.

2.6 ANC performance

The acoustic design of the headset primarily determines attenuation performance of ANC. A headset with poor acoustics may result in poor performance

This document assumes that you understand proper acoustic design for ANC feed-forward headsets and are familiar with measurement techniques for ANC headset.

3 ANC development

The first step of ANC development is to acquire suitable hardware and software. Qualcomm recommends the CSR8675 development board H13179v2 with an H13774v1 extension board. A customer-produced PCB with a CSR8675 can also be used. This document gives instructions based on the recommended Qualcomm development board.

The ANC requires:

- ADK
- ANC add-on for ADK
- MATLAB run-time environment version 8.4 (32-bit) to support the ANC tuning tool (download from mathworks.com)
- CSR8675 development platform (H13179v2 board with H13478v2 module and H13223v1 headset amplifier board)
- USB - SPI hardware
- Recording software, such as Adobe Audition
- Prototype headset with external microphone and speaker connections externally available for connection to development board (optional: internal microphones in ear cup)
- Professional-grade audio interface such as RME Fireface UC
- Head and torso simulator (HATS) or equivalent acoustic test fixture (with in-ear microphones)

3.1 CSR8675 development board setup

Figure 3-1and Figure 3-2 show the development board platform with USB-SPI interface and prototype ANC headset. The prototype headset has the microphones and speakers wired to the connectors on the development board.

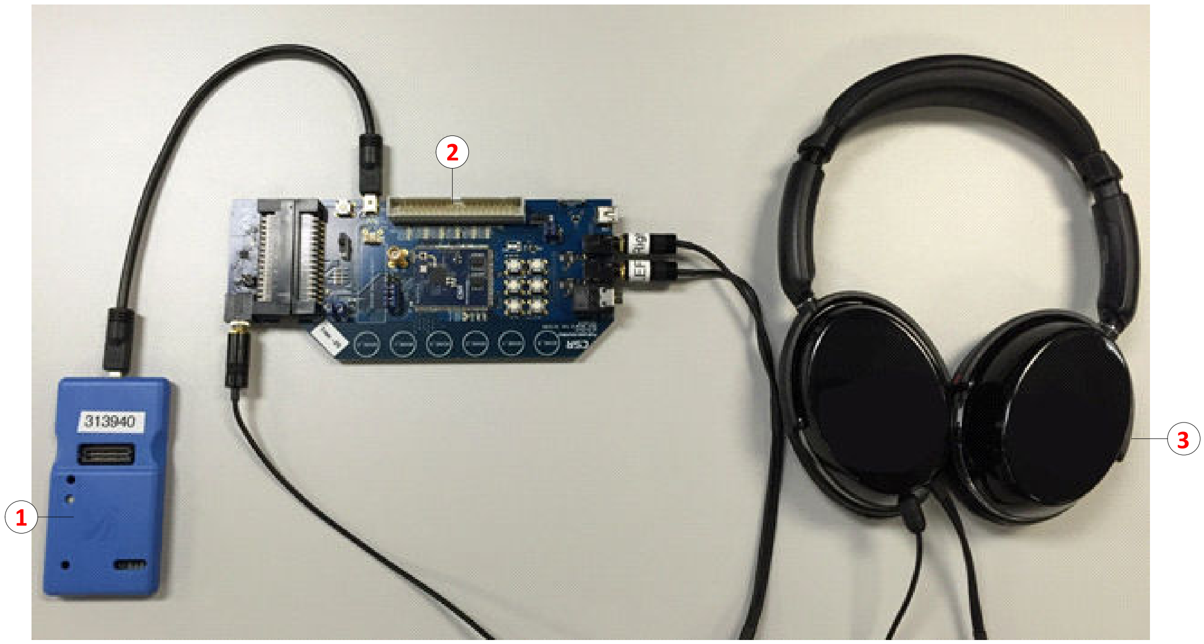

Figure 3-1 Development board with SPI interface and prototype headset

The labels refer to:

1. USB-SPI interface for communicating with CSR8675
2. CSR8675 development platform
3. Prototype ANC headset design with microphone connections available on 3.5 mm connectors

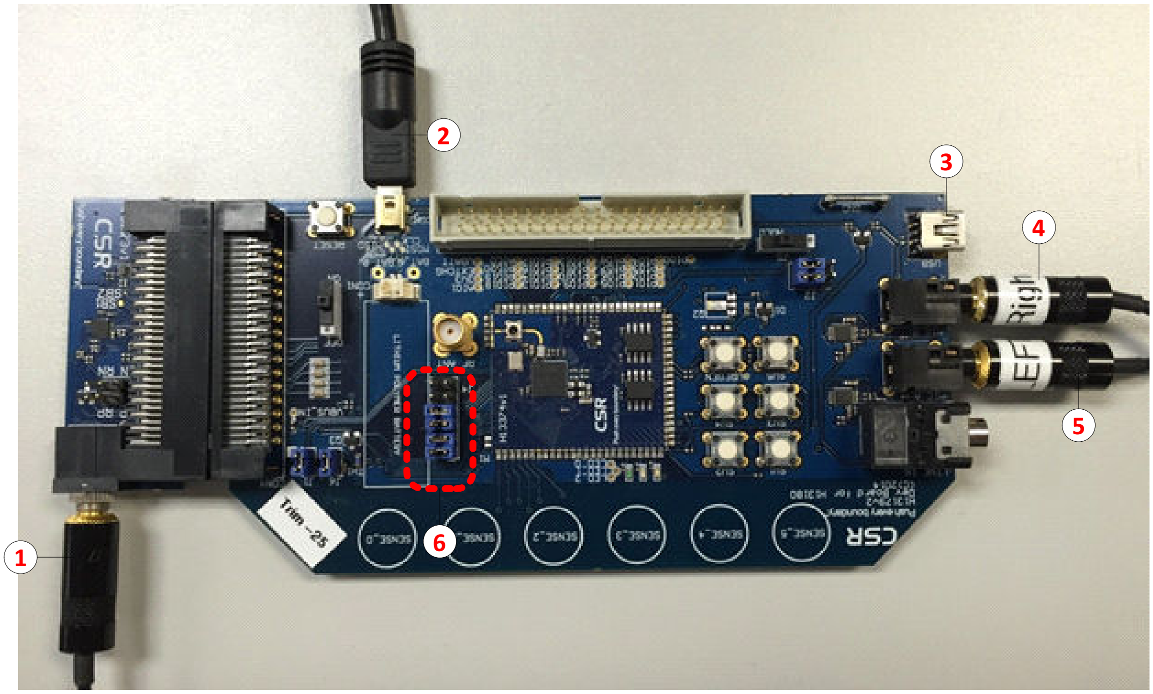

Figure 3-2 Development board with audio connections

The labels refer to:

1. Headset output connector. Stereo 3.5 mm jack to drive the headset
2. SPI cable
3. USB connector for power and USB audio (when applicable)
4. Right microphone input, 3.5 mm jack, mono
5. Left microphone input, 3.5 mm jack, mono
6. Red outline around differential audio connector for DAC output and ADC input. Useful when designing for differential inputs, debugging, or connecting to the DAC output before the headset amplifier.

3.1.1 Modifying the input capacitor

To achieve better low frequency response on the analog microphones, modify the microphone input capacitors. Change C8 and C13 from 150 nF to 2.2 uF. This change affects signals connected to the 3.5 mm mic input jacks CON5 and CON6. Without these modifications, attenuation at 100 Hz and below is difficult to achieve.

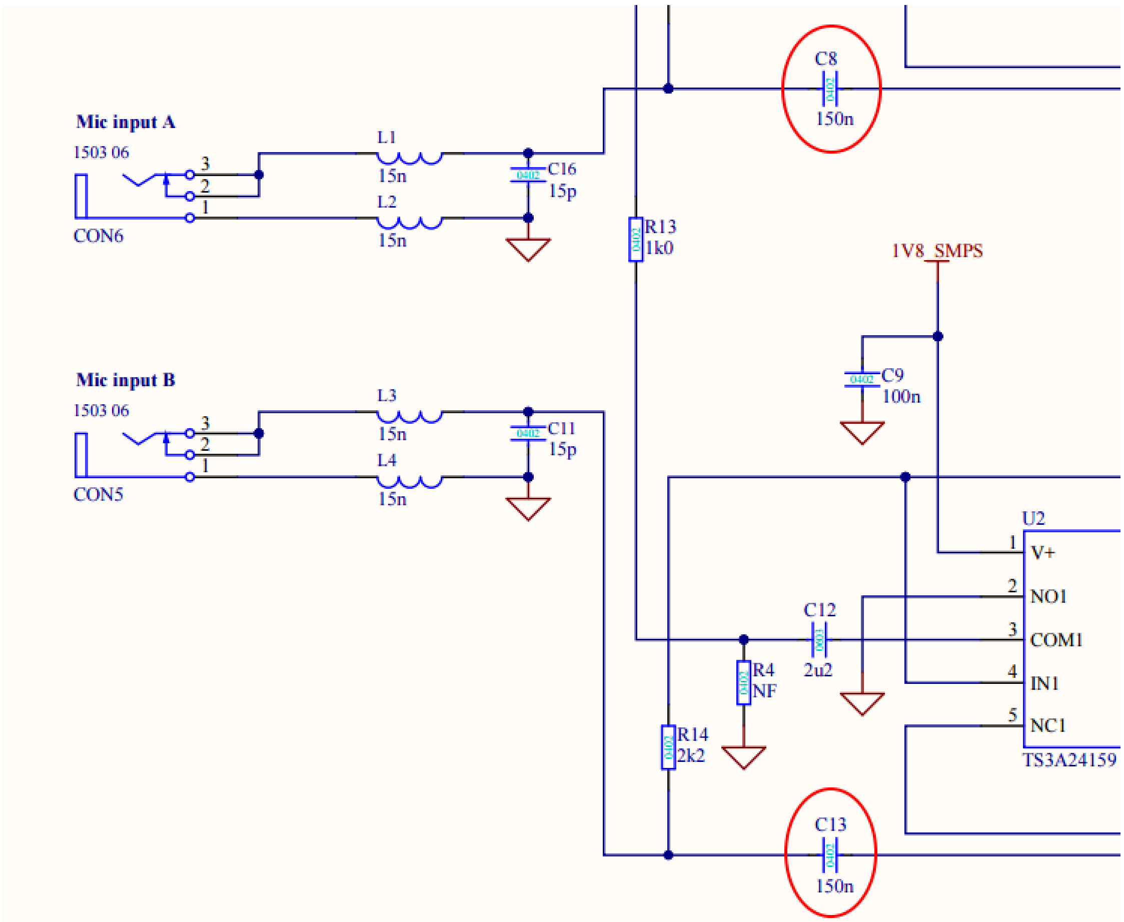

Figure 3-3 Input capacitor modification

NOTE If you use the differential connections on J4, the capacitor modifications are not required. However, this input does not provide any bias voltage, so you must provide the voltage externally or wire out the bias from connector J2.

For details, see the CSR8675 Carrier Board Schematic (H13179v2).

3.2 ANC example application

The ADK provides an example application that supports ANC. The ANC example app is a minimal VM application that only loads the components required for ANC. The application is available in the ADK folder: **<ADK> \apps\examples\anc_example.**

The application only routes signals required for ANC without the overhead of any audio processing, Bluetooth, or other features. This routing simplifies the setup of ANC laboratory evaluation, as opposed to using the sink application, which requires more configuration.

The example application loads coefficients and gains from the PSR file generated by the ANC tuning tool. To start an example PSR is provided with initialized settings. Appendix A describes the relevant PS Keys.

The ANC example app is in the ADK examples folder.

3.3 ADK headset application with ANC

The ADK provides a sink application that supports ANC. ANC features are integrated with the ADK sink application for headsets, where ANC can be used alongside voice and music streaming. See the *Audio Sink Application ANC User Guide* for details on how to build and configure this application.

The ANC Tuning Tool generates a PSR file as the final output of the tuning process. The ADK sink application uses this PSR file to load the ANC filter coefficients and gains. The Sink Configuration Tool configures the ADK headset.

3.3.1 ANC software tools and dependencies

The ANC tuning tool requires the MATLAB runtime to function. Install MATLAB runtime 8.4 32-bit, downloadable from www.mathworks.com.

More tools that are required and are included with the ADK are:

- PSTool
- Qualcomm® BlueLab™
- BlueFlash (part of Qualcomm® BlueSuite™)

3.4 Security key

ANC requires a security key. Qualcomm provides the security key when ANC licenses are purchased. Each key is only valid with specified Bluetooth addresses. Use PSTool to write the key to flash on the CSR8675. Qualcomm provides an evaluation security key for development purposes.

Table 3-1 Evaluation security key

Key address	Key value	Valid Bluetooth addresses
`0x2601` (License Key for Feature)	3465 A979 BC24 3B9F 1209	`0002-5B-00FF01 to 0002-5B-00FF05`

Use PSTool to enter the security key.

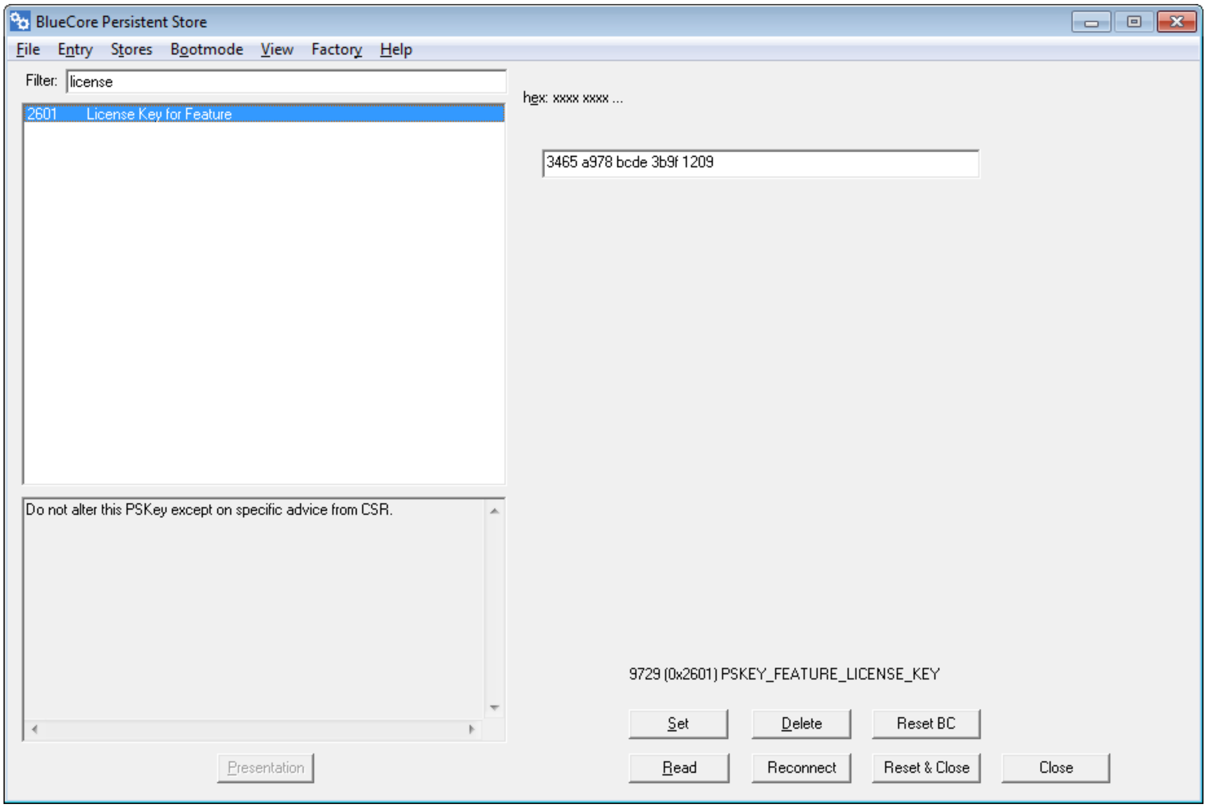

Figure 3-4 Entering security key in PSTool

4 ANC tuning tool

The ANC Tuning Tool is a stand-alone PC application included with the ANC add-on installer. The tool provides a process for calibrating gains and calculating the ANC coefficients.

The user must make a series of recordings in the laboratory using a prototype headset, which is used as inputs to the tool. These recordings require audio recording software such as Adobe Audition.

The graphical user interface (GUI) provides a complete environment that enables the user to execute all the design functions for the ANC, ending with downloading the final filter coefficients and gains into a CSR8675 device.

NOTE The tool requires the MATLAB runtime to function. Install MATLAB runtime 8.4 32-bit, from the mathworks website.

4.1 ANC tuning tool main panel

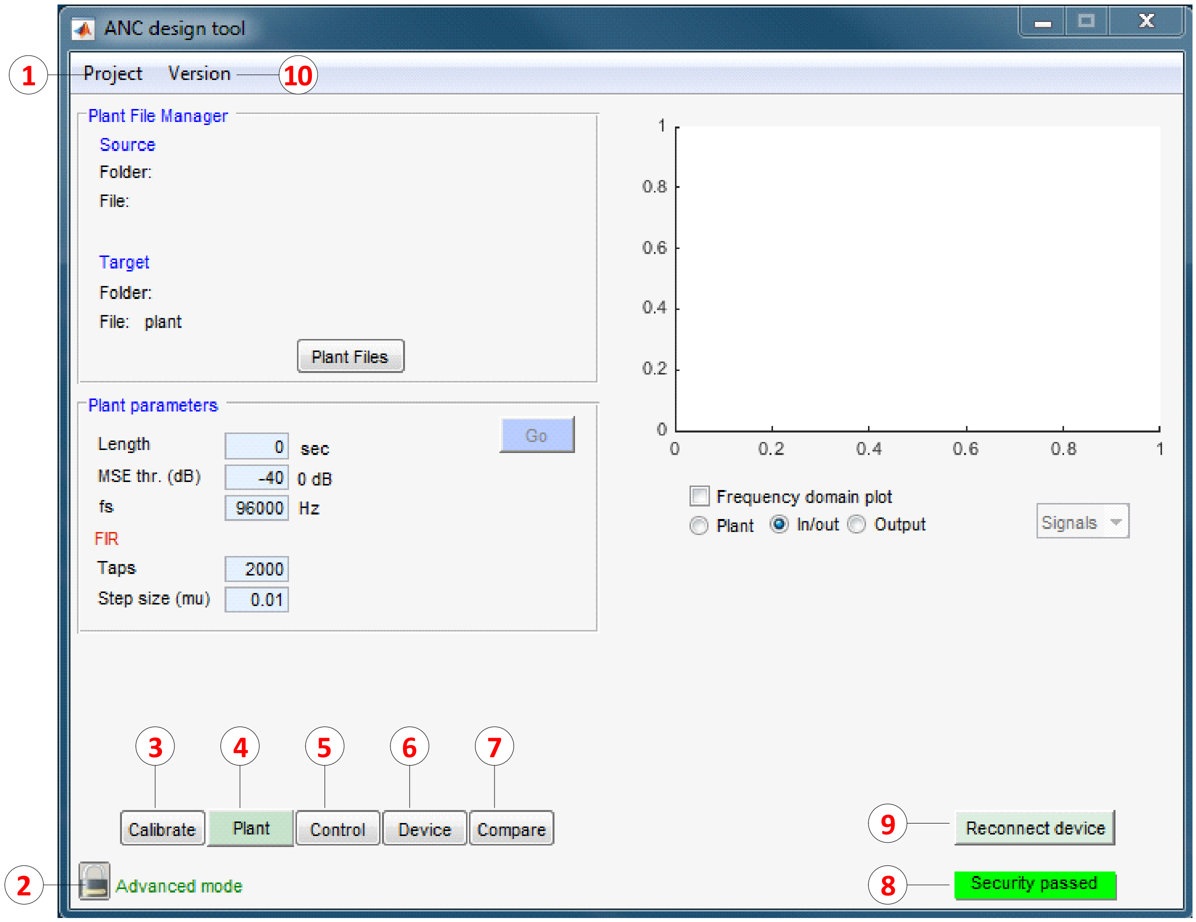

Figure 4-1 ANC tuning tool main panel

The labels refer to**:**

1. **Project menu:** Manages the design of a project to save and load different configurations. It contains these menu items:
   a. **New project:** Creates a new project, assigning default folders for wave files and output files. All other data is cleared.
   b. **Save:** Saves the current information from the design panels in a file.
   c. **Load:** Loads a previously saved project, populating all the design panel controls with previously saved data.
   d. **Save as:** Creates a new project based on the current information from the design panels.
2. **Advanced mode:** By default, the GUI is in advanced mode, which reveals all design options.
3. **Calibration panel:** Configures the different gains in the sidetone, along with routing, monitoring, and IIR enabling/disabling.
4. **Plant Panel:** Calculates the transfer function from the headset speaker to the internal microphone.

5. **Controller design:** Calculates the ANC controller based on a previously calculated plant and recordings from the internal (in ear) and external microphones.
6. **Device download:** Generates an output PSR file from a calibration file and an ANC controller file. The PSR can be downloaded to the CSR8675 device.
7. **Performance comparison:** Compares the performance of different ANC systems, based on both passive-noise cancellation and active-noise cancellation recordings.
8. Display that indicates whether the connected device has a valid ANC license.
9. **Reconnect device:** Re-initiates a SPI connection to connected hardware.
10. **Version:** Provides version information for the installed ANC Tuning Tool.

4.2 ANC tuning tool calibration panel

The calibration panel shows a block diagram of the ANC system with all associated gain blocks. This panel calibrates the gains to get the correct signal levels for ANC. Additionally, the display of gains, sample rate, and sidetone routing are useful for development and debugging.

The available controls are:

- ADC fs: ADC sampling rate.
- ADC ana. gain L/R: Sets the analog gain when using analog microphones. If digital microphones are used, this value is not applicable.
- Invert phase: Inverts the phase of the signal path.
- IIR filters enabled: Check to enable the IIR filter, uncheck to bypass (IIR filters need to be bypassed for the calibration and recording process)
- ADC dig. gain L/R: Sets the gain of the signal after it is digitized through the ADC. It is the only way to apply a gain to digital microphones.
- DAC dig gain L/R: Sets the DAC gain using the digital gain stage of the DAC.
- DAC ana. gain L/R: Sets the DAC gain using the analog gain stage of the DAC.
- Enable sidetone: Enables or disables the ANC algorithm. Useful when the user wants to disable it during testing.
- ST after InDigGain: Changes sidetone routing around ADC digital gain block (only the default value is supported in the ADK).

- ST before OutDigGain: Changes sidetone routing around DAC digital gain block (only the default value is supported in the ADK).
- DigG -> IIR: Changes order of IIR and ADC digital gain block (only the default value is supported in the ADK)

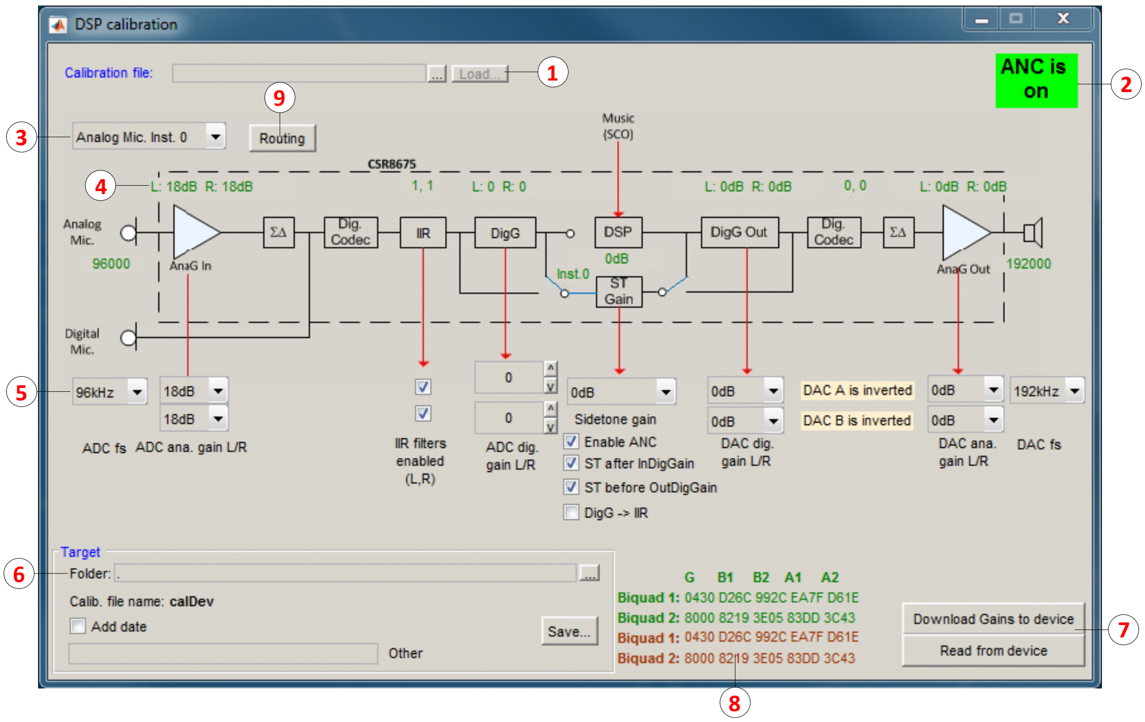

Figure 4-2 Calibration panel

The labels refer to:

1. Loads a previously saved calibration file.
2. Indicates the status of the ANC algorithm running on the chip:
   - ANC on (green)
   - ANC off (red)
   - No communication with the device (yellow)
3. Indicates microphone instance used for ANC.
4. Text in green shows current values on CSR8675 for gains and sampling rates.
5. Center section has controls for sampling rate and all gain blocks.
6. Use this section to save the gains to a calibration file. The user can choose the folder where it is saved and the filename to be used. The term `calDev` is always added to identify it as a device calibration file. This file is used when downloading the coefficients to the device.
7. These buttons write changes to the CSR8675 and read back the current values.

8. Shows the coefficient values in the IIR filter for left and right channels.
9. Opens the Microphone Routing panel
   Microphone Routing:

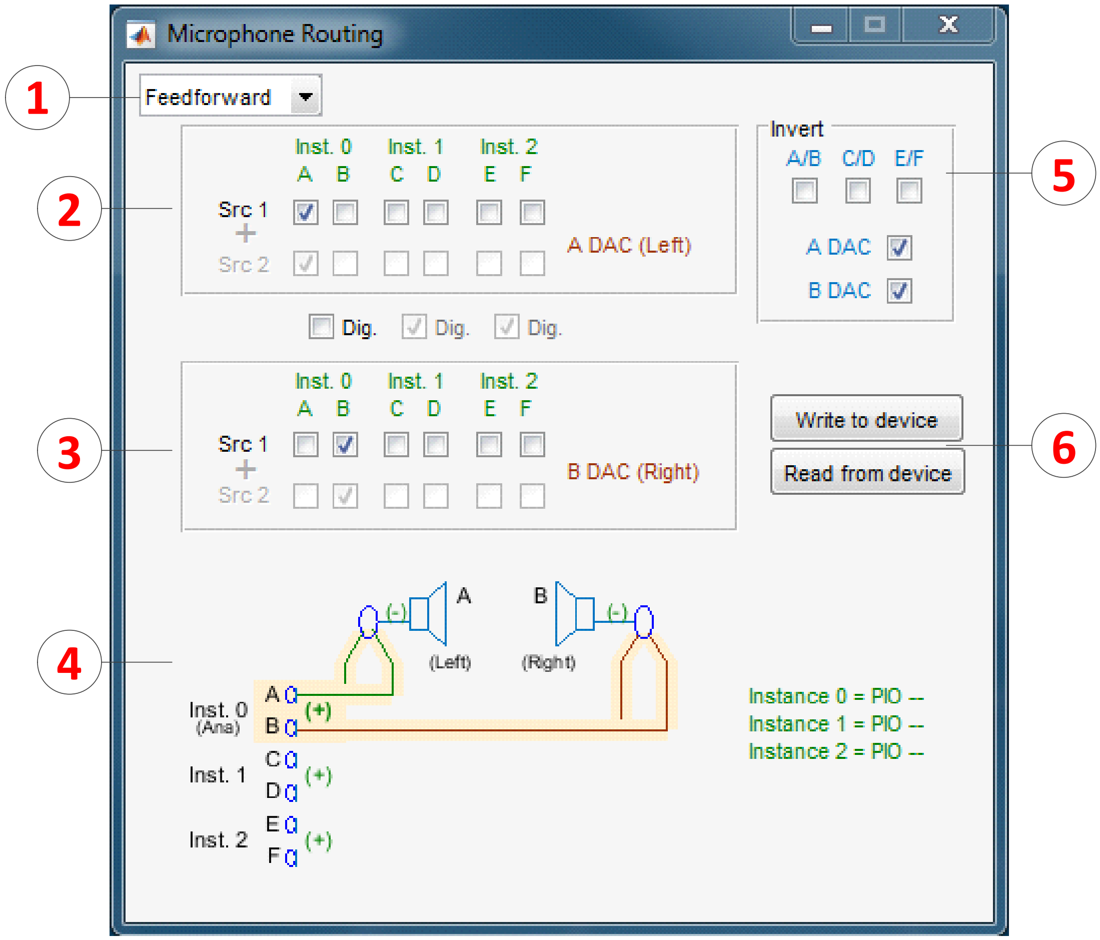

Figure 4-3 Microphone routing panel

The labels refer to:

1. Selects the ANC configuration being used (feedforward, feedback, hybrid).
2. Selects the signal going to the Left channel of the DAC. For analog microphones, the Left input corresponds to channel A of Instance 0, and the Right input to channel B of the same instance.
3. Selects the signal going to the Right channel of the DAC. For analog microphones, the Left input corresponds to channel A of Instance 0, and the Right input to channel B of the same instance.
4. Graphic depiction of the routing. The highlighted path matches the instance selected in the calibration panel.
5. Enables the user to invert the input signals (instances A/B – C/D – E/F) and the output channels (DAC A/B).
6. Write and read from the current device.

4.3 ANC tuning tool plant design panel

The plant panel calculates the plant, which is the transfer function of the acoustic path inside the headset ear cup, along with the electrical path introduced by the CSR8675.

Input the plant recordings (`.wav` files) and plant parameters to generate the plant. The output of the plant generation step is a `.mat` file, typically called `plant.mat`, which computes the controller.

The plant panel enables you to:

- Select the source wave files that contain the prerecorded signals.
- Select the filename that will store the plant information, to be used by the controller design panel.
- Select the required parameters.
- Plot the resulting transfer function, along with other important plots, such as the input and output signals, error signal.

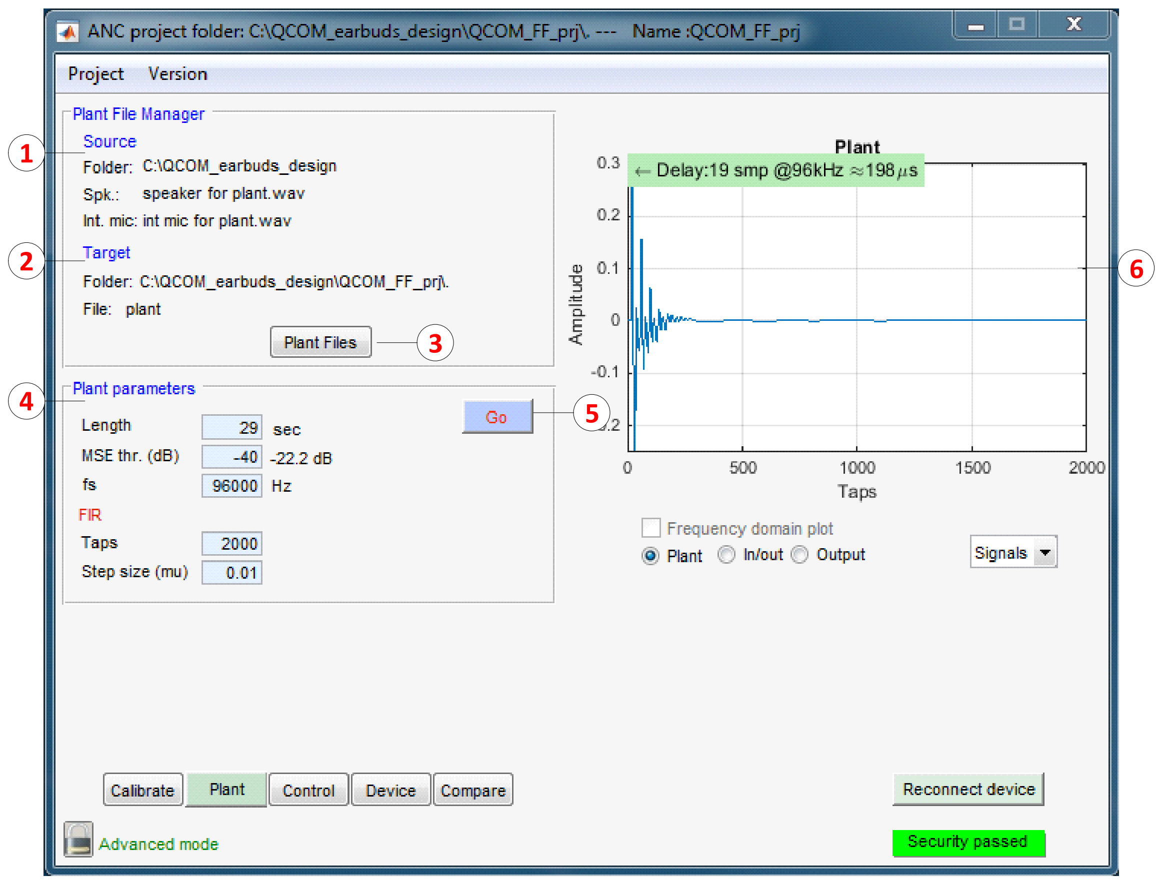

Figure 4-4 Plant design panel

The labels refer to:

1. Source folder and files.
2. Target folder and file.
3. Opens plant files panel.

4. Plant Parameters. Parameters that can be tuned are:
   a. Length: Length of time (in seconds) that calculates the plant. Recommended value is the length of the plant recording, typically 30 s.
   b. MSE threshold: When the Mean Squared Error (MSE) reaches this threshold, adaptation stops. Recommended value is -40 dB.
   c. Taps: Number of coefficients of the FIR filter. Recommended value is 2000.
   d. Step size: Adaptation rate. Select value to achieve best convergence. Typical values in the range of 0.01 to 0.1.
5. **Go**: Click to calculate the plant after loading files and setting parameters.
6. Current plot: Display various plots of the plant, such as:
   - Impulse response
   - Time domain
   - Frequency response

4.4 ANC tuning tool plant files panel

The plant files panel selects the recordings for plant calculation, with an option of having mono or stereo source files. This panel also enables the user to select the target folder and the filename that stores the results from the computation.

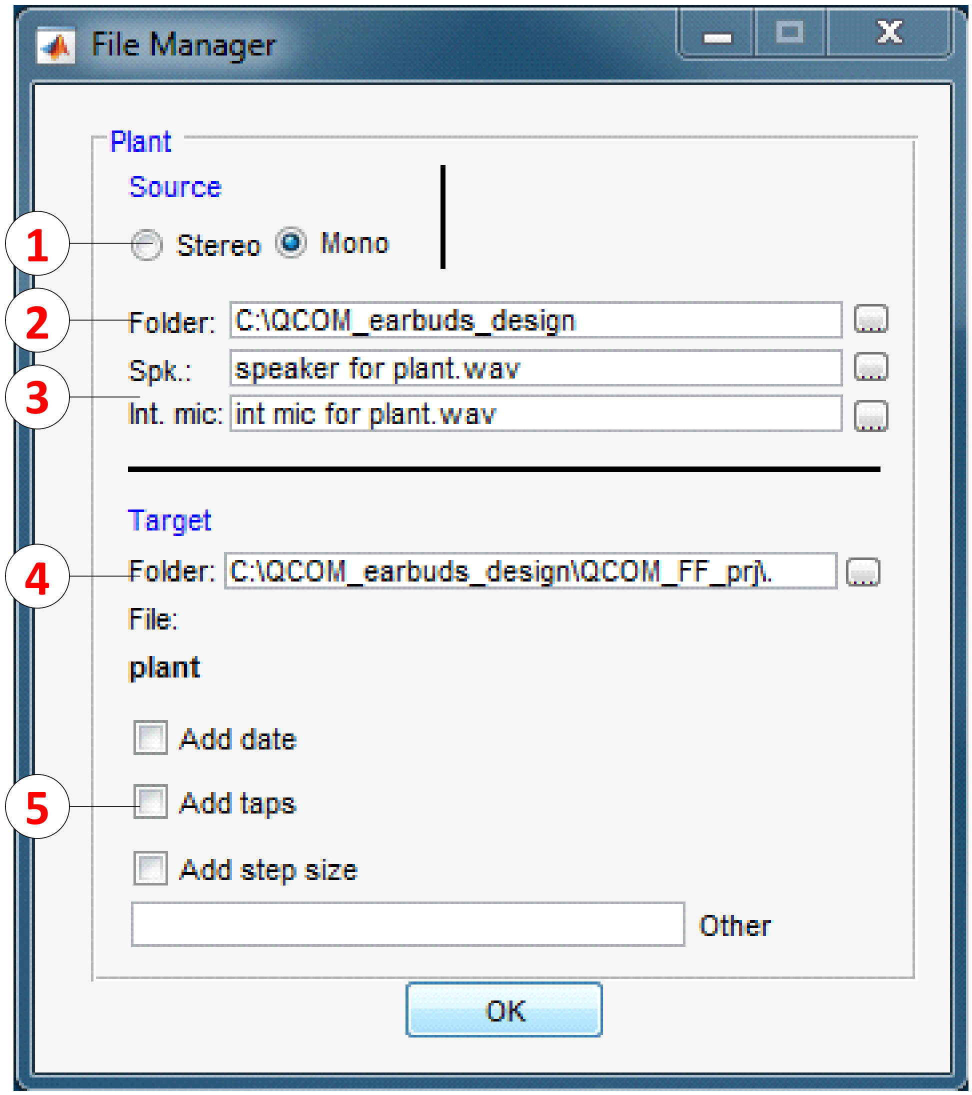

Figure 4-5 Plant files panel

The labels refer to:

1. Stereo or Mono file selection. Enables the user to use a stereo `.wav` file or two mono `.wav` files.
2. Folder: Selects the parent folder containing the recordings.
3. File: Selects one or more .wav files that contain the recordings for speaker and internal mic.

4. Target Folder: Selects the folder for calculated plant files to be saved.
5. File naming options. By default, the plant file generated is called `plant`. Optionally the date, number of taps, step size, or other arbitrary string can be added to customize the filename. These additions are useful if multiple plant files are generated when debugging.

4.5 ANC tuning tool controller design panel

The controller design panel enables the user to calculate the controller, based on the transfer function between the speaker and the internal microphone (plant) and the internal/external microphone recordings.

The controller panel enables the user to:

- Select the source `.wav` files that contain the prerecorded signals.
- Select the filename that stores the controller informat. This filename is downloaded to the CSR8675 in a subsequent stage of the design.
- Select the required parameters.
- Plot the resulting controller, along with other important plots, such as the input and output signals, and error signal.

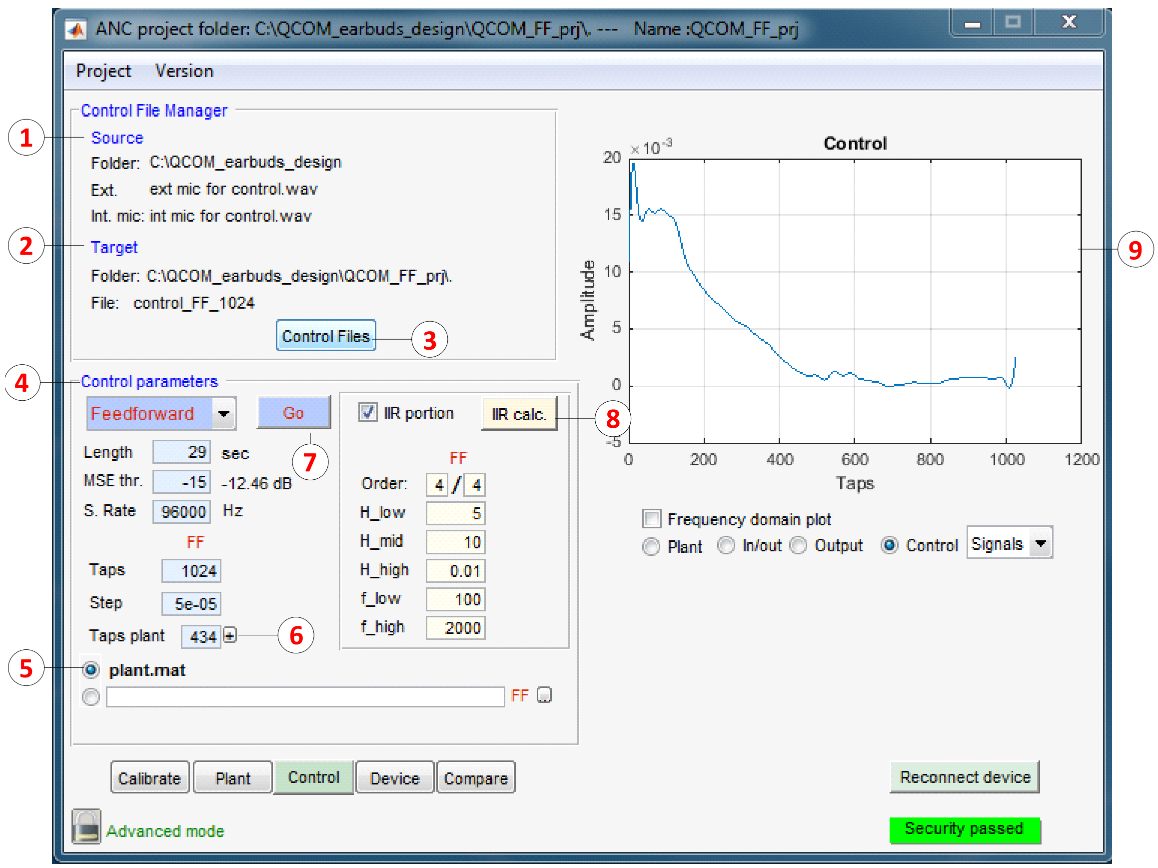

Figure 4-6 Control panel

The labels refer to:

1. Source folder and files
2. Target folder and file
3. Control Files button. Opens the control files panel to choose recordings to use
4. Parameters. Parameters that can be tuned in this section are:
   a. Length: Length of time (in seconds) that calculates the controller. Recommended value is the length of controller recording, typically 30 s.
   b. MSE threshold: When the Mean Squared Error (MSE) reaches this threshold, adaptation stops. Recommended value is -30.
   c. Taps: Number of coefficients of the FIR filter. Recommended value range 640 to 2500 taps.
   d. Step size: Adaptation rate. Recommended values in the range of 0.000001 to 0.001. Choose step size to get the best filter convergence (smallest MSE error).
   e. S. Rate: The sampling rate being used, based on the recordings loaded. S. rate should always be 96 kHz for CSR8675 ANC designs.
   The parameters inside of the IIR portion (yellow) cannot be changed from here. Modify them from from the IIR panel (button 8).
5. Plant source. Plant to be used by the controller algorithm. The first option is the last plant calculated in the **Plant design** panel. The second option enables the user to load a previously calculated plant.
6. Taps plant: Number of coefficients to be used from the plant. The **+** button enables the user to select the zero-crossing graphically
7. **Go**: To start the algorithm that computes the controller, click **Go**.
8. IIR calculation: Opens the IIR Controller Design panel.
9. Current plot: Can display plots of the controller such as:
   - Time domain signals
   - Result
   - FIR, IIR, and quantized IIR impulse response
   - FIR, IIR, and quantized IIR frequency response
   - Spectrum attenuation
   - Zero-pole plot

4.6 ANC tuning tool IIR controller design panel

When the FIR portion of the controller is calculated (which takes most of the processing time), the user can design the IIR filters based on the precalculated FIR filter, without performing the full analysis process. This speeds up the design, because parameters can be changed without having to go through the FIR calculation. It also enables the user to fine-tune the IIR performance.

NOTE To use more taps on the FIR or use a different plant, go through the online controller design process again.

The IIR controller design panel provides some of the same parameters as the other controller design panel.

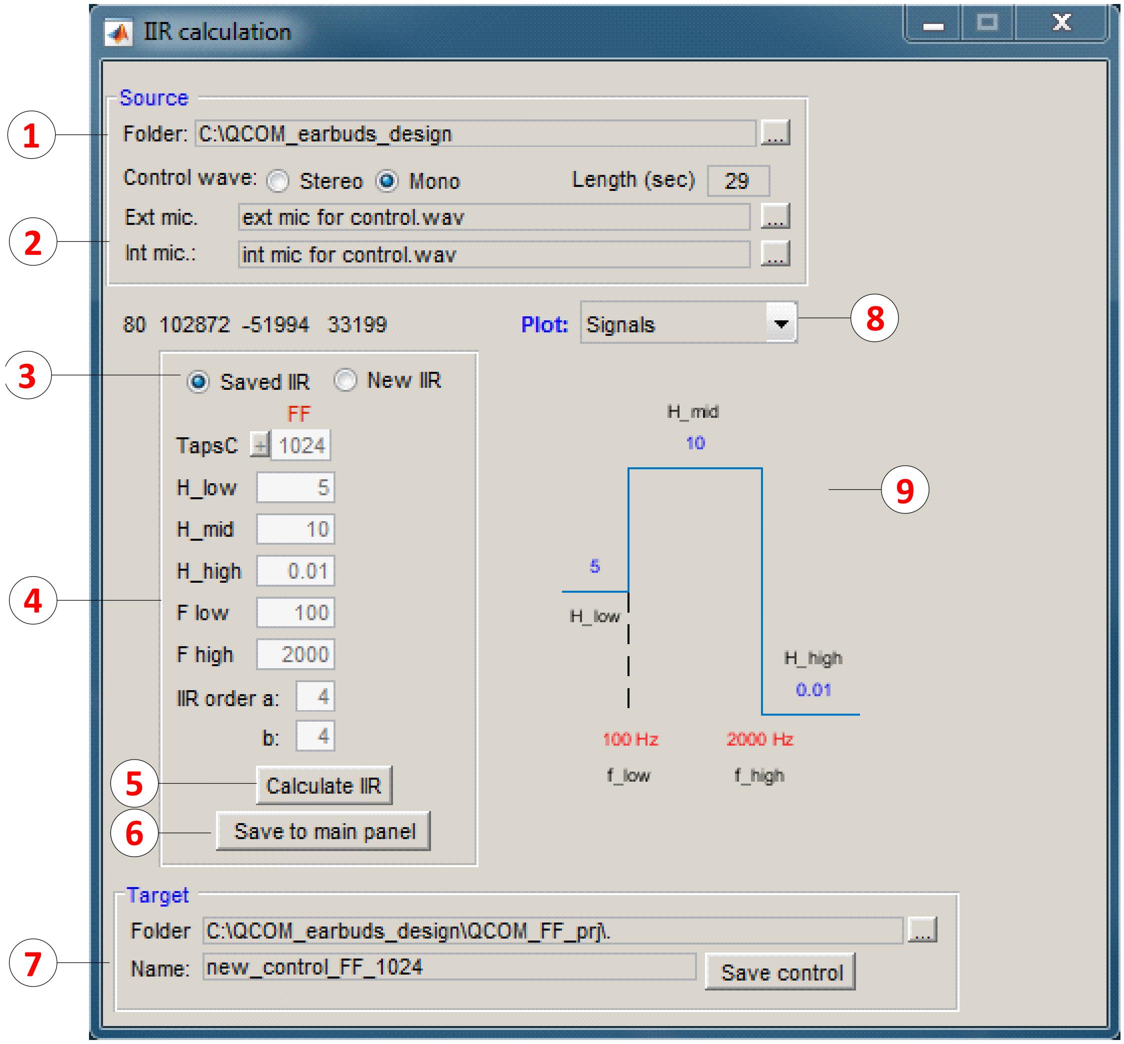

Figure 4-7 IIR calculation panel

The labels refer to:

1. Source folder.
2. .Wav file selection.
3. Switch between current IIR parameters and new IIR parameters.
   Parameters:
   - Taps C: Number of taps to be used from the FIR controller. Recommended value range 640 to 2500 taps.
   - H_low: IIR computation weighting for the frequencies below `F_low. Recommended value range from 1 to 10.`
   - H_mid: IIR computation weighting for the frequencies between `F_low` and `F_high. Recommended value range from 1 to 10.`

- ◻ H_high: IIR computation weighting for the frequencies above `F_high. Recommended value range from 0`.01 to 1.
- ◻ F_low: Transition frequency lower bound. Recommended value from 70 Hz. to 2500 Hz. F_low must be less than F_high.
- ◻ F_high: Transition frequency upper bound. Recommended value range from 700 Hz. to 3000 Hz. F_high must be greater than F_low.
- ◻ Control order: Denominator order (from 1 to 4) and numerator order (from 1 to 4). Recommended value is 4, which is the maximum filter order available in the CSR8675.

4. Calculate IIR. To calculate a new controller, click Calculate **IIR**.
5. Save to Main Panel: Copy new parameters to main Panel.
6. Target folder and filename. Use to save the newly designed control file.
7. Plot figure of newly designed filter.
8. Graphical representation of the IIR parameters.

The IIR weighing parameters are relative weighting coefficients. These weightings determine how to best match the IIR curve to the FIR curve. A lower weighting means that region of the filter is given less priority to match as areas with higher weighting. Select the weighting until achieving a satisfactory FIR to IIR matching and corresponding theoretical attenuation.

The frequency region from 100 Hz to 2 KHz is given a weight of 10, which is greater than the relative H_low and H_high coefficients (5 and 0.01 respectively). In this case, the filter design algorithm gives greater priority to match the IIR and FIR between 100 Hz and 2 kHz.

4.7 ANC tuning tool device panel

The device panel provides the interface to generate a `.PSR` file with the ANC gains and coefficients calculated by the ANC Tuning Tool for download to a CSR8675. The user selects the appropriate calibration and controller files to generate the PSR. The PSR file is the final output of the tuning process.

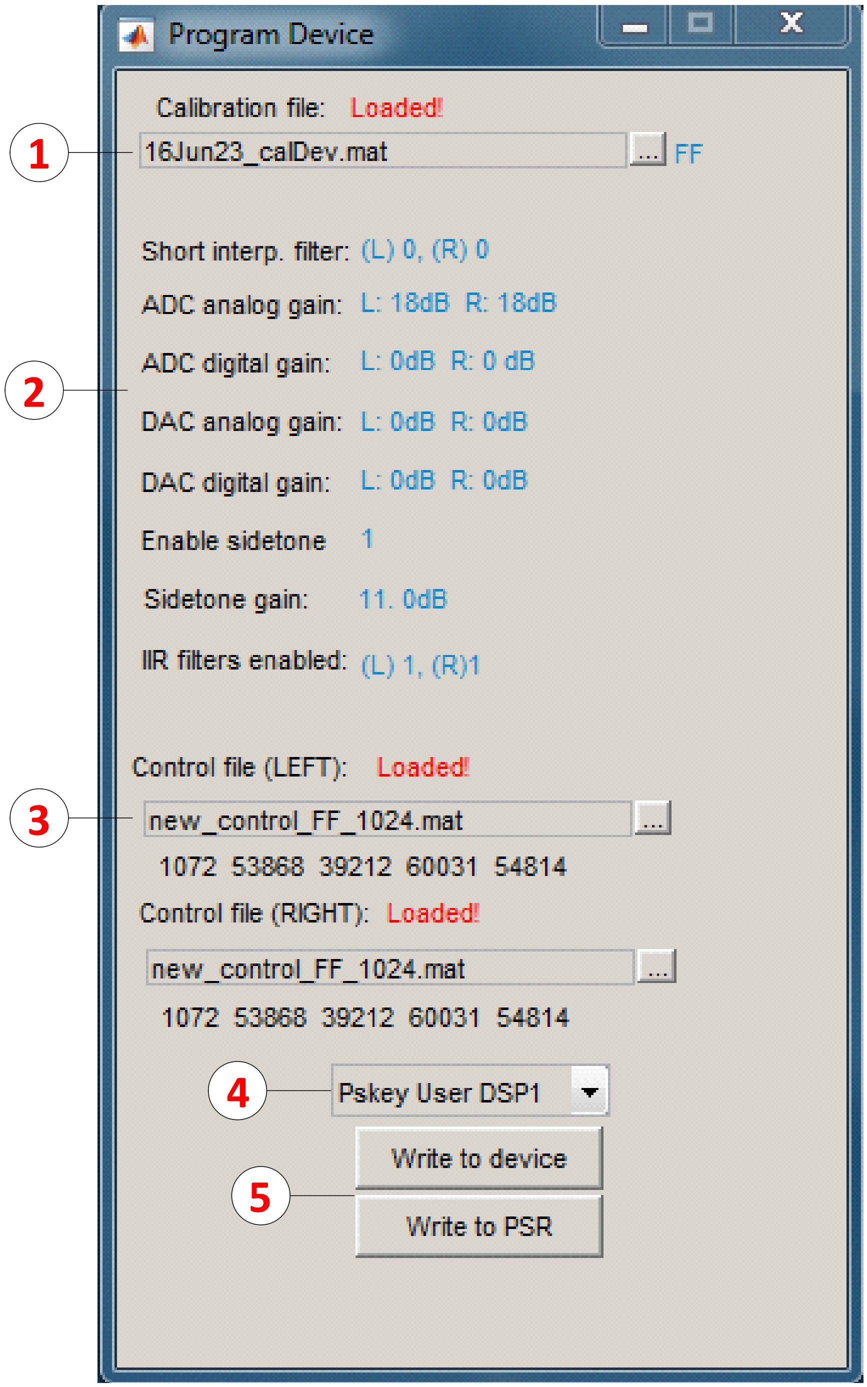

Figure 4-8 Device panel

The labels refer to:

1. Calibration file selection.
2. Displayed gain values from selected calibration file.
3. Control file selection for right and left channels.
4. The PS Key to write coefficients and gains to. The ADK supports DSP1 for ANC keys. DSP2 is used for leak-through mode, but can also be used for a second ANC filter. The other PS Keys are not supported by the ADK by default, but could be used in a custom application.
5. Buttons to write the design to a connected board or to a PSR file.

4.8 ANC tuning tool compare panel

The compare panel provides a quick way to evaluate the ANC performance of a completed design. Use this panel to input different recordings made with ANC active and with ANC disabled, and view the amount of attenuation between the two recordings.

The compare panel uses two recordings per configuration, where it:

- Calculates the frequency response of a signal with no ANC (only the passive attenuation provided by the headset).
- Subtracts from it the frequency response of the ANC algorithm (active), producing the total attenuation of the system.

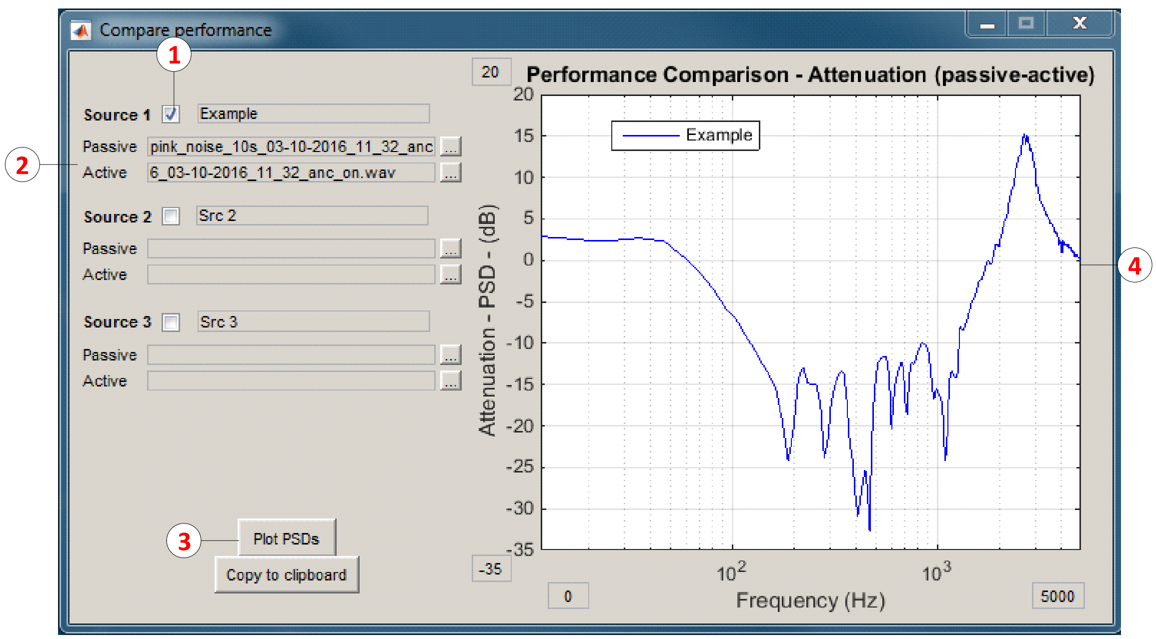

Figure 4-9 Compare panel

The labels refer to:

1. Selects if want to plot this specific source and provide a plot name
2. File selection for passive and active recordings to compare.

3. Buttons to update plot and copy plot to clipboard.
4. Plot area.

4.9 ANC tuning tool project structure

The ANC Tuning Tool saves many files in the project folder.

Typical files include:

- `plant.mat`: Plant data
- `plant.txt`: Text file containing plant information
- `control_FF.mat`: Control data
- `control_FF.txt`: Control parameters in text format
- Various `.jpg` files: Saved screenshots from each design step
- `15Jun26_calDev.mat`: Saved calibration data with date string included in the filename
- Any `.psr` files created are saved at the same location as the `control.mat` file
- Project recordings (`.wav` files) can be saved anywhere, but Qualcomm recommends saving them in the same folder as the rest of the project files.

5 ANC calibration

The first step in the ANC design procedure is to adjust the various gains in the CSR8675 and the levels in the test equipment. Calibrated gains must be properly tuned to complete the tuning process. The calibration panel of the ANC tuning tool GUI provides control over all the gains in the CSR8675 sidetone.

To ensure the recording levels in the `.wav` files (in dBFS) match the digital level (in dBFS) inside the CSR8675, calibrate the sound card gains and sound pressure level of the ambient noise in the test chamber.

When the calibration is complete, all gains are set to the correct value for the remainder of the tuning process. A calibration file is saved when the calibration process is complete.

The CSR8675 development board also contains a headset amplifier that drives a 3.5 mm output jack. This amplifier has a nominal gain of 9 dB. The differential audio signals on the J4 header are before the headset amplifier in the signal path. See the CSR8675 development board documentation for more information.

NOTE The CSR8675 full scale level for the ADC and DAC is different. DAC output is +1.9 dB greater ADC input signal. A -5dBV signal at the ADC input results in -3.1dBV signal at the DAC output, when all internal gains are set to 0 dB.

For all calibration, the internal IIR filter is disabled (bypassed).

5.1 Calibrating the sound card input levels

1. On the sound card input channel, connect a function generator set to generate a 2 Vpp 1 kHz sine wave.
2. Use recording software to record the input from the sound card channel. Adjust the sound card recording level so the recorded peak-to-peak value is 0 dBFS full scale. The recording level can be adjusted usually through a gain adjustment on the sound card, either in on the software control panel or hardware knob on the front of the panel (varies between different sound cards).
3. Repeat the calibration for each input channel.
4. The sound card inputs are now calibrated to 0dBFS = 2Vpp.

Depending on the type of hardware used, you may calibrate to a level other than 2 Vpp to get better headroom or SNR in the recordings.

5.2 Calibrating sound card output level for noise SPL in test chamber

Calibrate the ambient noise level in the test chamber to 85 dBSPL as measured near the headset under test. You may use a different SPL value for calibration based on your product requirements.

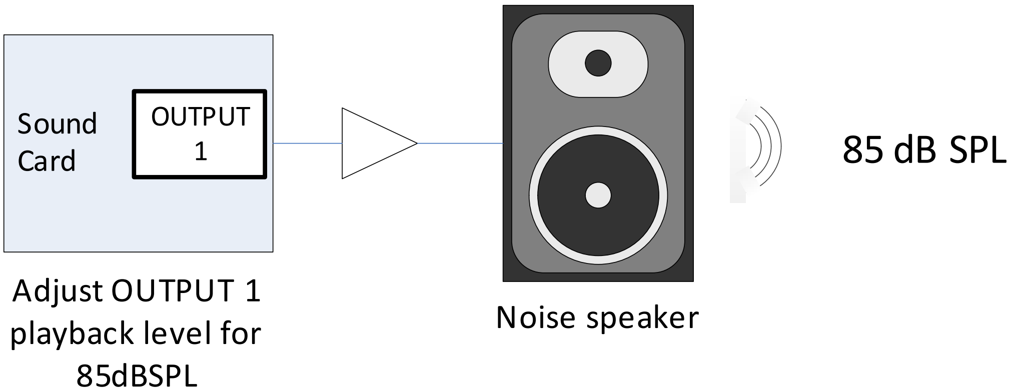

Figure 5-1 SPL calibration

1. Locate the noise file `control_characterization.wav` in the folder **<ADK>\tools\bin \sampleAncAudio**
2. Using audio software, play the noise to output channel 1 on the sound card, which should be connected to a power amplifier and speaker in the test chamber.
3. Using an SPL meter, adjust the output level of the signal until the nominal level is at 85 dBSPL (C-weighted, slow averaging).

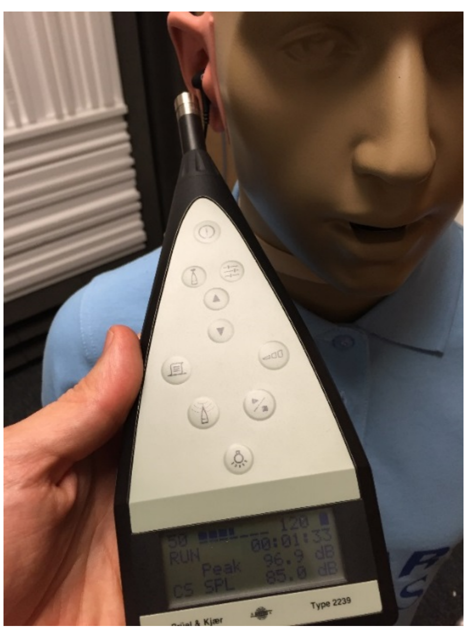

Figure 5-2 Measuring noise level with SPL meter

5.3 Calibrating ADC gains on CSR8675

This step calibrates the input gain of the ADC (microphones) to achieve the optimum signal level to the CSR8675.

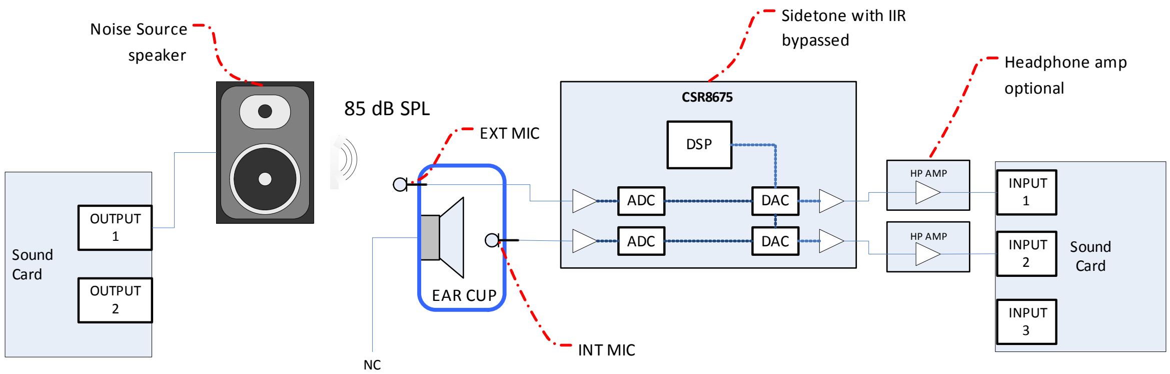

Figure 5-3 Microphone calibration

1. Locate the noise file `control_characterization.wav in the folder` **<ADK>\tools\bin\sampleAncAudio**.
2. The internal and external microphones should be connected to CSR8675. Output 1 on the sound card should drive the noise source speaker.
3. Using audio software, play the noise file to sound card output 1. The SPL level should be equivalent to the previously calibrated value of 85 dB SPL.
4. Monitor the input levels on input 1 and 2.
5. Calibrate the microphones by adjusting the ADC gain on each microphone until the peak recording level is -6dBFS (0.5 x full scale).
6. Adjust the ADC digital gain as needed to get a more accurate level.

NOTE For digital microphones, calibration is the only gain stage that can be adjusted for the ADC before the IIR filter.

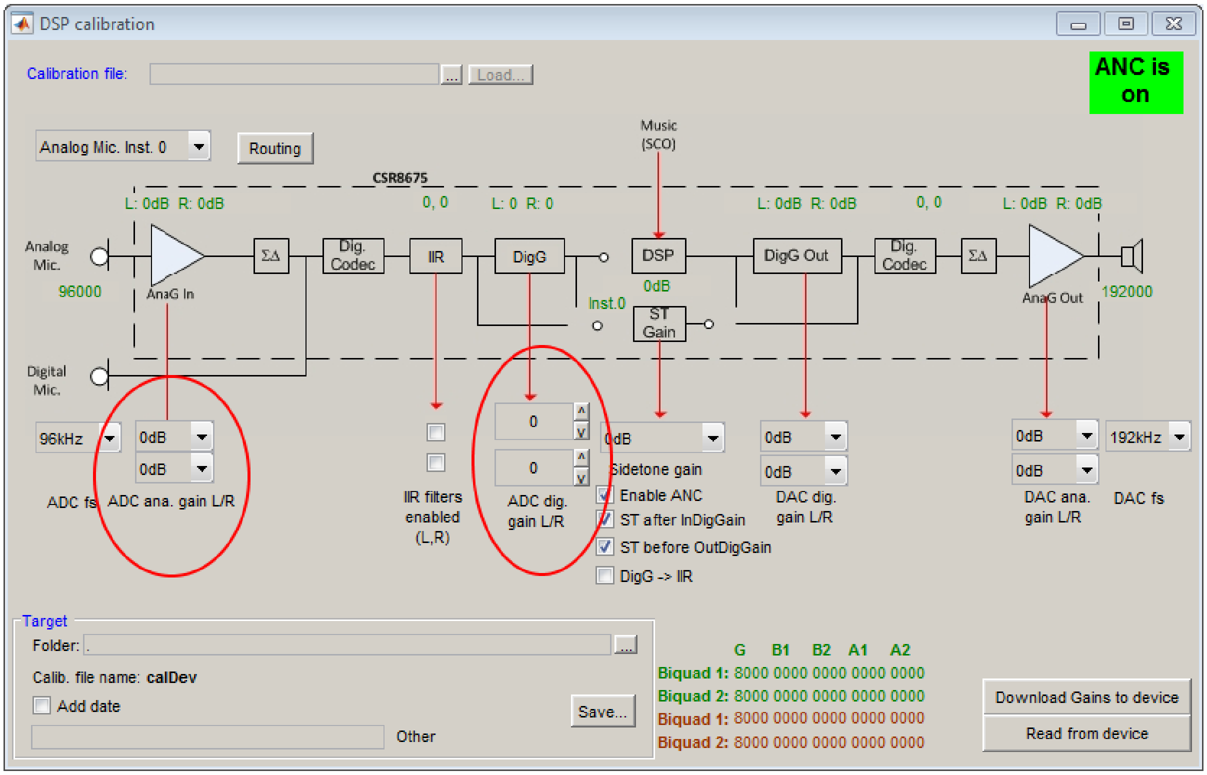

Figure 5-4 ADC gain adjustment in calibration panel

5.4 Calibrating the sound card output level for the headset

This step calibrates the output level of the sound card channel driving the headset speaker. The speaker must be able to generate an antinoise signal in equal amplitude to the ambient noise (as measured by the internal microphone).

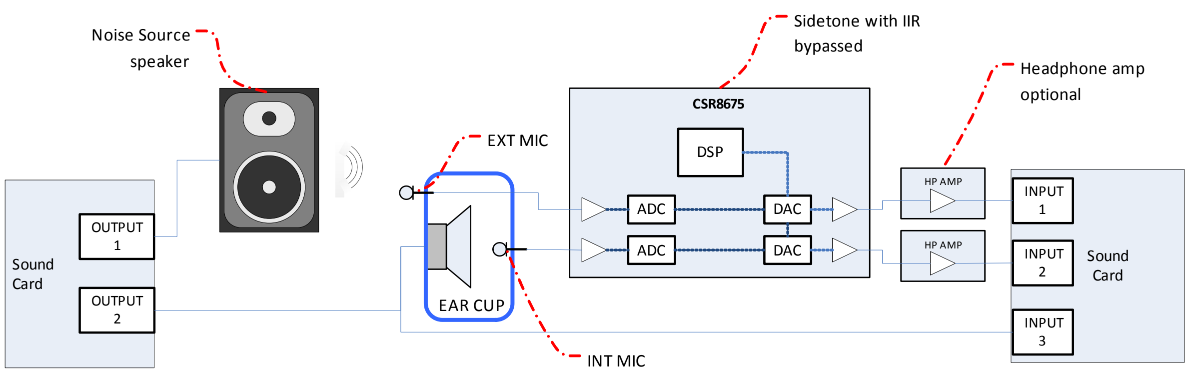

Figure 5-5 Sound card level calibration

1. Locate the noise file `plant_characterization.wav` in the folder **<ADK>\tools\bin \sampleAncAudio**.
2. Connect the system. Ensure that a sound card output channel is driving the headset speaker.

3. Using audio software, play the noise to the output channel 2 connected to the headset speaker and record on the input channel 3. Do not play to the noise source speaker on output 1.
4. Adjust the output level on the sound card output channel 2 until the input recording level on channel 2 is at -6 dBFS. Do not change the microphone gain from the previously calibrated value.

5.5 Calibrating DAC gain on CSR8675

To calibrate the DAC output level, using the same setup from step 4 above and adjust the DAC gain until the internal mic recording level is the same as the speaker reference signal. With these signals equal, the calculated plant has a gain close to 1, which is ideal. In the previous step, the mic level on input 2 was adjusted to -6 dBFS, after the DAC is adjusted this level is lower.

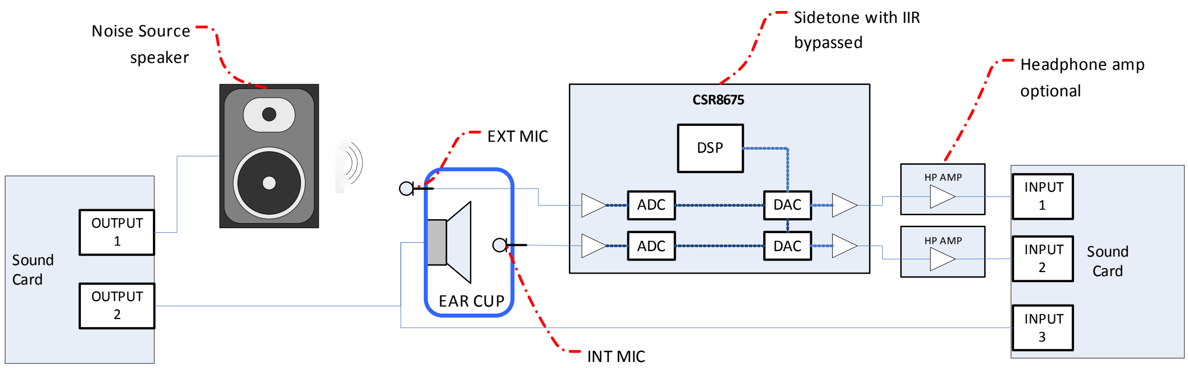

Figure 5-6 DAC gain calibration

1. Locate the noise file `plant_characterization.wav` in the folder **<ADK>\tools\bin \sampleAncAudio**.
2. Connect the system. Sound card output channel 2 should be driving the headset speaker.

3. Using audio software, play the noise to the output channel 2 connected to the headset speaker and record on the input channels 2 and 3.
4. Adjust the DAC gain until the recording level on channel 2 and channel 3 are approximately equal. Copy the DAC gain to both left and right channels. Do not adjust the ADC gain from its previously calibrated value.

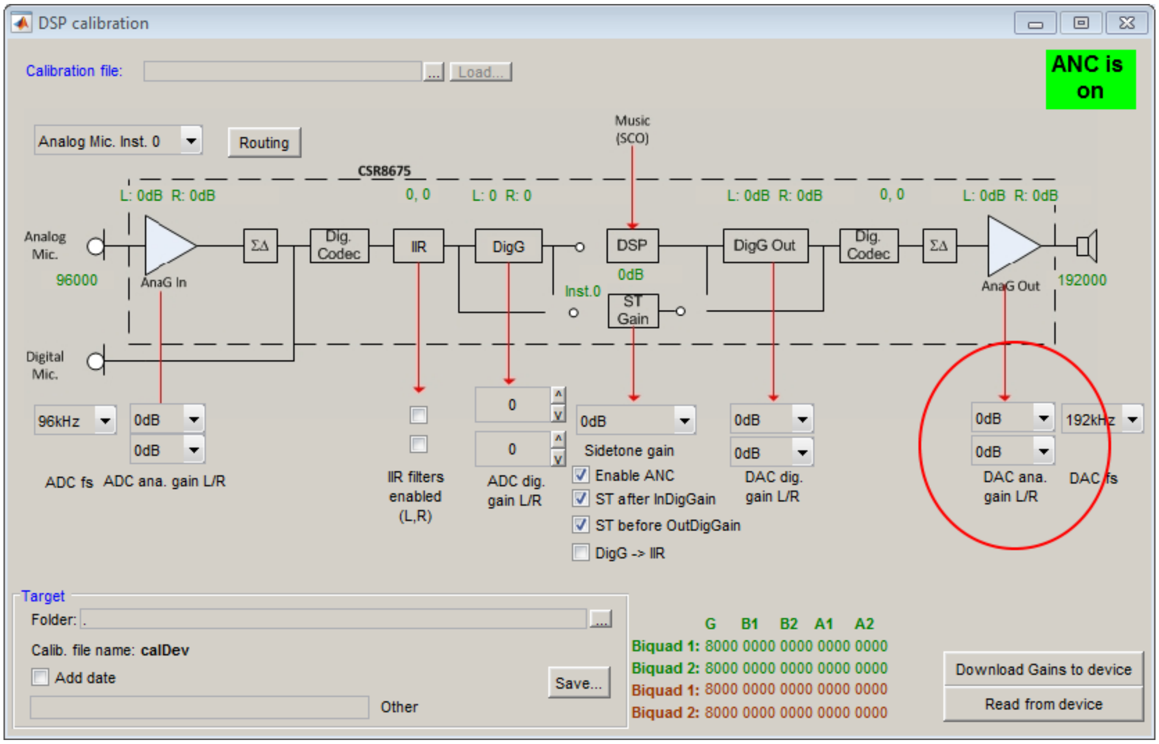

Figure 5-7 DAC gain adjustment in calibration panel

5.6 Finalizing calibration

To finalize calibration, use the **Save** button on the calibration panel and save to a calibration file that contains gain values that are later downloaded to the CSR8675. The saved calibration file is required later to complete the design.

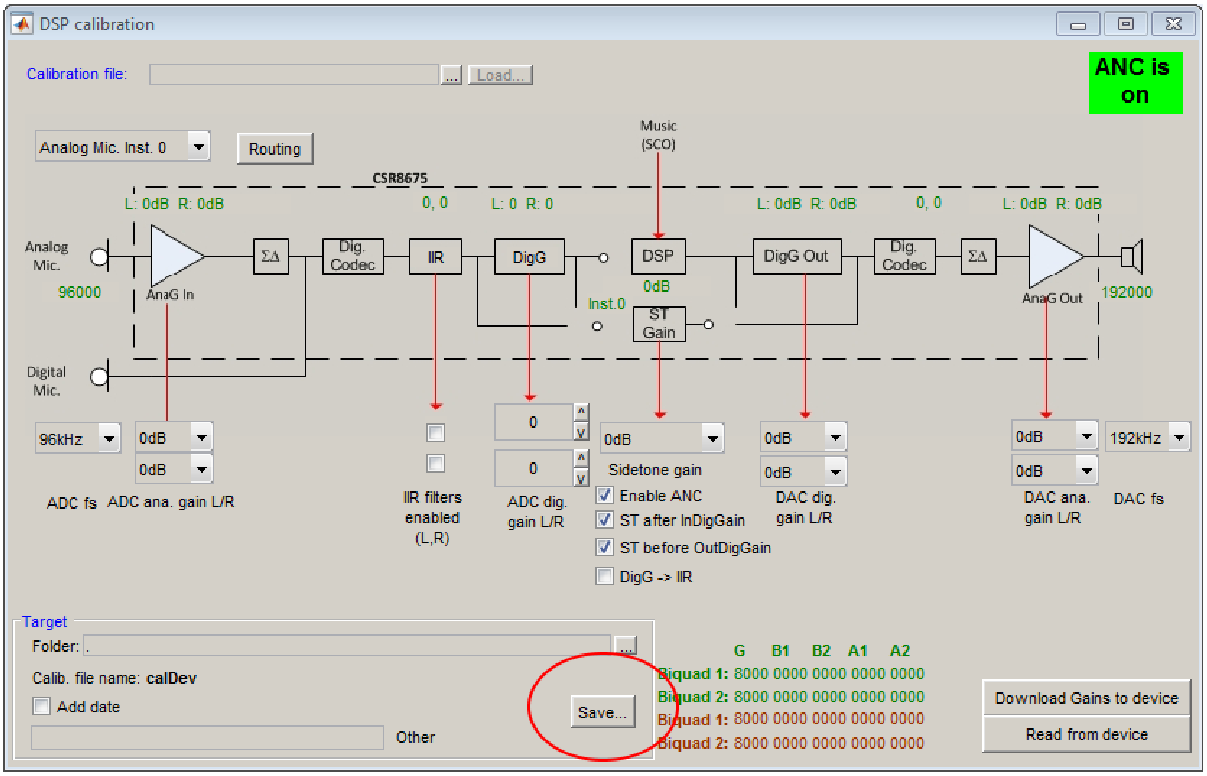

Figure 5-8 Finalizing calibration

6 ANC filter design examples

6.1 Short circuit example design

The short circuit example design checks the development board and ANC algorithm for proper operation. The short circuit design does not use the acoustical part of the system, so the results do not depend on the type of headset used, and the measurements can be easily replicated.

In this design, the headset is removed and recordings are made by using a short circuit in place of the acoustic path between the speaker and internal microphone. This means the plant is ideal, only containing the transfer function of the development board.

The short circuit design can check the maximum theoretical attenuation which is possible with a particular development board and source recordings.

6.1.1 Making recordings

Make a design with the provided recordings. After obtaining a good result, make your own recordings and create a new design and verify that both short circuit designs match.

Provided example recordings are available in **<ADK>\tools\bin\sampleAncAudio\ ShortCircuit.**

- Speaker: short_circuit_speaker_for_plant.wav
- Int. Mic: short_circuit_int_mic_for_plant.wav
- User recordings: To make the plant recordings for the short circuit, make the usual connections but with the acoustical path removed.

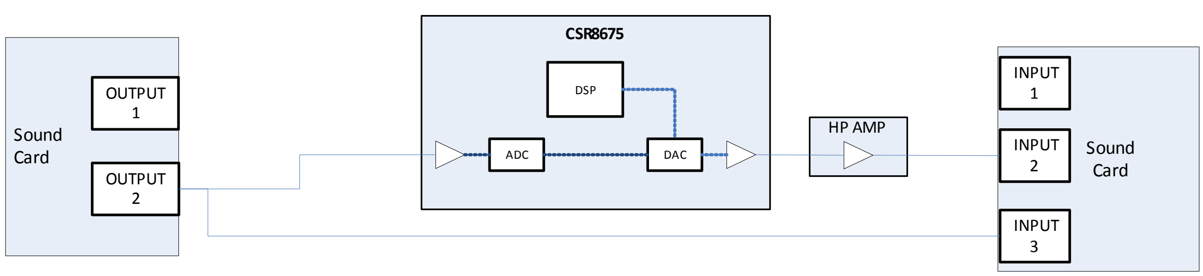

Figure 6-1 Short circuit recording setup with acoustic path removed

6.1.2 Create a new project

1. Select the New project option from the tool's menu. Select the folder where the project will be located and a name for the project.

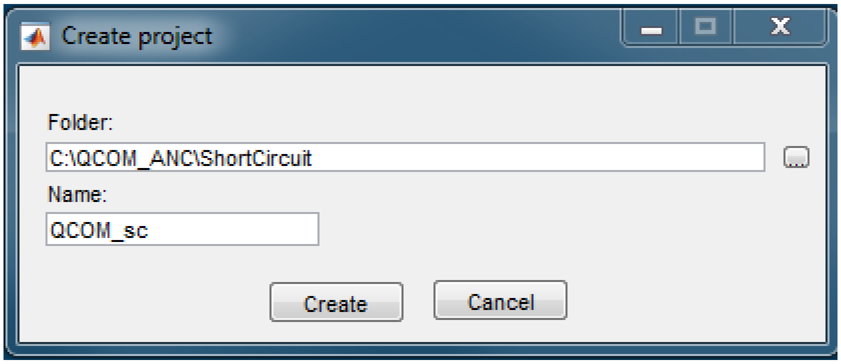

Figure 6-2 Create project

2. Choose the folder where the recordings are located and the folder where the results are expected to be stored (project folder is the default).

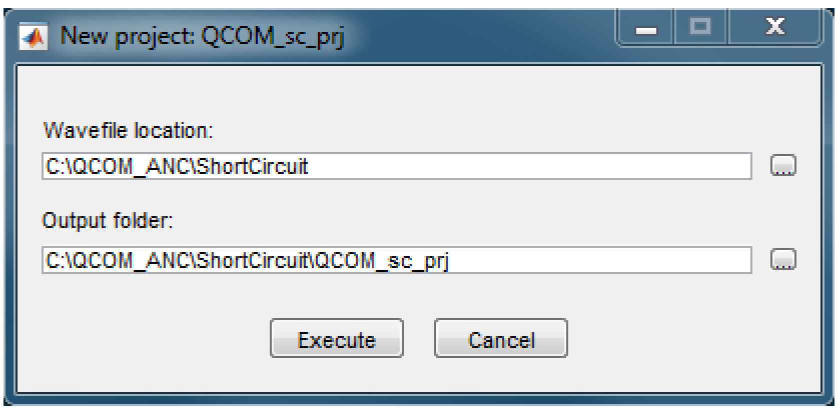

Figure 6-3 Project folder locations

6.1.3 Plant design

The plant is the impulse response of the electrical path inside the ear cup. To start, navigate to the plant panel by pressing the **plant** button.

Click the **Plant Files** button in the Plant File Manager section to choose the plant files.

Choose the speaker and internal mic. files provided with the tool:

- Speaker: `short_circuit_speaker_for_plant.wav`
- Int. mi.: `short_circuit_int_mic_for_plant.wav`

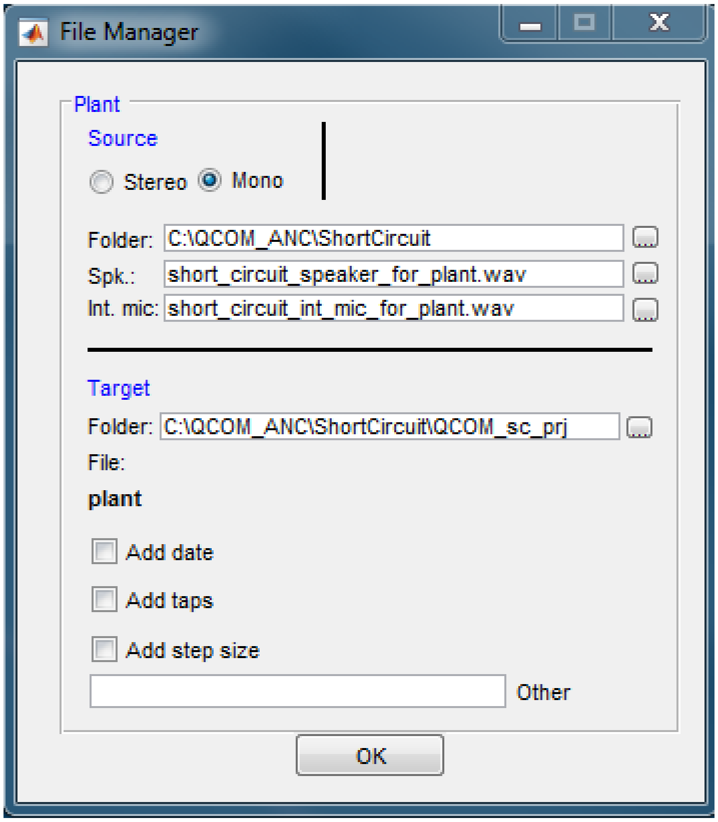

Figure 6-4 Plant files for short circuit design

Using the default values for the parameters, start calculating the plant. Click **Go**.

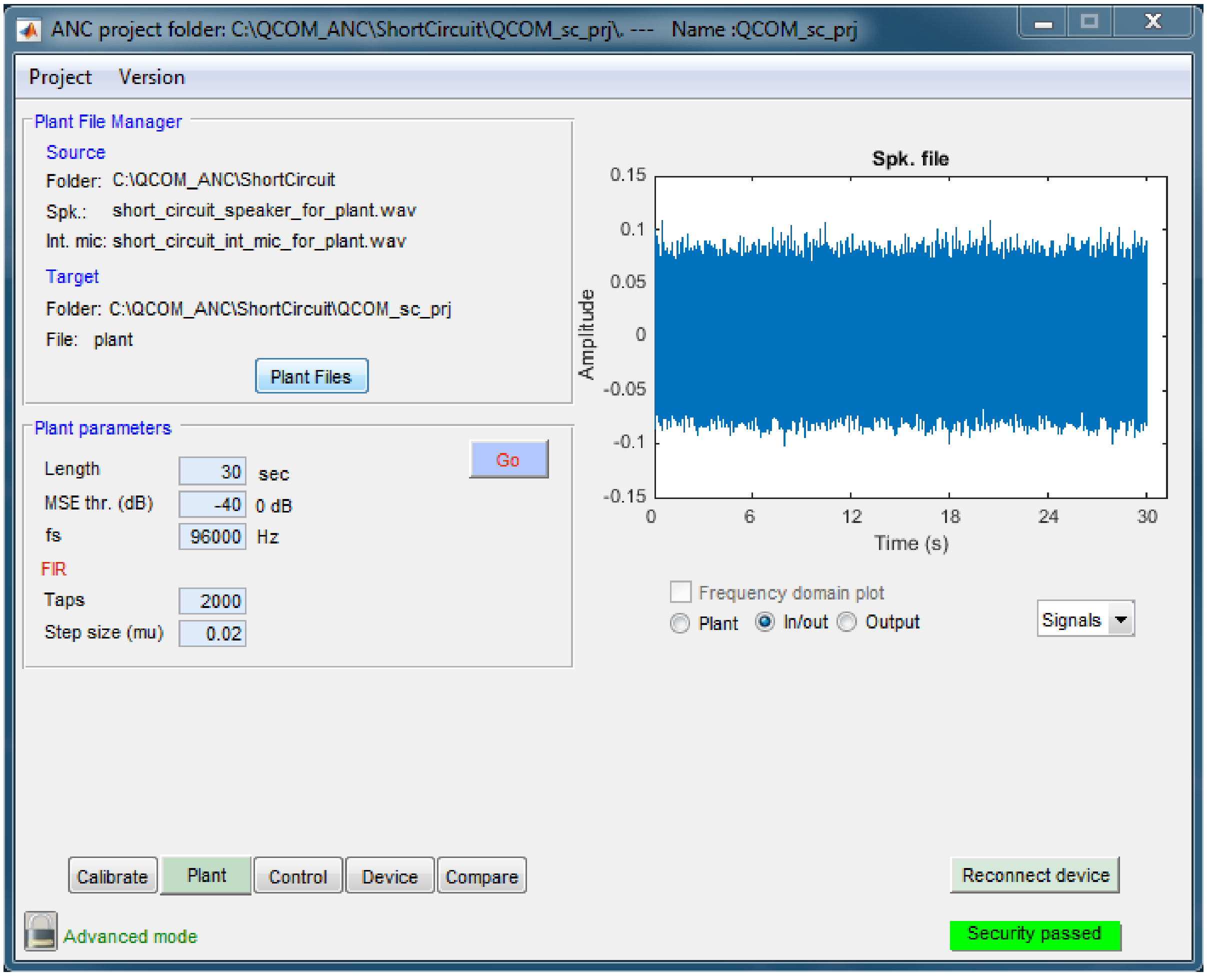

Figure 6-5 Calculating the plant for the short circuit design

The Progress bar displays the Minimum Square Error. The lower it is, the better the solution.

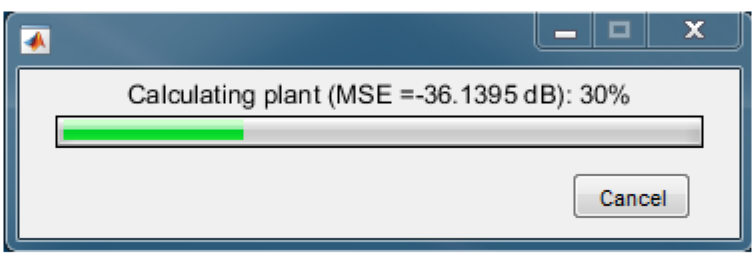

Figure 6-6 Plant calculation in progress

The plant has been calculated. The impulse response displays the total delay of the acoustic and electrical path. Because this example does not have an acoustic path, the plant shows the electrical delay only.

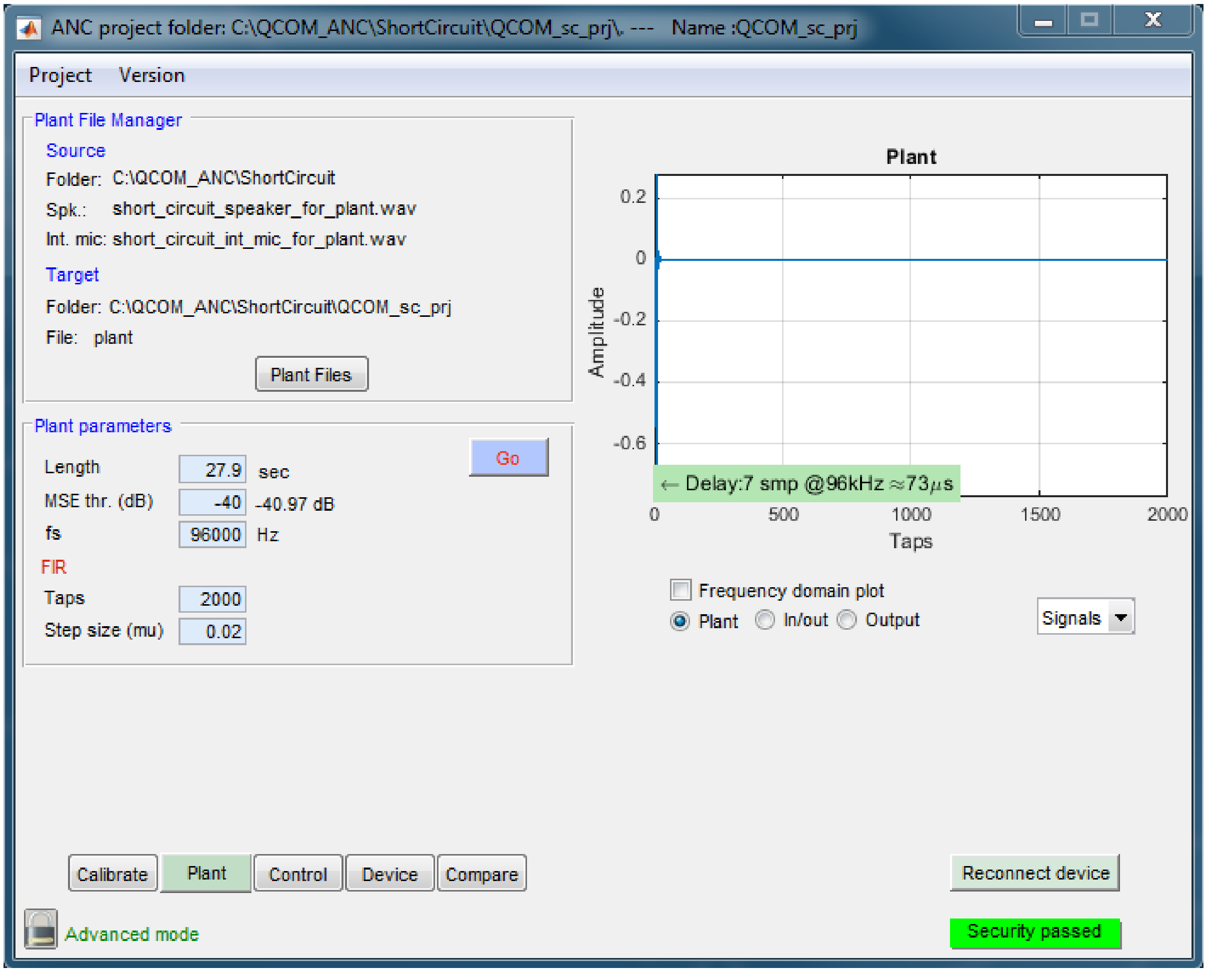

Figure 6-7 Calculated short circuit plant impulse response

In this example, the delay introduced by the development board (ADC + DAC) is approximately 73 µs.

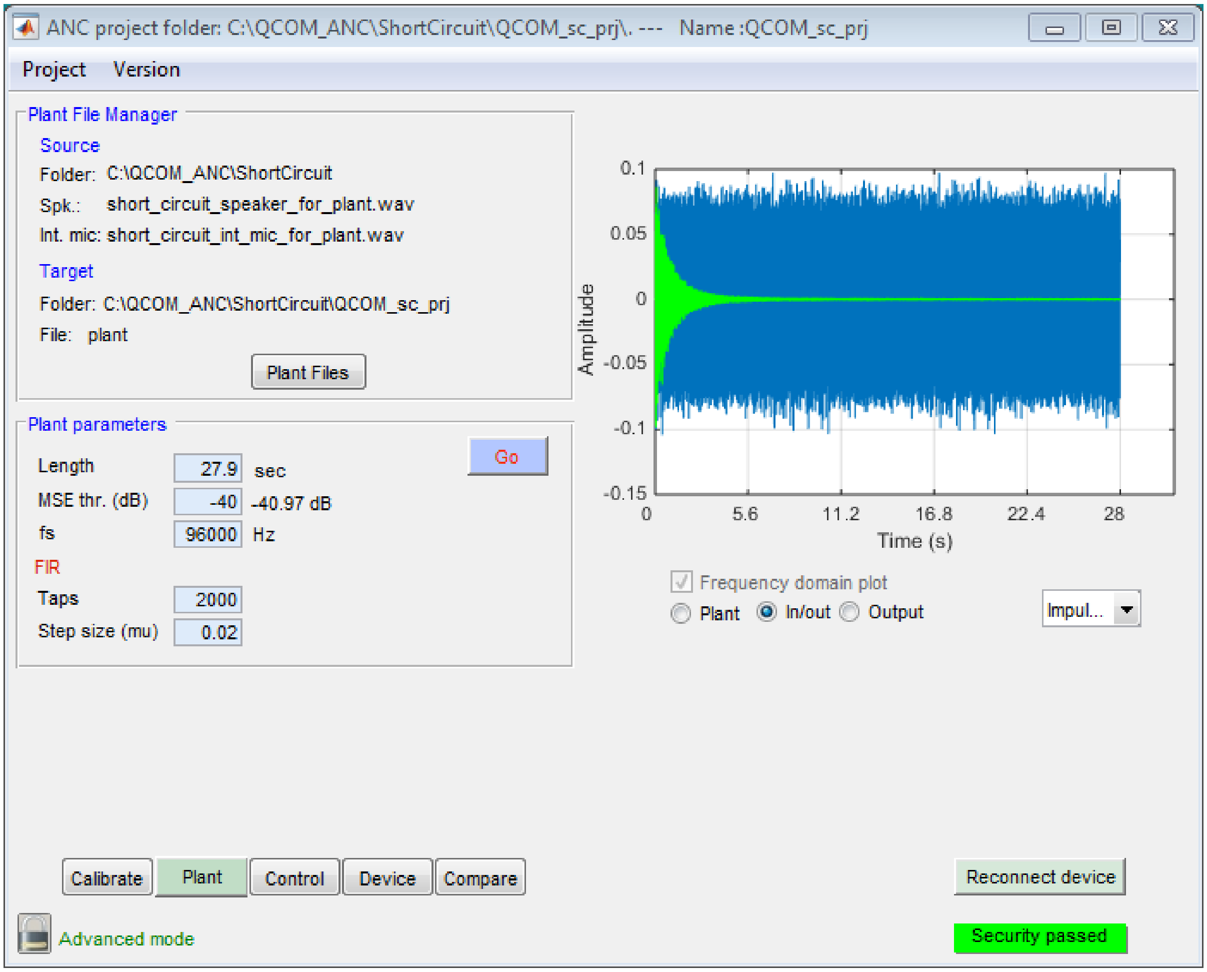

Figure 6-8 Short circuit plant convergence

The green plot shows plant convergence. The smaller the value as the plot goes to the right indicates better convergence. The final convergence value is shown next to the **MSE thr** box. A small MSE value of -40 dB indicates the algorithm is working well with the given recordings.

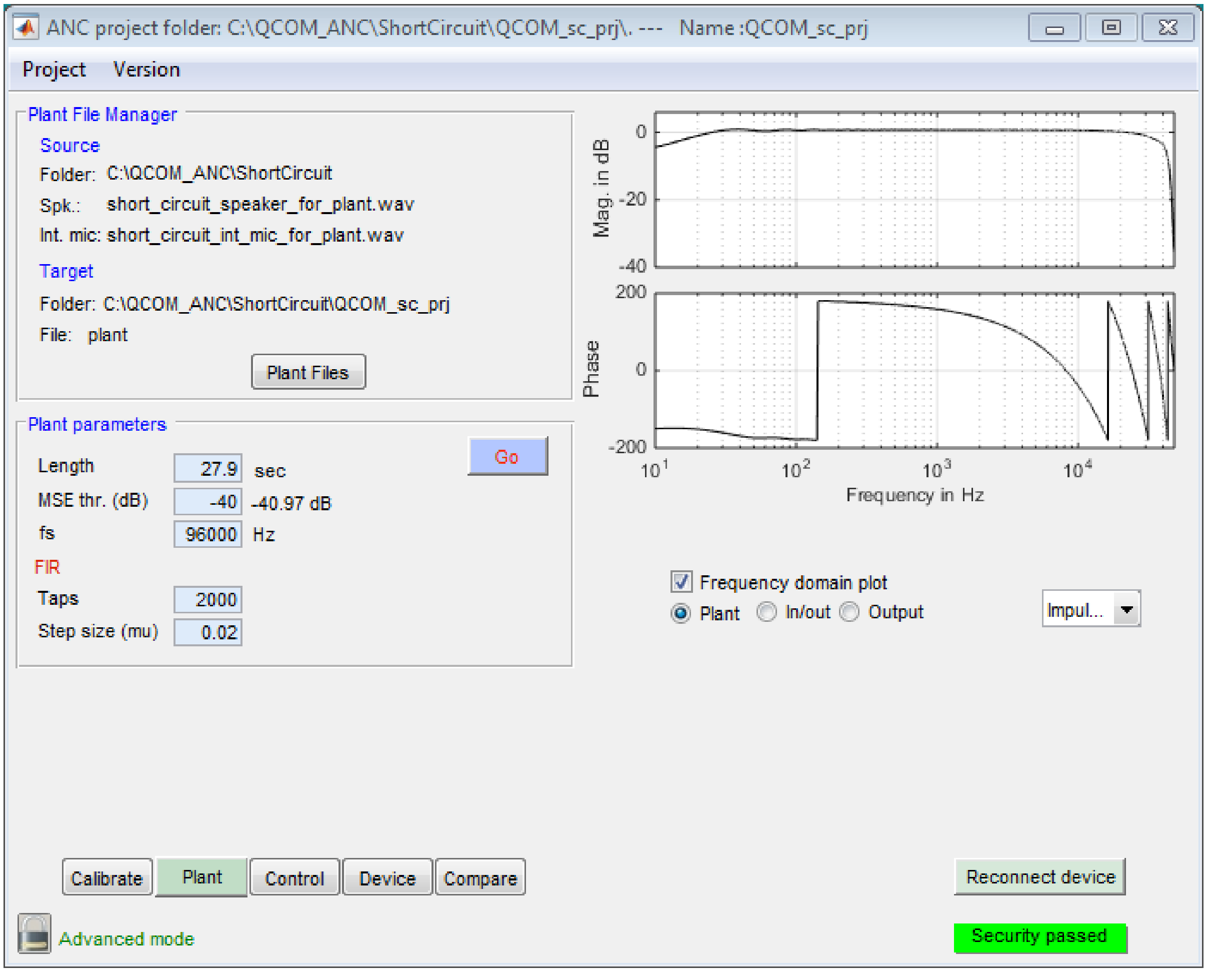

Figure 6-9 Short circuit plant frequency and phase response

6.1.4 Controller design

After obtaining the plant impulse response, calculate the controller:

Navigate to the control panel by pressing the **control** button and select the input wave file. In the case of the short circuit design, the signal at the input is the same as the one at the output, so choose `control_characterization_mono.wav.` for both internal microphone and external microphone: There is no need to make any recordings.

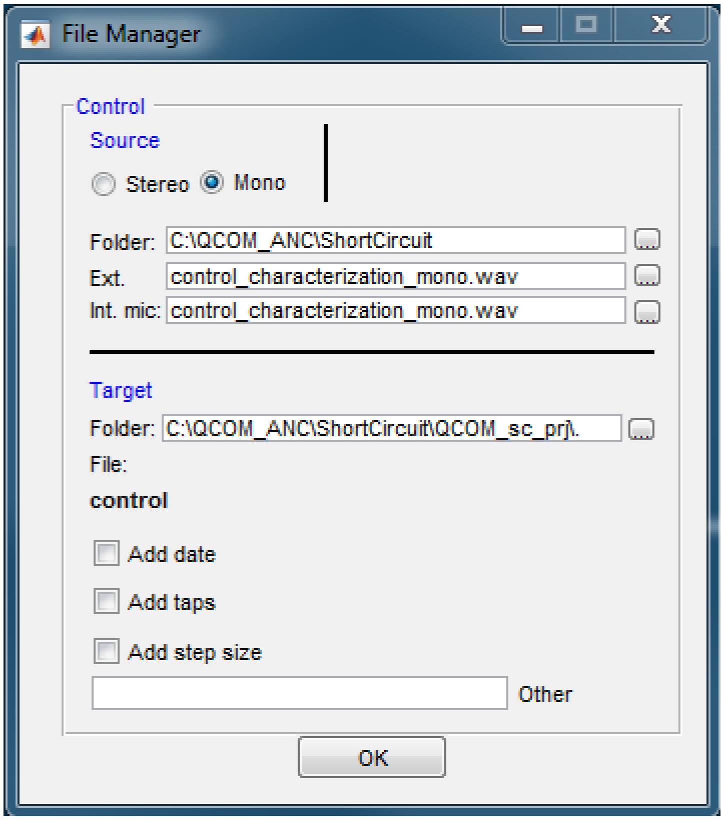

Figure 6-10 Control recordings for short circuit design

The control panel calculates both FIR and IIR filters, but only the parameters for the FIR can be modified. After the FIR is calculated, open a new panel needs to tune the IIR parameters.

When selecting how many taps to use from the plant, select when the coefficients have reached 0. From the plant panel, plot the Impulse Response of the plant and see how many taps it takes for it to reach 0. In this example, the impulse response is reaching zero at tap number 30. Any number of taps greater than 30 is acceptable.

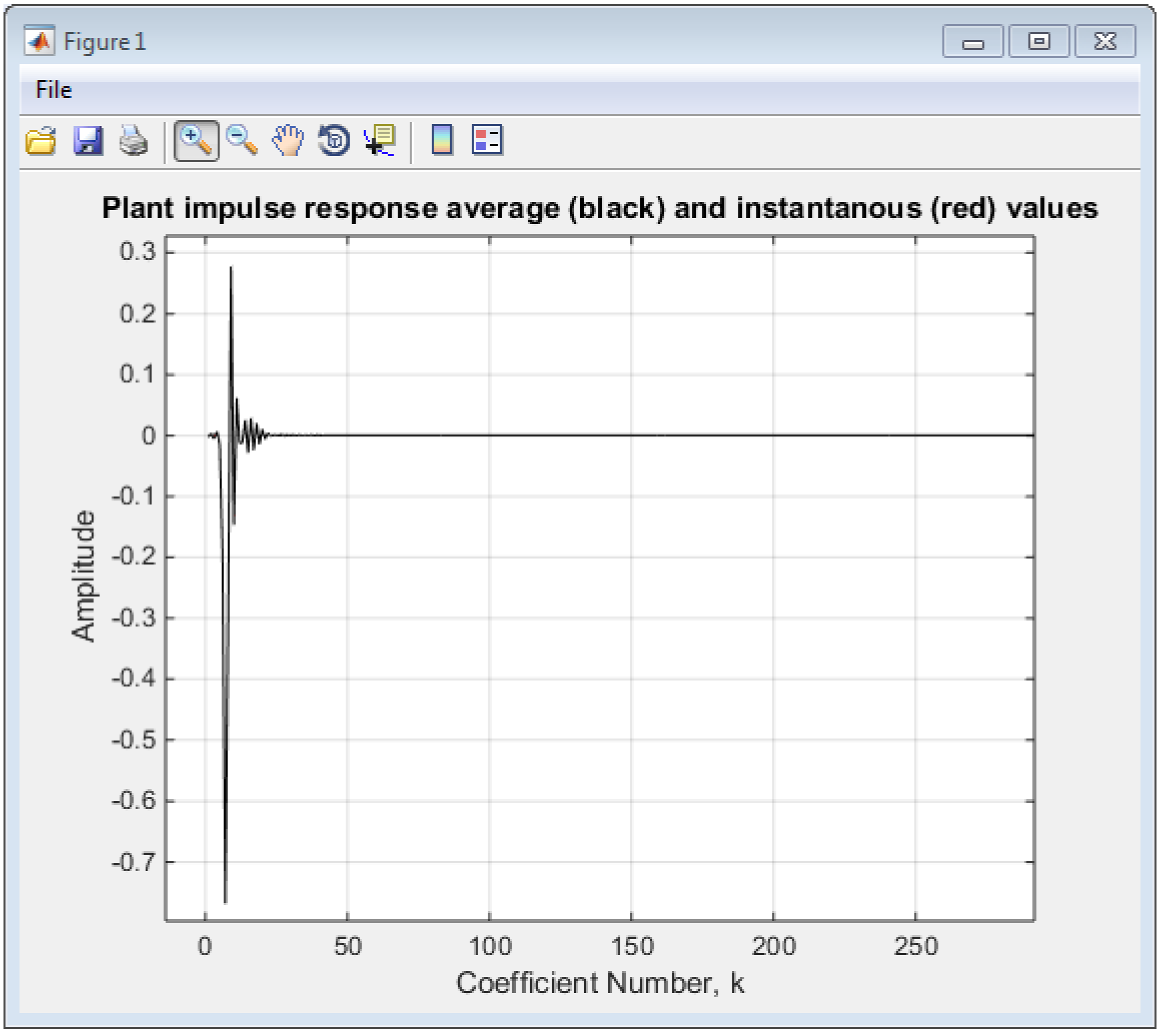

Figure 6-11 Plant impulse response detail for taps for plant selection

Set the control parameters as click **Go**.

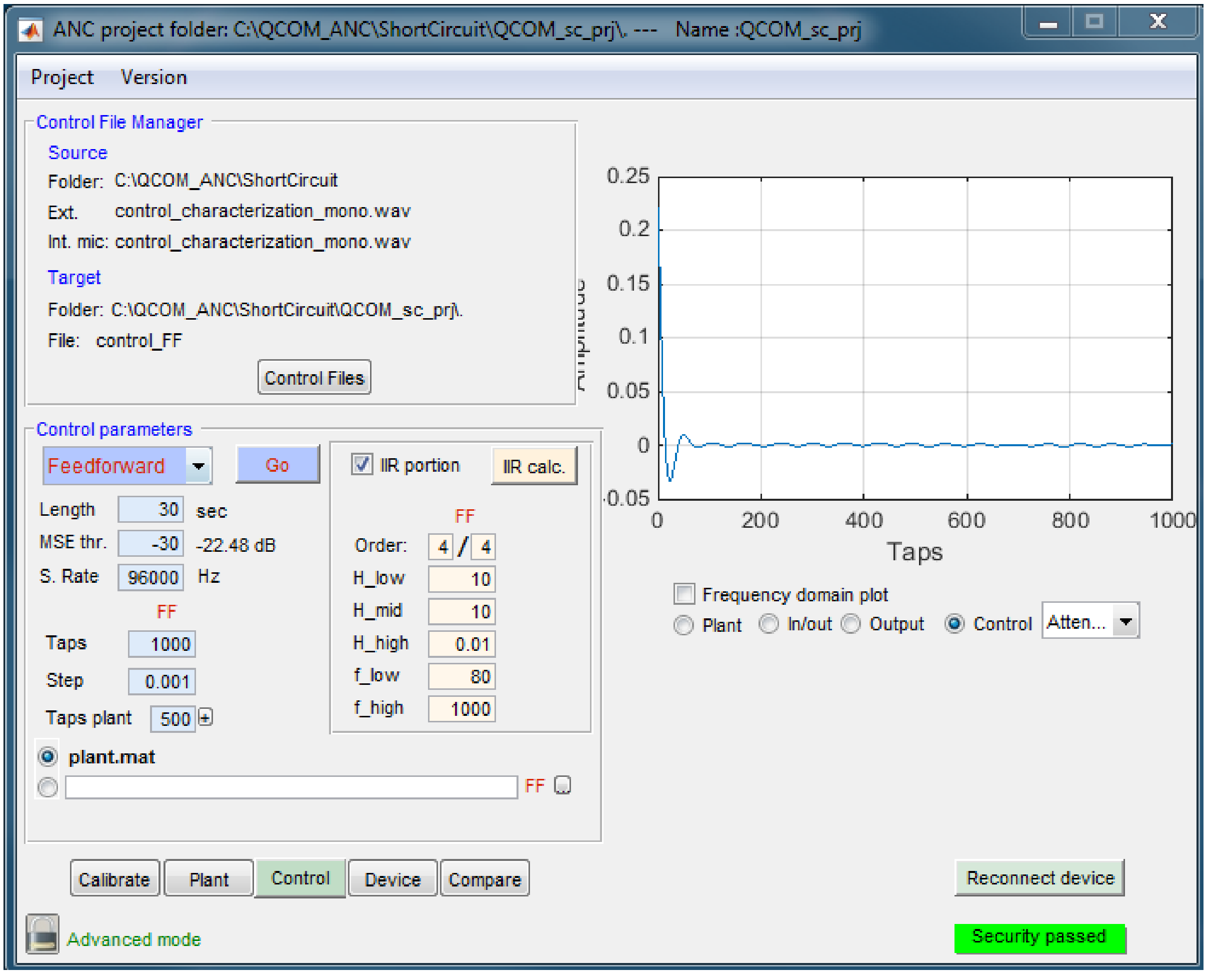

Figure 6-12 Parameters for controller in short circuit design

While the control is being calculated, a progress bar with the MSE and a plot of the current attenuation display.

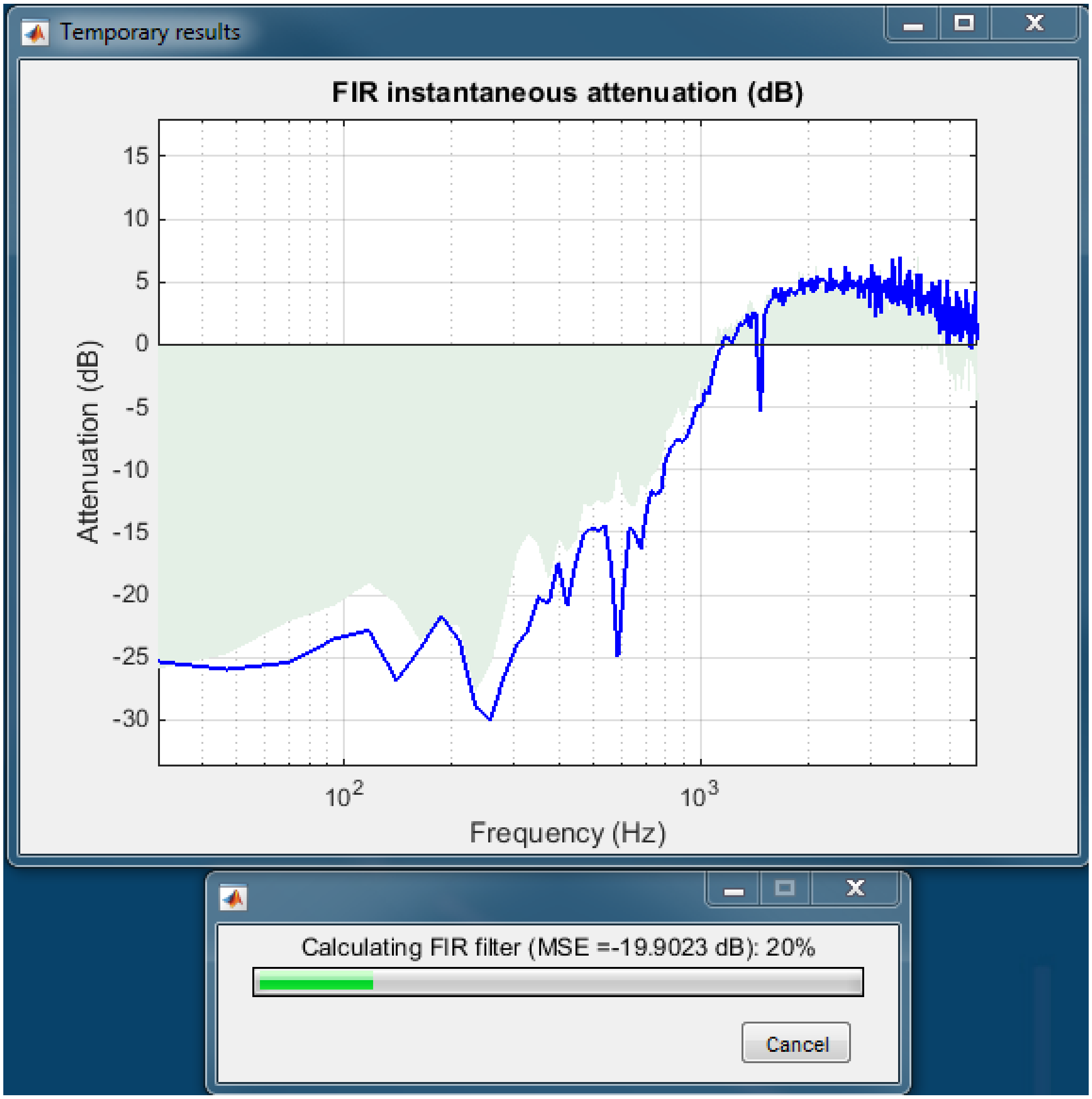

Figure 6-13 Controller calculation in progress

The in/out option in the plot panel shows the quality of the convergence of the controller. The blue signal is input and the green signal is output.

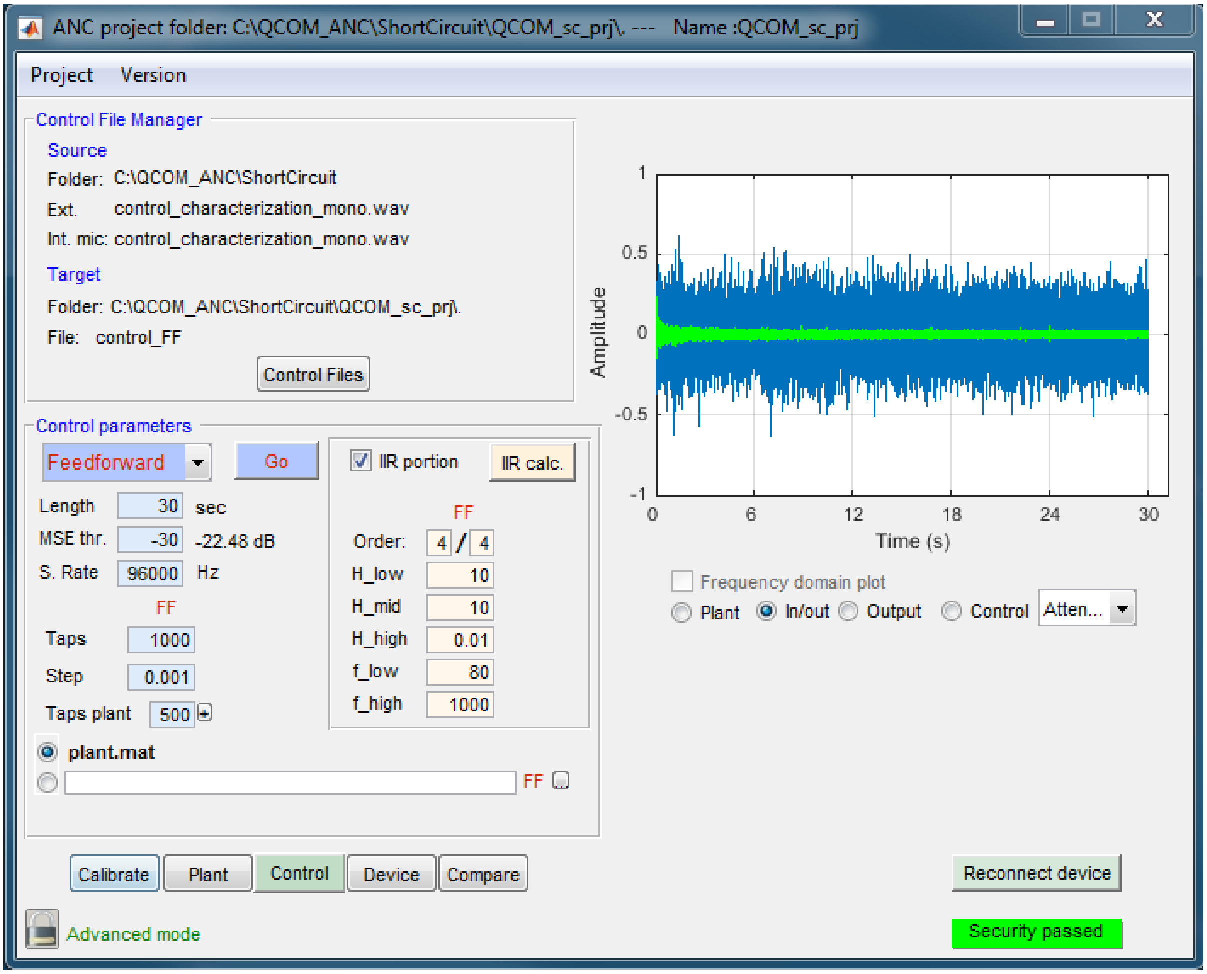

Figure 6-14 Short circuit controller convergence

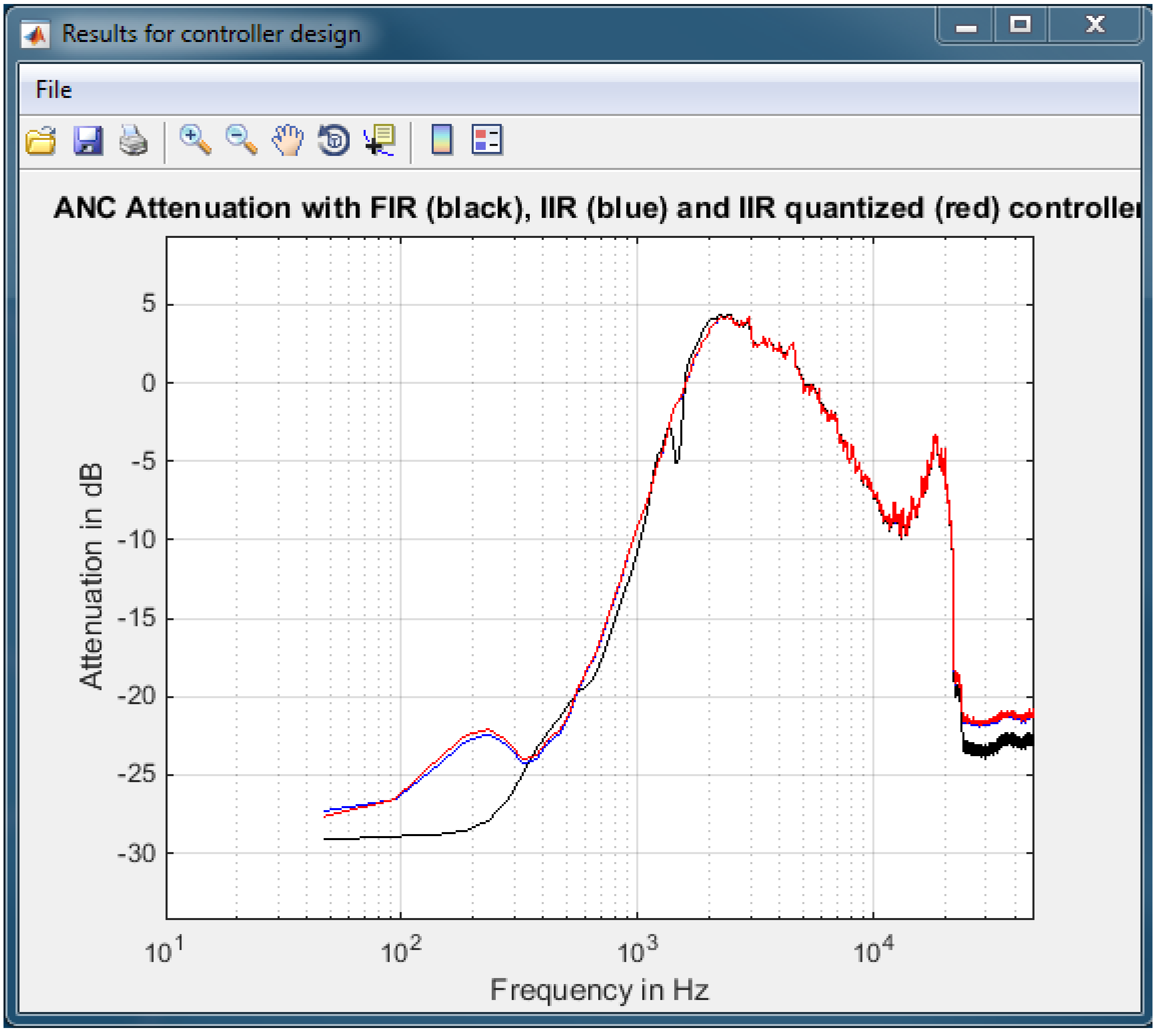

Figure 6-15 Theoretical attenuation of short circuit design

Because this design is done with a short circuit recording and not an actual headset, the results are strictly theoretical. However, the design can show how much attenuation is possible under ideal conditions.

6.2 Earbud example design

The earbud design example uses a pair of Qualcomm feed-forward earbuds.

This section describes a design using a Example recordings are provided. These are feed-forward earbuds developed for demonstration purposes.

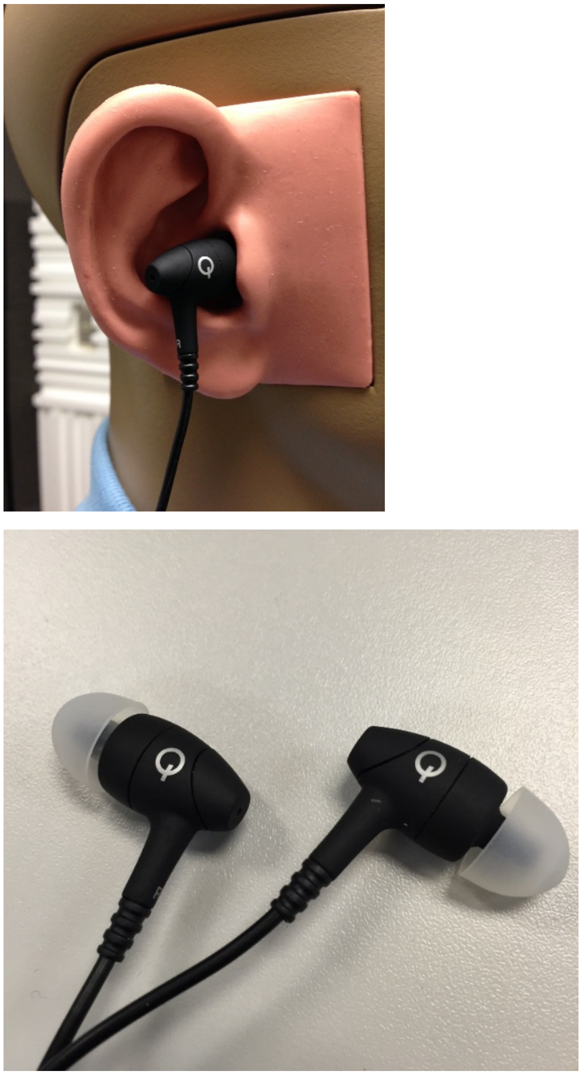

Figure 6-16 Qualcomm example earbud

6.2.1 Making recordings

Make a design with the provided example recordings. After obtaining a good result, make you own recordings and create a new design, and verify that both earbud designs match.

Provided example recordings are available in the folder in the folder **<ADK>\tools\bin \sampleAncAudio\ QCOM_earbuds.**

6.2.2 Create a new project

Select the New project option from the tool's menu. Select the folder where the project will be located and a name for the project.

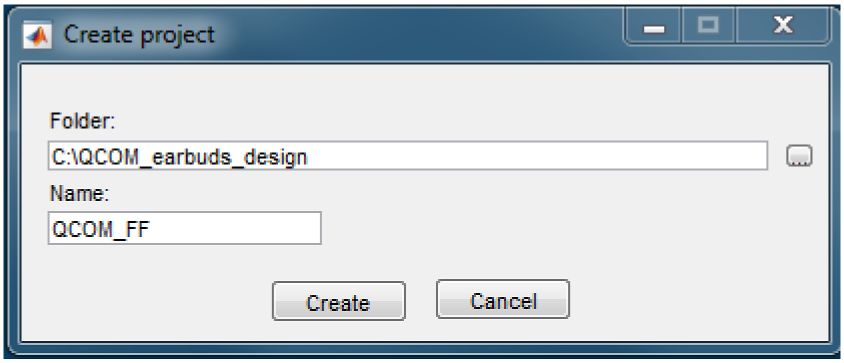

Figure 6-17 Create project panel

Choose a wave file location and target folder.

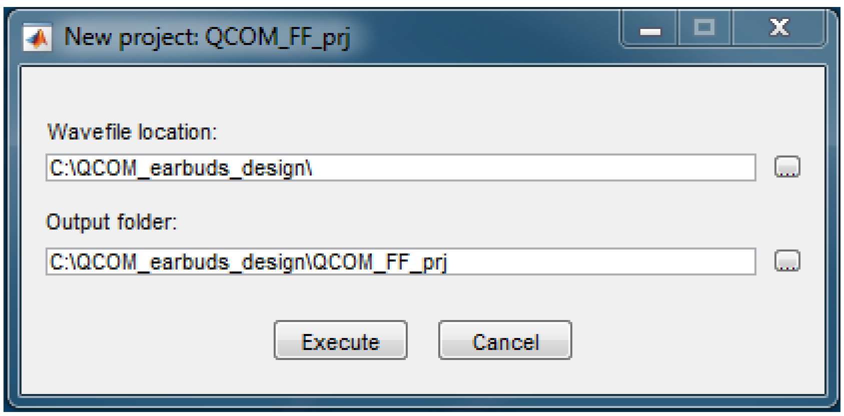

Figure 6-18 Project folder locations

6.2.3 Plant design

The plant is the impulse response of the electrical path inside the ear cup. To start, navigate to the plant panel by pressing the **plant** button.

Click **Plant Files** in the Plant File Manager section to select the plant files.

For a stereo wave file, identify the signals in the Left and Right channels, either the internal microphone or the speaker.

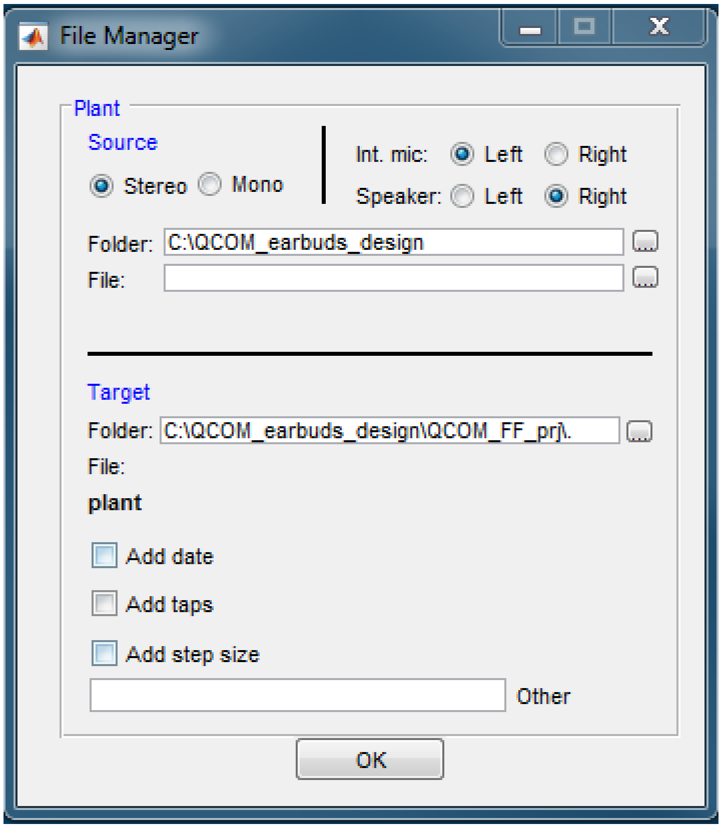

Figure 6-19 Choosing stereo files

For mono wave files, assign the correct signal to each file.

The target filename has the date, number of taps, step size, and other information appended to it.

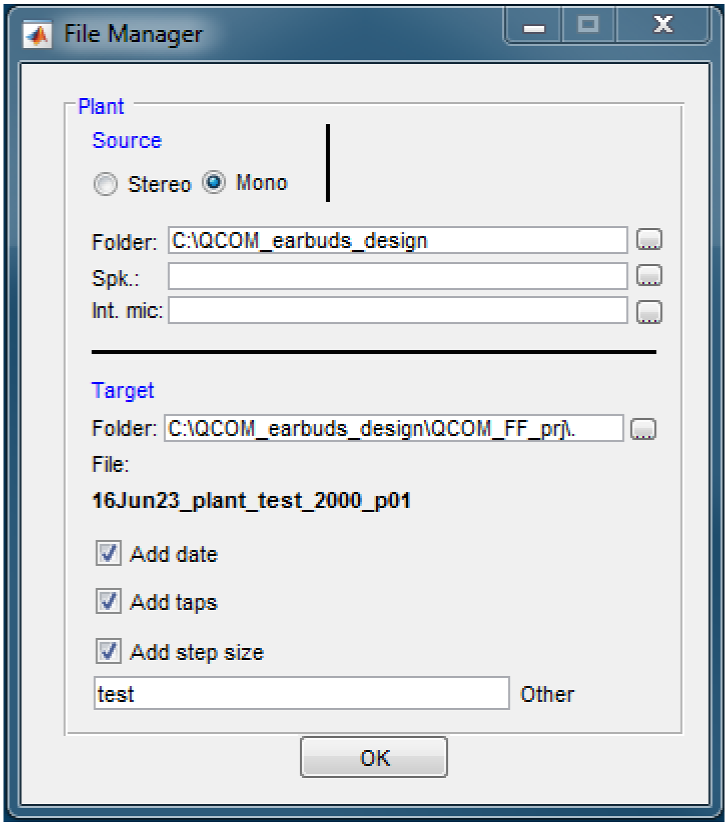

Figure 6-20 Choosing mono files

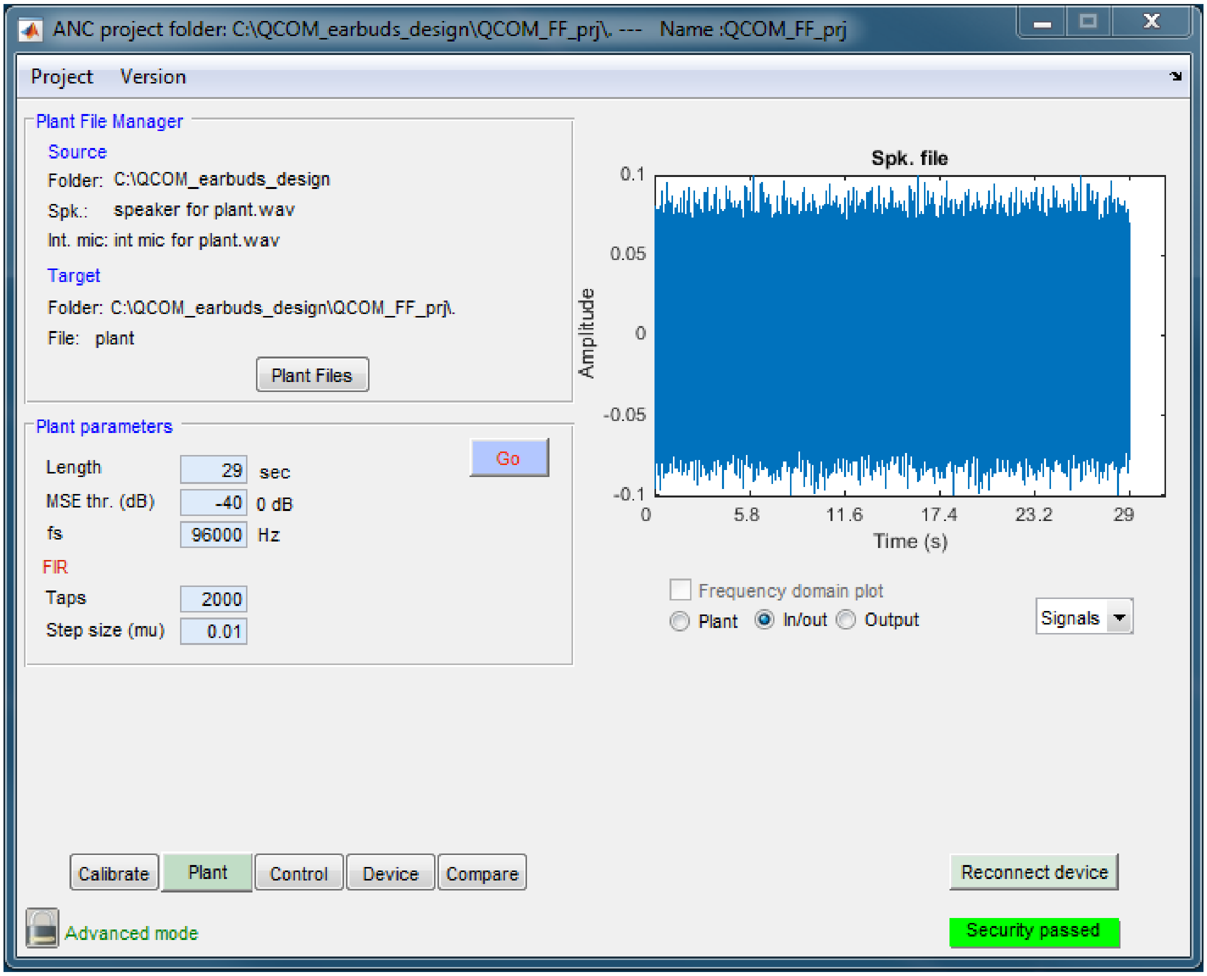

Figure 6-21 Calculating the earbud plant

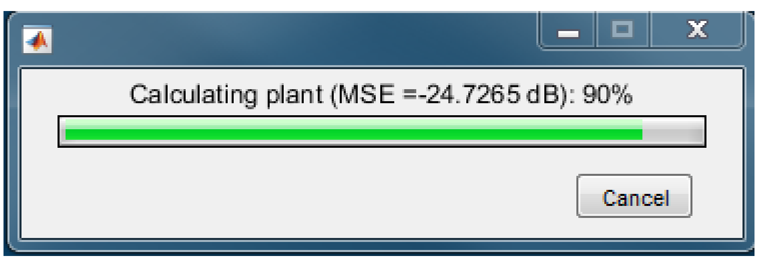

Figure 6-22 Earbud plant calculation in progress

After the plant has been calculated, the impulse response displays the total delay of the acoustical and electrical path.

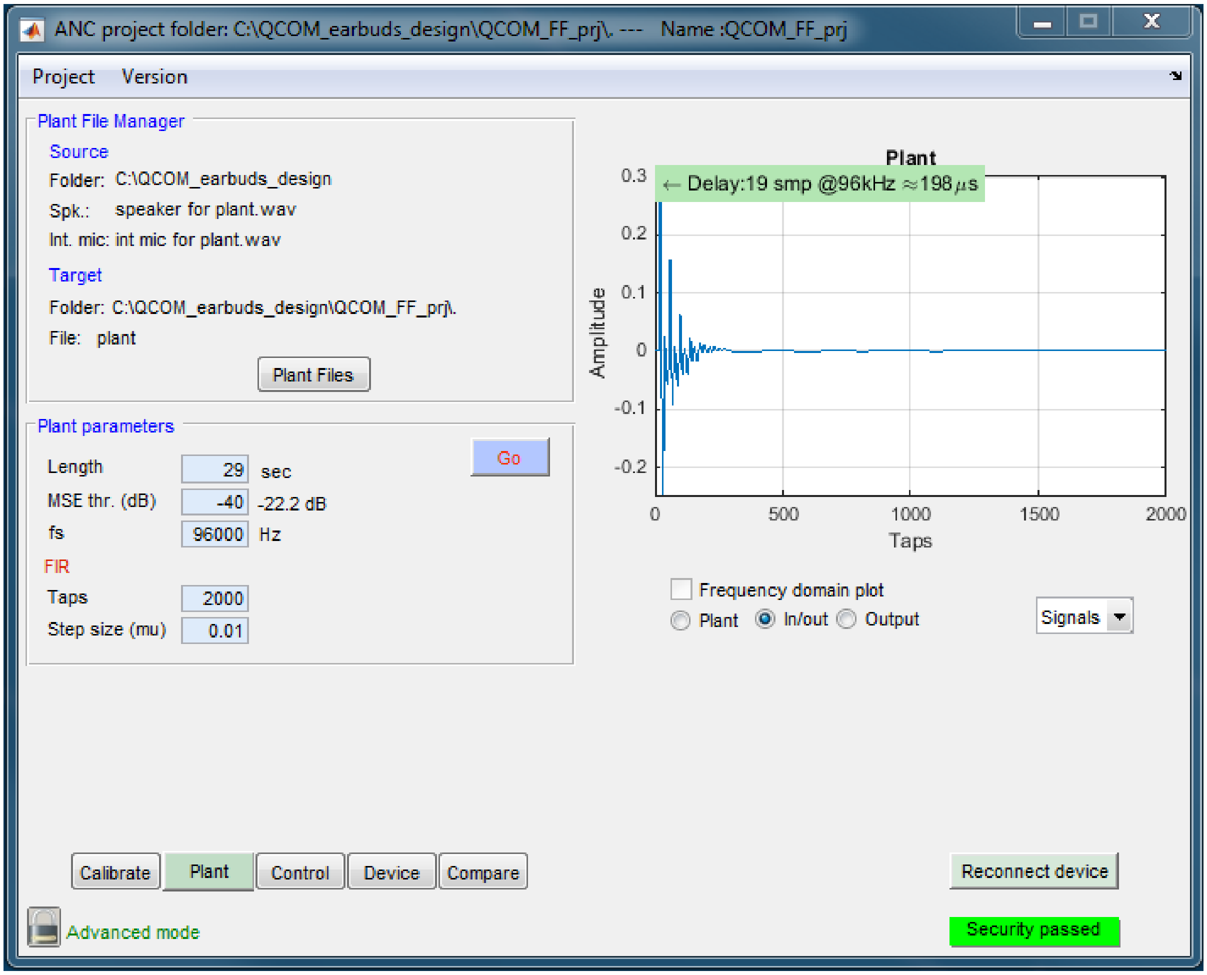

Figure 6-23 Earbud plant results

This window can display the frequency response of the plant.

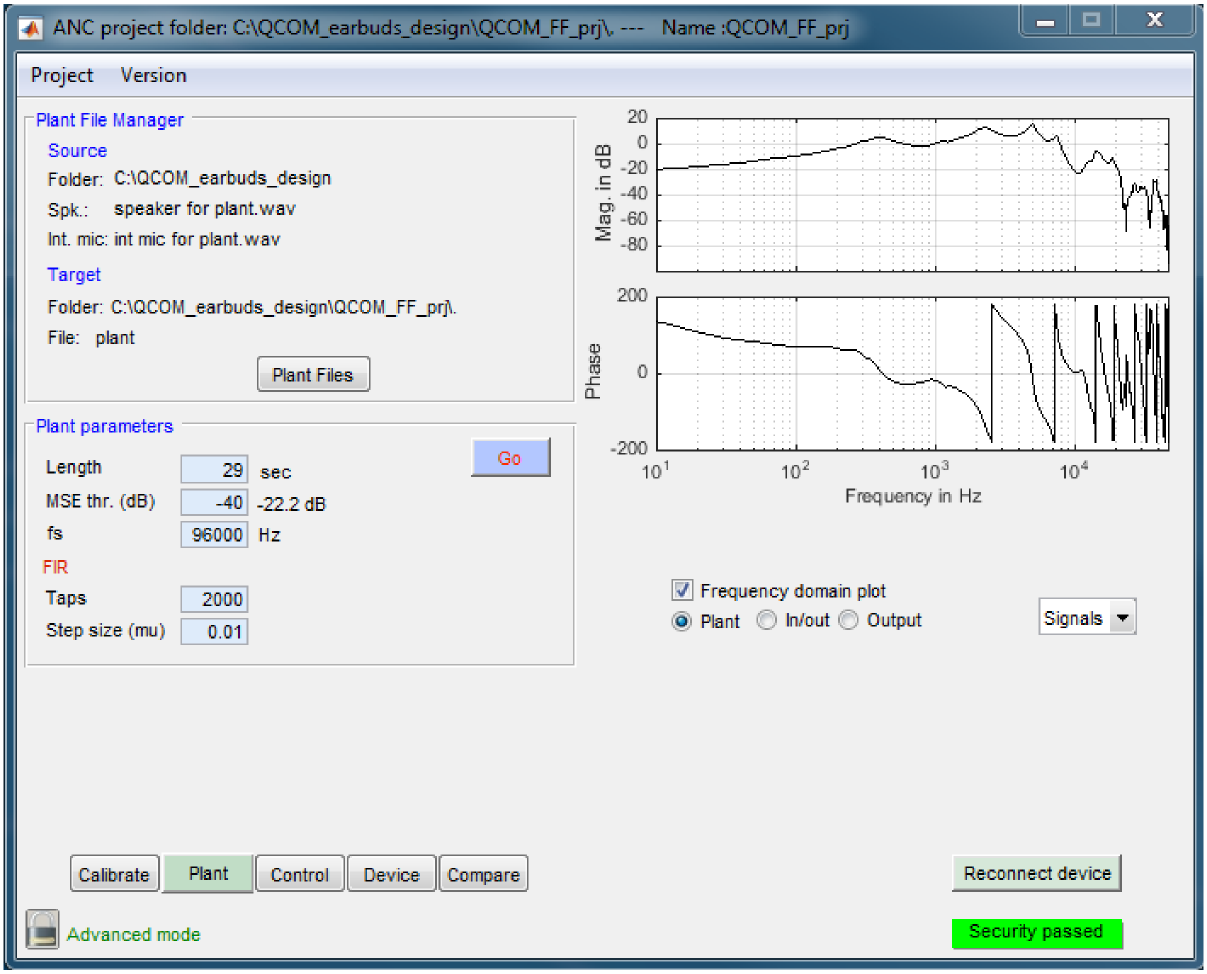

Figure 6-24 Earbud example plant frequency response

6.2.4 Controller design

After obtaining the plant impulse response, calculate the controller:

Navigate to the control panel by pressing the **control** button and choose the input wave file. If the wave file is not on the same folder, the name is highlighted in red.

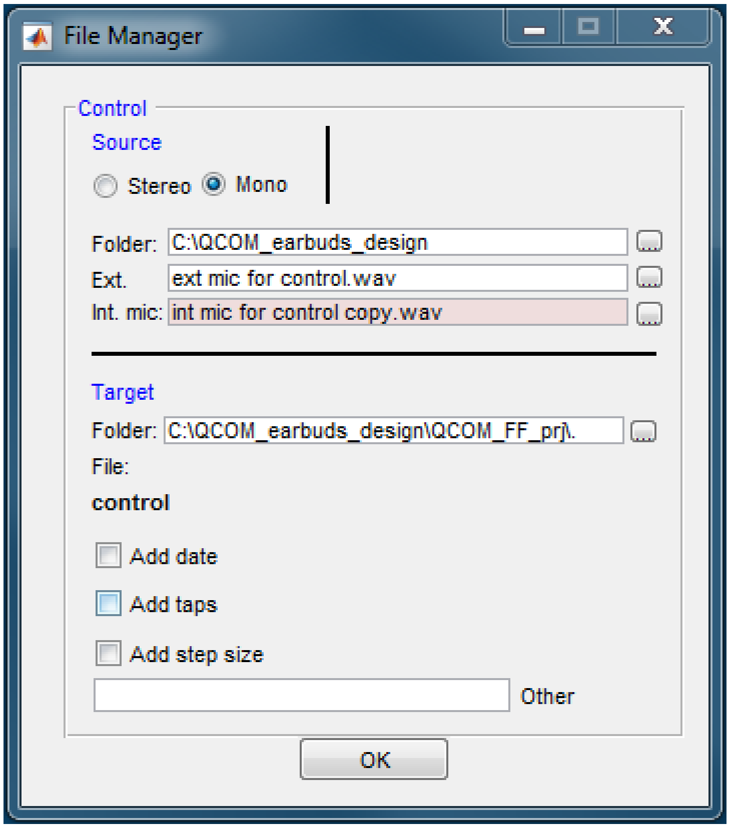

Figure 6-25 Earbud example controller files with wrong int. mic. file

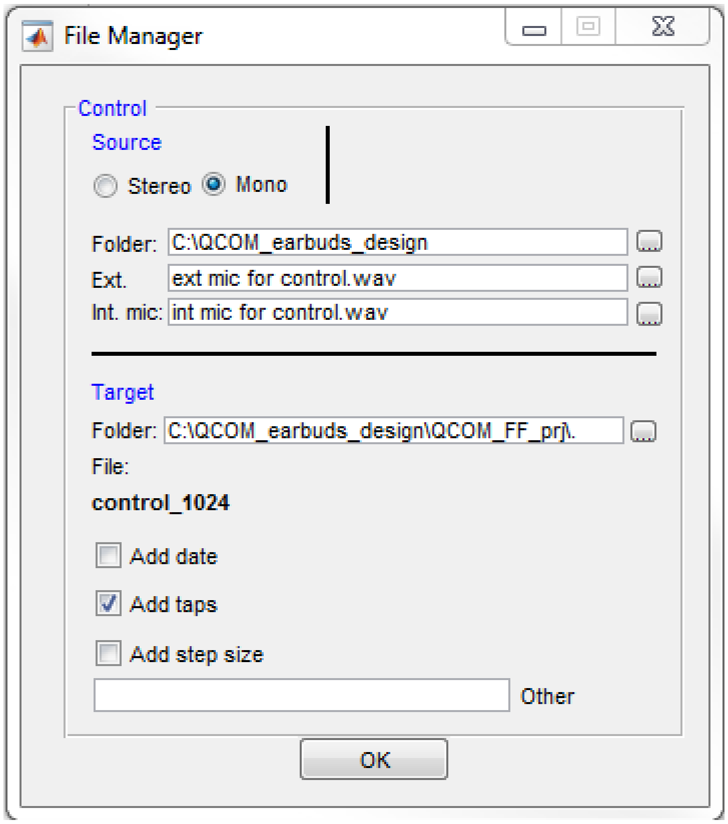

Figure 6-26 Correct files for controller

Calculate the FIR in the Control Panel. The IIR parameters cannot be modified here, so after the FIR is calculated, open a new panel to tune these parameters.

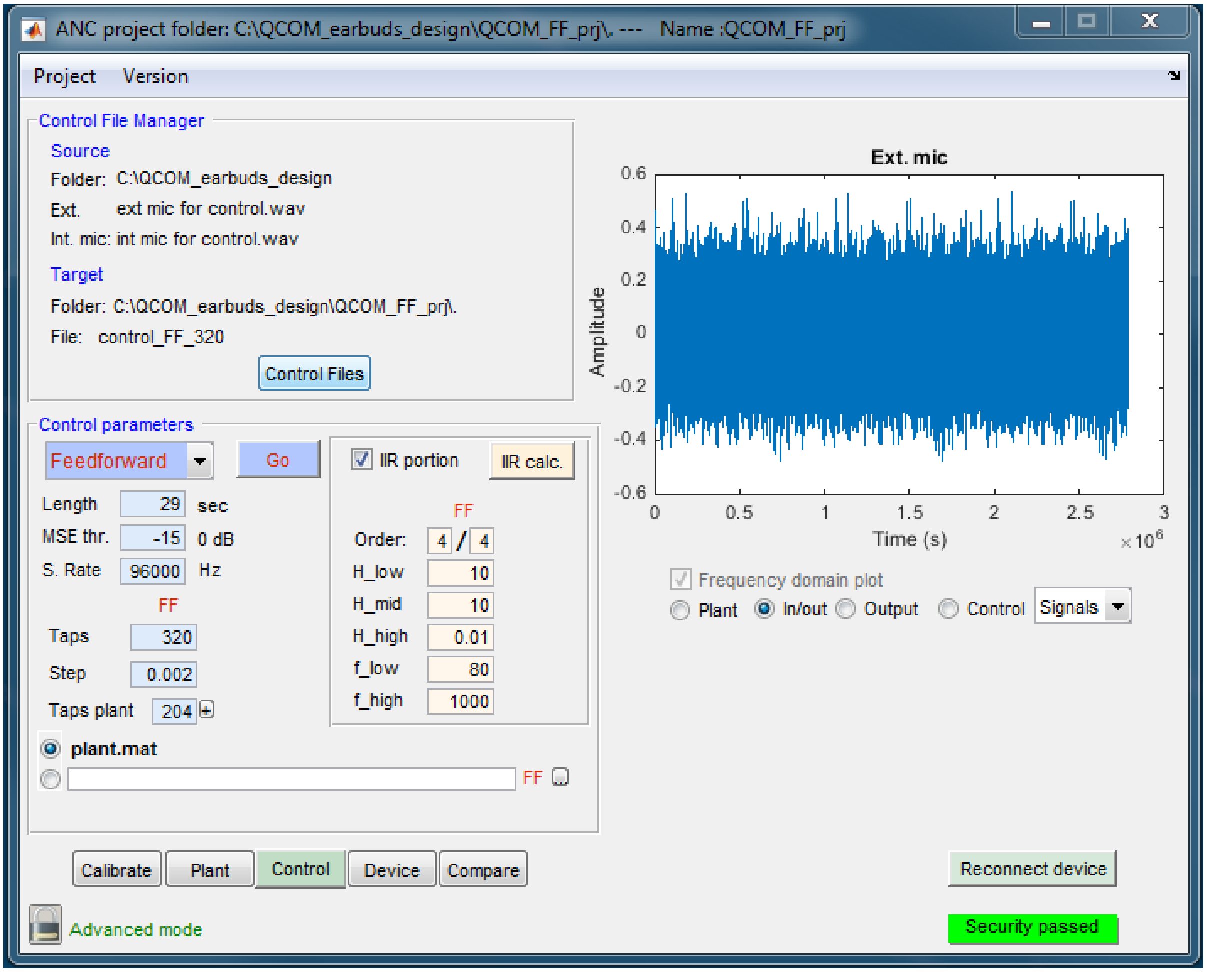

Figure 6-27 Earbud controller design parameters

While the control is calculated, a progress bar with the MSE and a plot of the current attenuation displays.

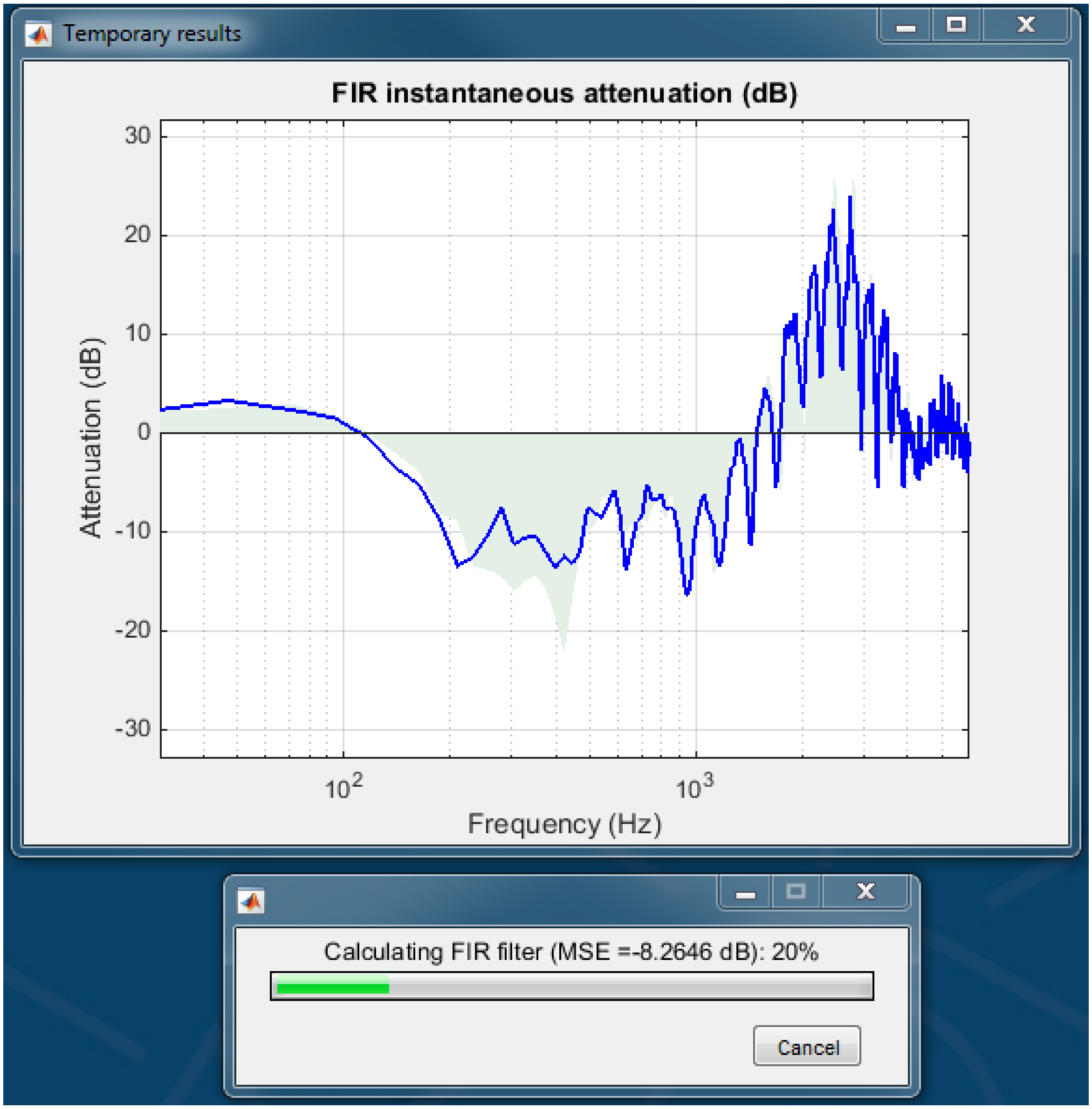

Figure 6-28 Earbud controller calculation in progress

If the resulting control requires the polarity to be inverted, an error dialog appears.

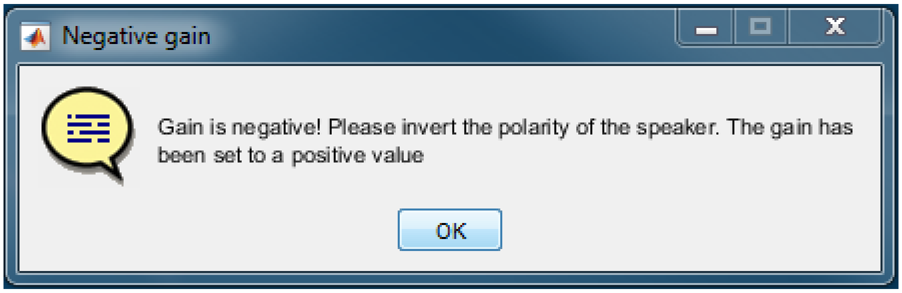

Figure 6-29 Polarity of the speaker warning

When choosing how many taps to use from the plant, select a point where the signal has died out.

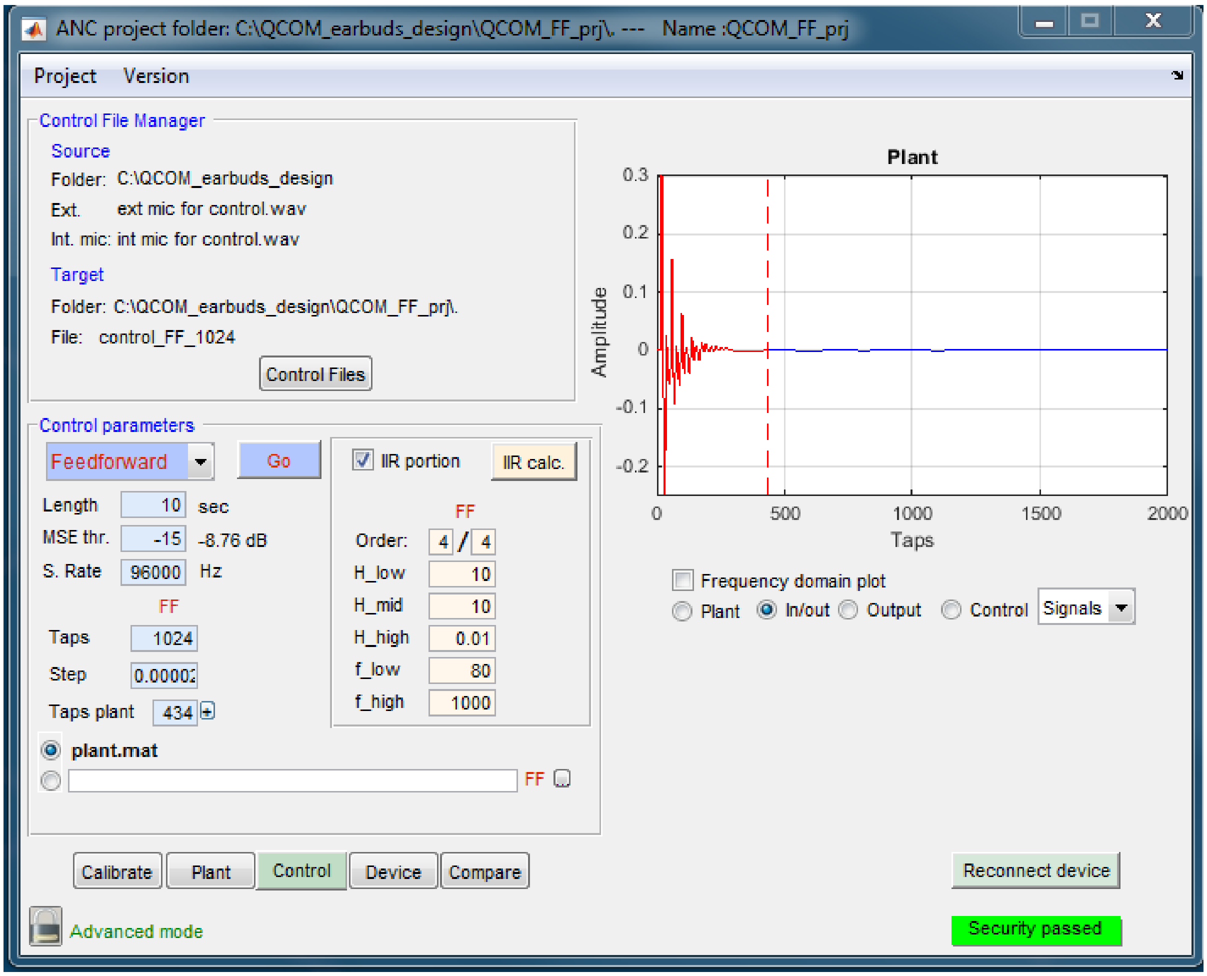

Figure 6-30 Selecting number of taps from the plant

The in/out option in the plot panel shows the quality of the controller's convergence. This example uses only 10 s of the recording to ensure that the controller converges with these parameters.

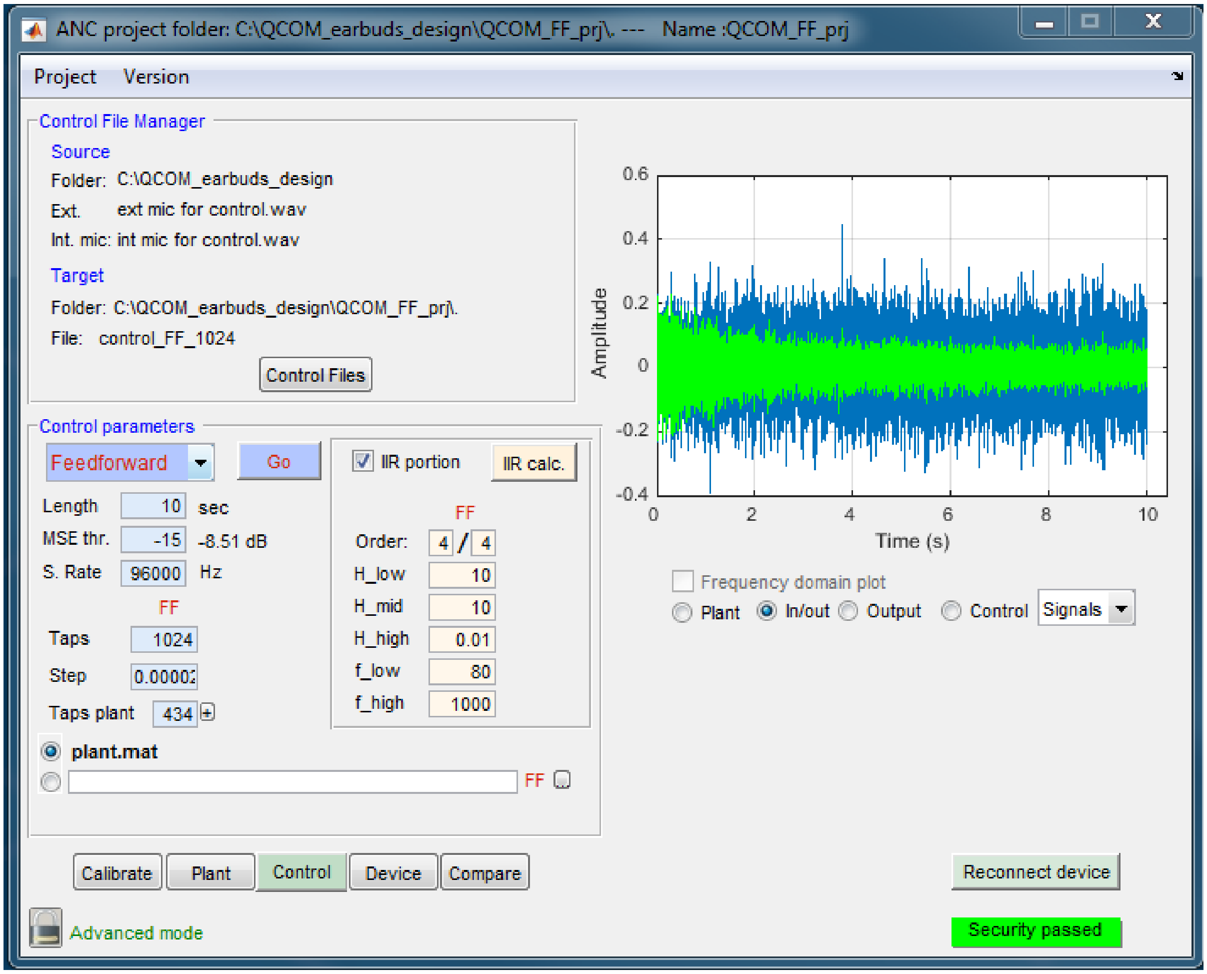

Figure 6-31 Controller convergence plot

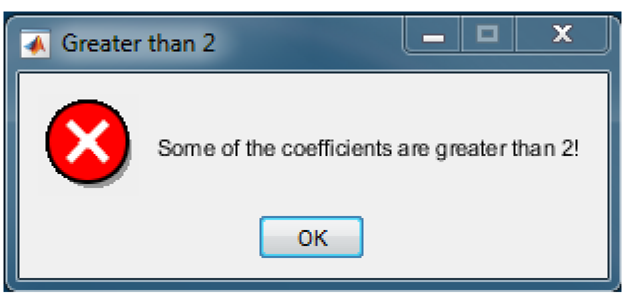

Figure 6-32 Coefficients magnitude warning

To attempt better convergence, increase the step size. In this example, the length is changed back to 29 s to process the entire signal.

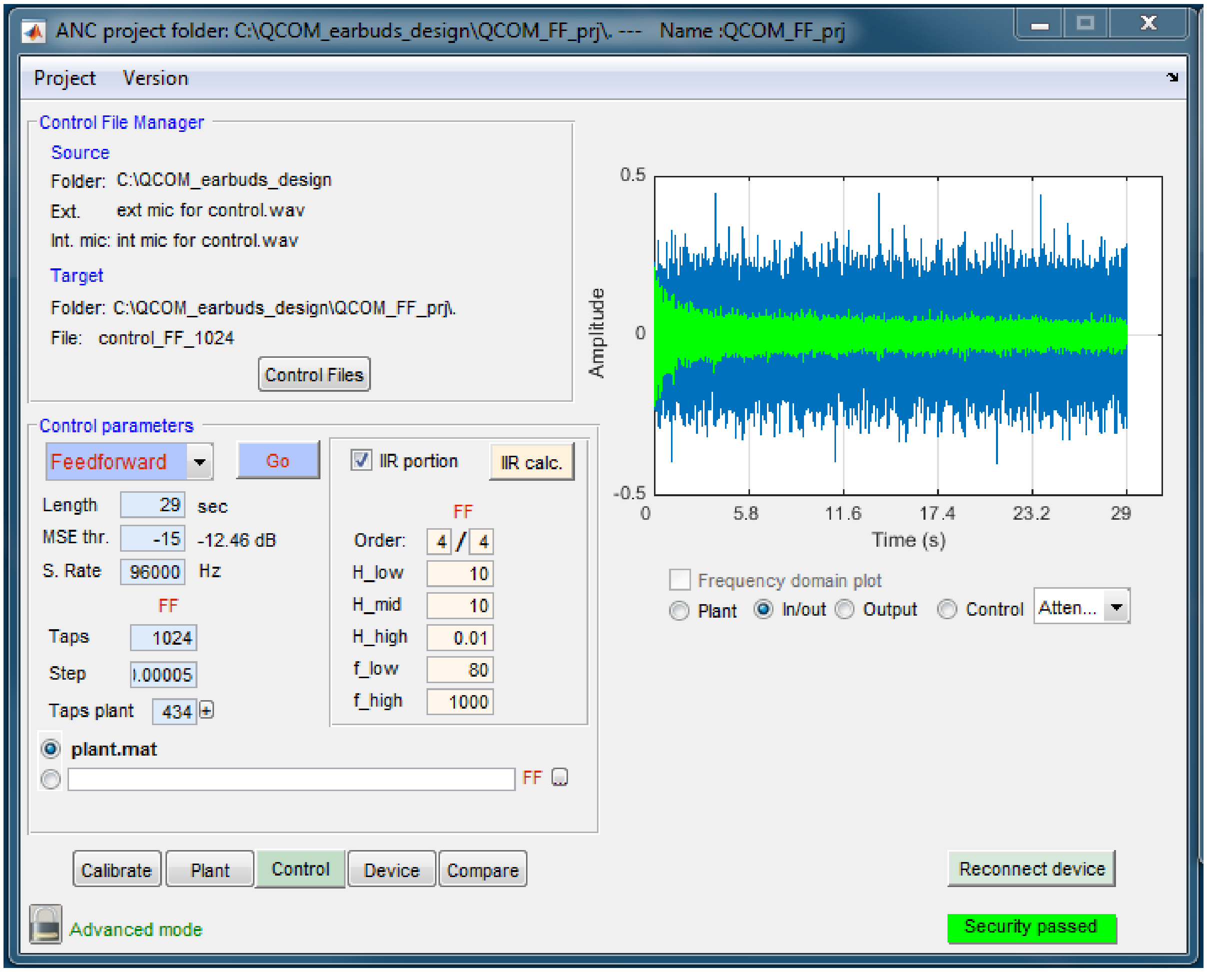

Figure 6-33 New earbud controller parameters

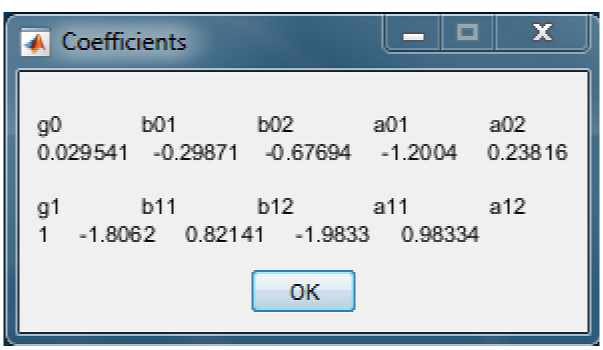

Figure 6-34 IIR coefficients

After the calculation is finished, plot the theoretical attenuation and compare the FIR and IIR performances. The plot shows that the attenuation for the FIR is good (black line), but the IIR parameters require tuning for a better fit (blue and red lines).

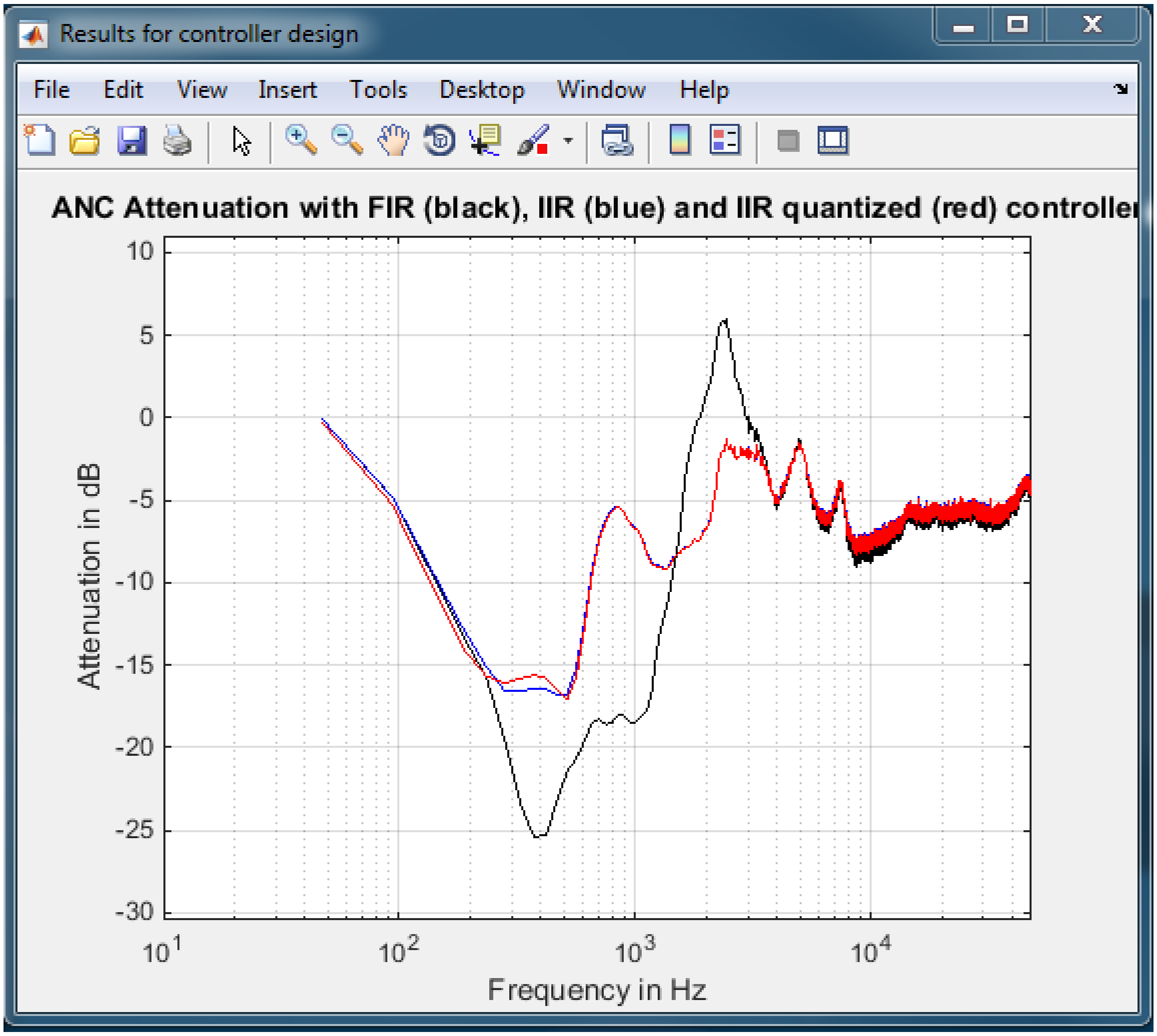

Figure 6-35 Theoretical attenuation for earbuds design

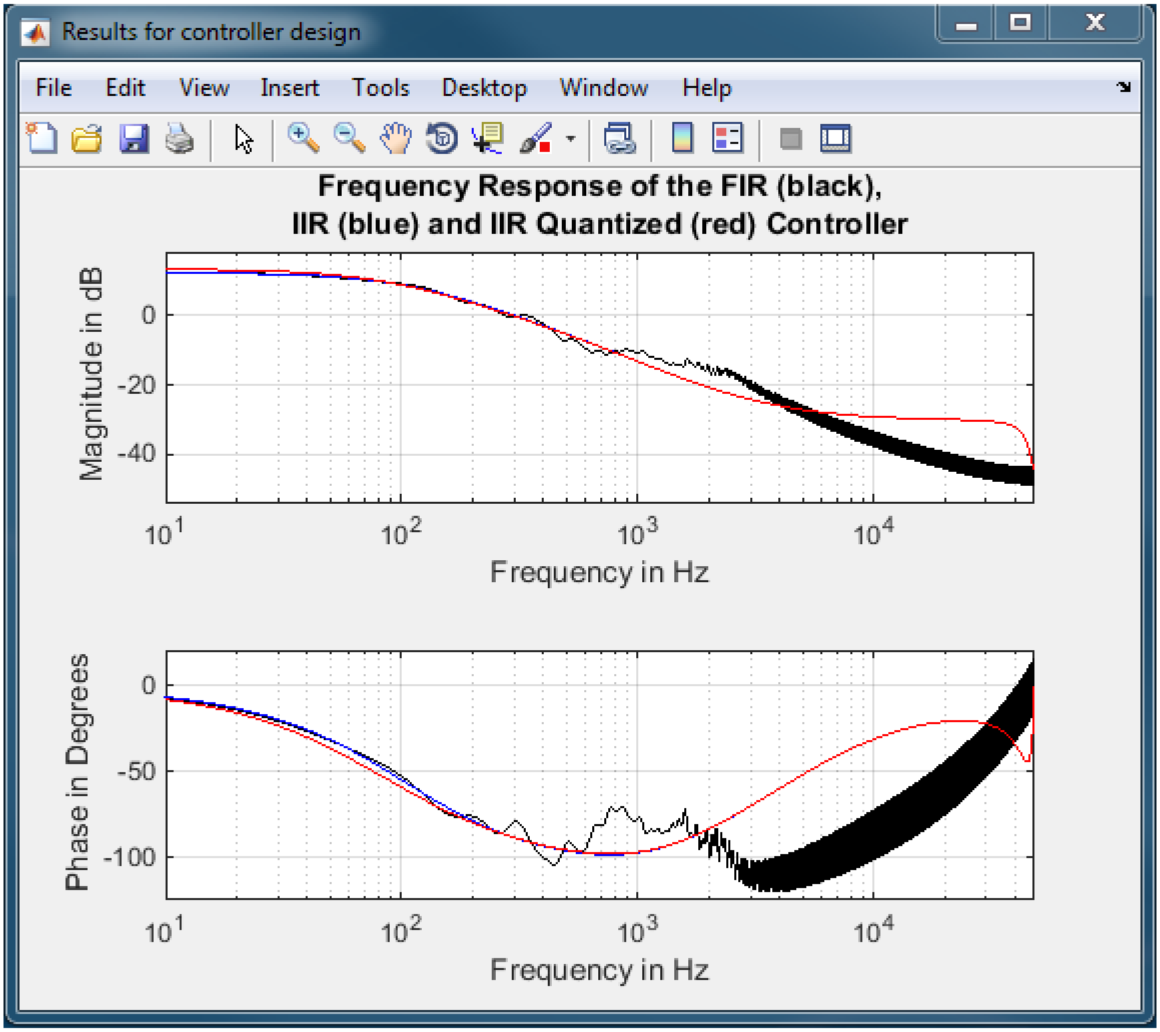

Figure 6-36 Frequency response for the controller

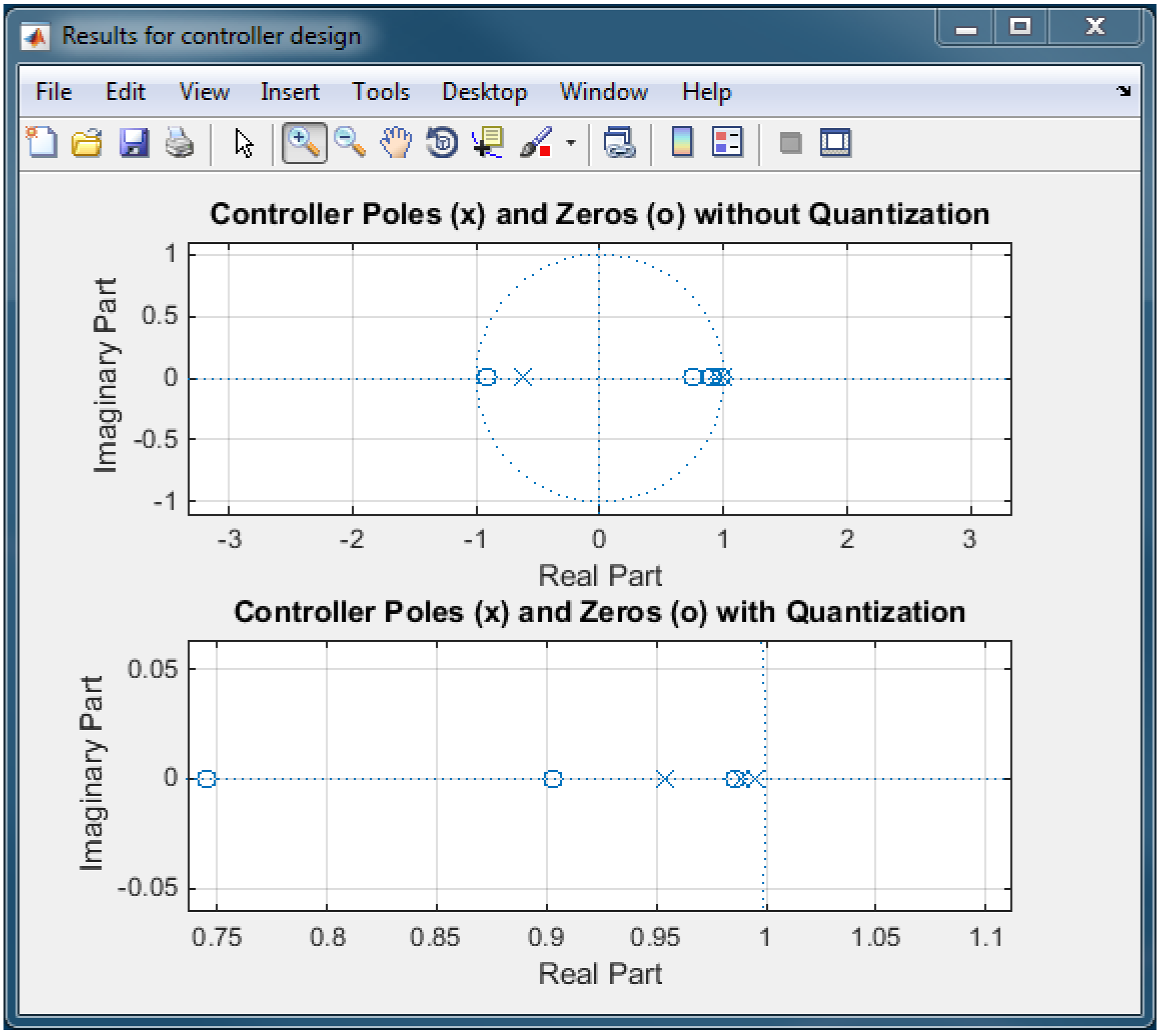

Figure 6-37 Zero-pole plot

To open the IIR panel and modify parameters, click **IIR**. This example shows plotted IIR response with the default parameters.

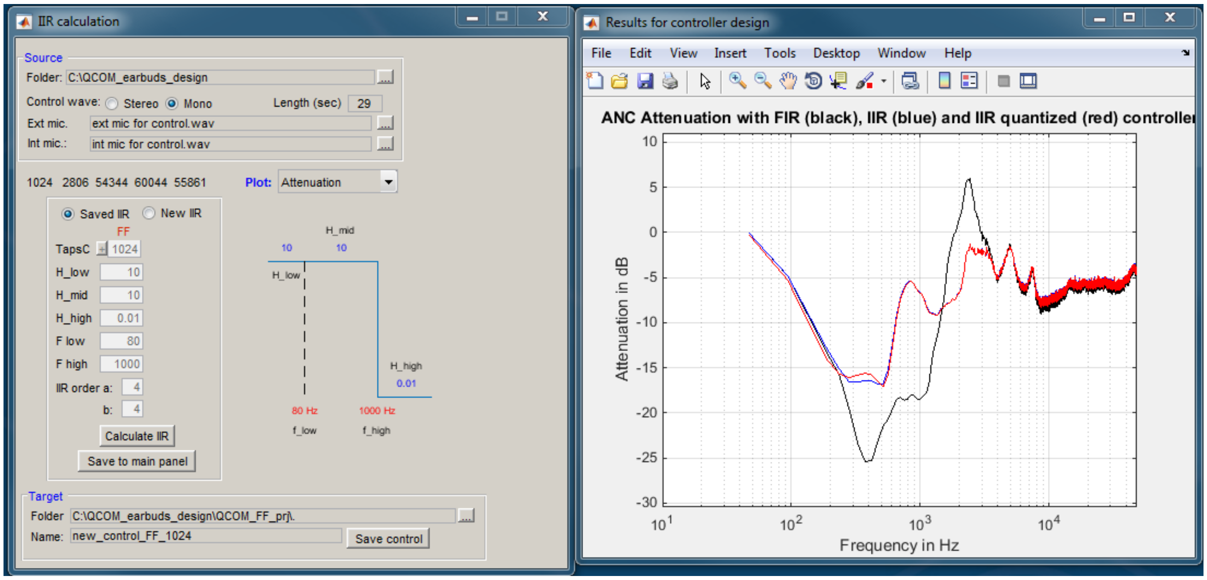

Figure 6-38 Calculating a better IIR filter

This example shows modified parameters.

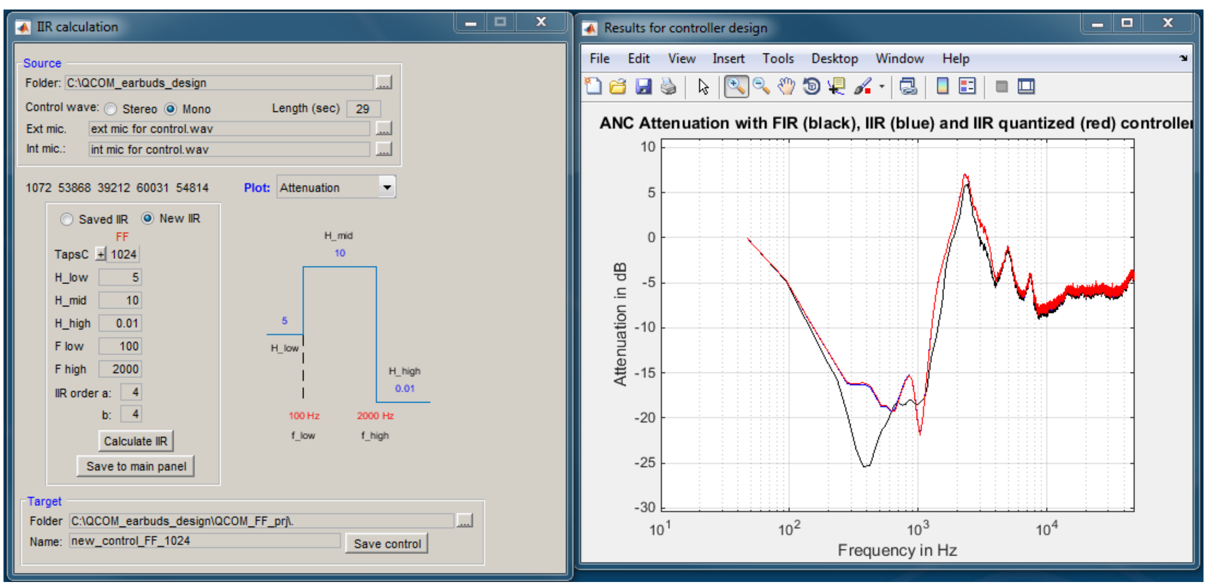

Figure 6-39 New IIR parameters

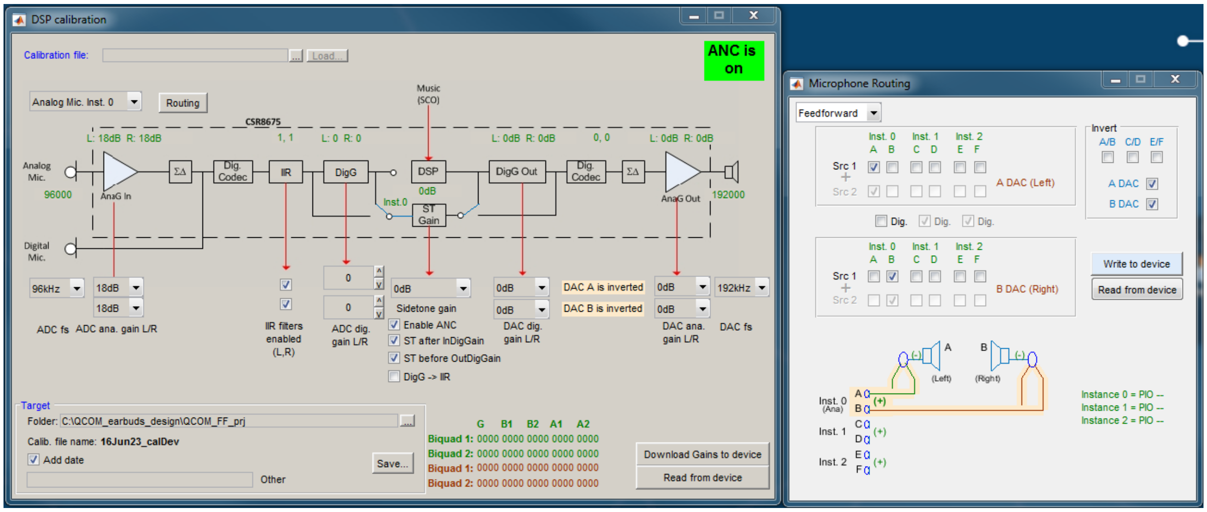

Figure 6-40 Calibration panel for earbud design

6.2.5 Download example design to CSR8675

After downloading the calibration and controller files to the CSR8675, select the appropriate calibration file and controller files for left and right channel.

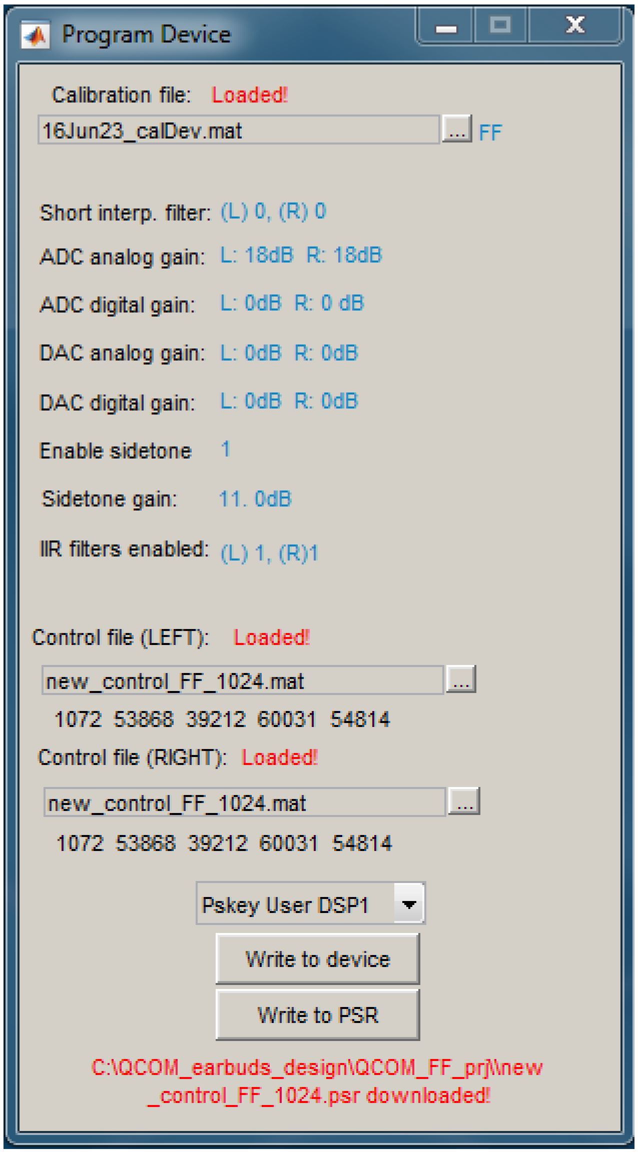

Figure 6-41 Downloading the controller to the CSR8675

6.2.6 Performance measurement of Qualcomm earbud

After the design is downloaded to the CSR8675, attach the headset and measure ANC performance. Make recordings with ANC on and off. The compare panel plots the performance, where the plot is the difference in PSDs between passive and active recordings.

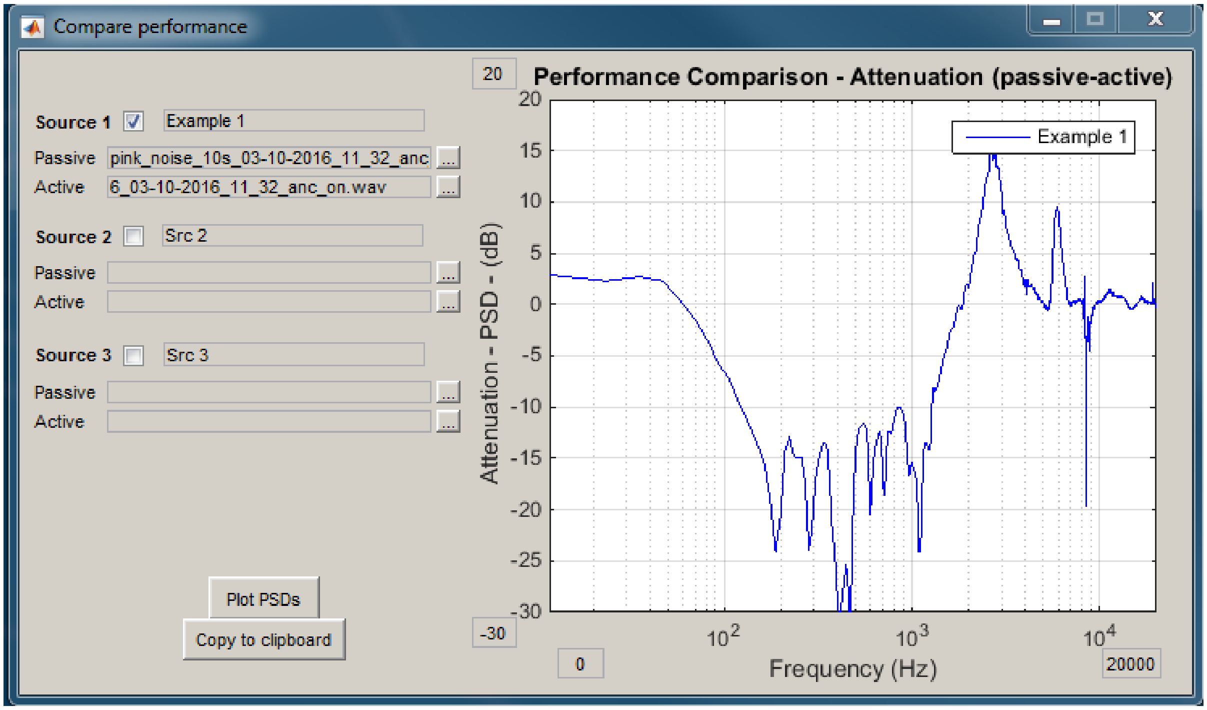

Figure 6-42 Measured Performance for the earbud design

6.3 Full Design

The full design uses the customers own headset.

6.3.1 Recordings

After calibration, the CSR8675 is ready to make the recordings needed for plant and controller design. Before making recordings, verify that the IIR filter is bypassed and the gain values match those set during calibration.

Recordings must be at the same sample rate as used by the ADC, which is 96 kHz. If the `.wav` files input to the Tuning Tool are at a different sample rate, then the IIR filter calculation is incorrect.

6.3.2 Plant recording

The plant recording captures the acoustic path inside the ear cup.

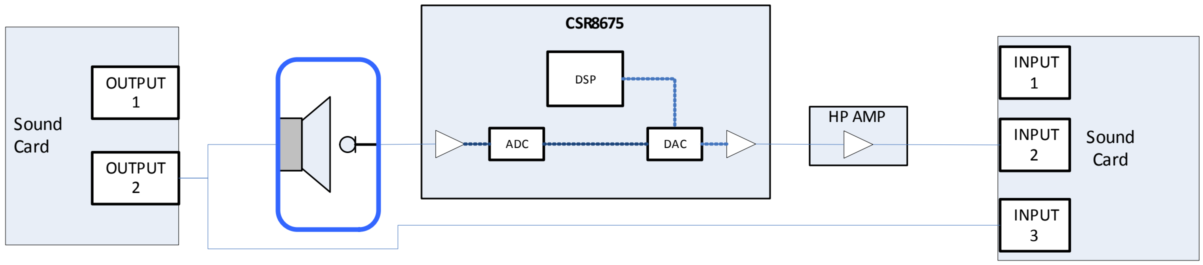

Figure 6-43 Plant recording setup

To record the plant:

1. Connect the system.
2. Play the `plant_chracterization.wav` file to output channel 2 on the sound card.
3. Record the internal microphone on channel 2 and the speaker reference signal on channel 3. Save these recordings for the plant calculation.

NOTE Configure the CSR8675 development board to use analog input for this step. The CSR8675 must be part of the signal path to capture the electrical delay of the CSR8675 in the plant recording.

The goal of this recording is to capture the transfer function between speaker and internal microphone, which includes the delay from both the acoustic path and CSR8675.

6.3.3 Controller recording

The controller recording captures the differences between the external and internal microphones.

To record the controller, ensure that the system is connected. The setup shows both channels of a CSR8675 (left and right) used for the two-microphone connections. The internal microphone can be in a HATS system or a microphone mounted inside the headset ear cup.

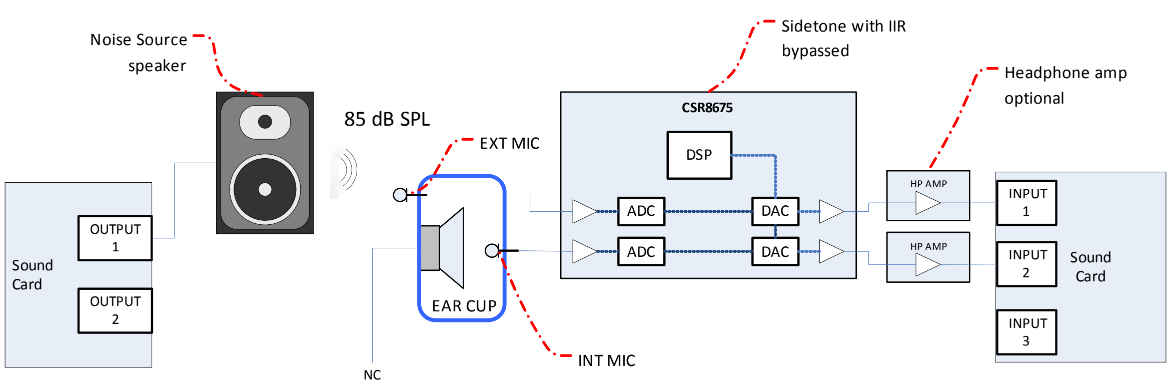

Figure 6-44 Controller recording setup

1. Using the source file `control_characterization.wav`, play the file to channel 1 on the sound card. The noise level in the chamber was previously calibrated to 85dBSPL. Do not play any signal to the headset speaker on channel 2. It is a good idea to disconnect the headset speaker completely for this step.
2. Record signals from the external microphone and internal microphone as connected to sound card inputs 1 and 2. The microphone recordings are time-aligned, so they must be routed through the CSR8675 to keep the delay equal.

6.3.4 Plant design

The plant is the impulse response of the acoustical path inside the ear cup and the electrical path (microphone, board, and speaker).

To design the plant:

1. Navigate to the plant panel by pressing the **plant** button.
2. Load appropriate plant recordings through the plant files panel.
3. Enter plant design parameters (such as taps, step, and size).
4. Click **Go** to initiate the plant calculation.

6.3.5 FIR controller design

After obtaining the plant impulse response, calculate the controller:

1. Navigate to the control panel by pressing the **control** button.
2. Load appropriate control recordings through the control files panel.
3. Enter control design parameters.
4. Press **Go** to initiate the control calculation.

6.3.6 IIR controller design

The IIR calc. panel design uses the same basic parameters as the regular controller design. This panel does not re-compute the FIR filter part of the design, and is therefore much faster to iterate through different parameters.

NOTE Only the IIR portion of the design is recalculated. Otherwise the resulting controller design is the same as the regular controller design panel.

6.3.7 Download design to CSR8675

Using the device panel, load the calibration file and a control file for each ear.

7 ANC testing

After downloading a completed design to the CSR8675, test the attenuation performance. Make recordings in an acoustic test chamber using the in-ear microphones of a HATS system or other test fixture.

With the headset properly mounted on the HATS, two recordings should be made, one with ANC active and another with ANC disabled. The ANC disabled recording should capture only the passive attenuation of the headset. To plot the attenuation performance, analyze these two recordings with the compare panel.

Because headset fit can vary, test the performance with multiple fits on the test fixture to find the average attenuation.

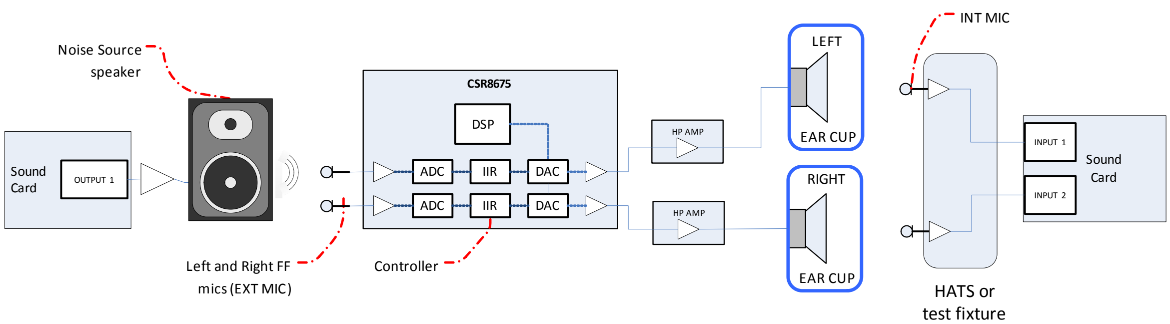

Figure 7-1 ANC testing setup

A PS Key reference

The ANC Tuning Tool generates a PSR file, which contains coefficients, gains, and configuration bits.

A.1 ANC security key

ANC requires a security key to function. Qualcomm provides customers with a security key and corresponding range of valid Bluetooth addresses.

Update PSKEY_FEATURE_LICENSE_KEY with a valid security key and use PSTool to merge.

A.2 DSP PS Keys

ADK sink application supports ANC filters in two different filter slots.

PSKEY_DSP1 stores the ANC Active mode coefficients. PSKEY_DSP2 can store a second set of coefficients, which is referred to as leak-through mode. Leak-through mode coefficients can be a bandpass filter designed to enable the user to listen to external sounds, or a second set of ANC coefficients.

The ANC Tuning Tool generates a PSR file with the PS Keys listed in Table A-1.

NOTE When using the ANC feature in an ADK project, ensure that PSKEY_DSP1 and PSKEY_DSP2 are present.

Table A-1 PS Keys generated by ANC

Word	Value	Channel
1	Gain0 Coefficient	Left
2	b01 Coefficient	Left
3	b02 Coefficient	Left
4	a01 Coefficient	Left
5	a02 Coefficient	Left
6	Gain1 Coefficient	Left
7	b11 Coefficient	Left
8	b12 Coefficient	Left
9	a11 Coefficient	Left
10	a12 Coefficient	Left
11	DC Blocker Coefficient	Left

Table A-1 PS Keys generated by ANC (cont.)

Word	Value	Channel
12	Gain0 Coefficient	Right
13	b01 Coefficient	Right
14	b02 Coefficient	Right
15	a01 Coefficient	Right
16	a02 Coefficient	Right
17	Gain1 Coefficient	Right
18	b11 Coefficient	Right
19	b12 Coefficient	Right
20	a11 Coefficient	Right
21	a12 Coefficient	Right
22	DC Blocker Coefficient	Right
23	Analog ADC Gain	Left
24	Analog ADC Gain	Right
25	Analog DAC Gain	Left
26	Analog DAC Gain	Right
27	Digital ADC Gain	Left
28	Digital ADC Gain	Right
29	Digital DAC Gain	Left
30	Digital DAC Gain	Right
31	Configuration	Both

Table A-2 Word 31 Bit Fields

Word 31 Bits	Value
[15:6]	Reserved
[5]	ADC sampling rate: ■ 1 : 96 kHz ■ 0: 48 kHz
[4]	DAC sampling rate ■ 1: 192 kHz ■ 0: 96 kHz
[3]	DAC invert left channel ■ 1: Inverting ■ 0: Non-inverting
[2]	DAC invert right channel ■ 1: Inverting ■ 0: Noninverting
[1:0]	Reserved

A.3 More PS Keys used by ANC

The ADK has other tools that store ANC information and configuration data in PS Keys.

- Sidetone gain information is stored in in the Bits [3:0] of PSKEY_USR44.
- The sink configuration tool stores information about ANC user events and configuration, these parameters can only be modified using the sink configuration tool
- The ANC PEQ provided in music manager is a PEQ block which toggles on and off with ANC. The Universal Front End configures this PEQ and save the values to PS Keys.

Document references

Document	Reference
Audio Sink Application ANC User Guide	80-CT526-1 / CS-00330567-UG
BlueSuite User Guide	80-CT512-1 /CS-00118613-UG
CSR8675 Data Sheet	80-CT512-1 / CS-00232426-DS
PSTool User Guide	80-CT424-1 / CS-00101505-UG
xIDE User Guide	80-CT405-1 /CS-00101500-UG

Terms and definitions

Active Mode	Mode where the CSR8675 ANC function is enabled
ADC	Analog-to-digital Converter
ADK	Audio Development Kit
ANC	Ambient Noise Cancellation
Qualcomm® BlueCore™	Group term for Qualcomm's range of Bluetooth wireless technology chips
Bluetooth	Set of technologies providing audio and data transfer over short-range radio connections
Controller	Calculated filter based on the transfer function from external microphone to internal microphone and the Plant.
CSR8675	QualcommIC, which supports ANC
cVc	Clear Voice Capture
DAC	Digital-to-analog Converter
DC	Direct Current
DSP	Digital Signal Processor
External microphone (EXT MIC)	Microphone mounted on the headset to capture noise outside the ear cup. This microphone makes recordings for the design of the controller and picks up noise in the final feed-forward system.
FIR	Finite Impulse Response (filter)
GUI	Graphical User Interface
HATS	Head and Torso Simulator
IC	Integrated Circuit
IIR	Infinite Impulse Response (filter)
Internal microphone (INT MIC)	A reference microphone mounted inside the headset ear cup or inside a Head And Torso Simulator (HATS) system, which makes recordings for design and measurement.
Leak-through Mode	Mode where the ANC function is disabled and the microphones amplify ambient audio, enabling the user to hear sounds such as speech.
LED	Light Emitting Diode
MIC	Microphone
PCB	Printed Circuit Board
Plant	Transfer function from the headset speaker to the internal microphone.
PS Key	Persistent Store Key
Sidetone	Signal path between ADC and DAC on CSR8675, which contains a programmable IIR filter.

SPI	Serial Peripheral Interface
SPL	Sound Pressure Level
USB	Universal Serial Bus
xIDE	Qualcomm's Integrated Development Environment