

EZAIRO[®] 7150 SL

Firmware Bundle V1 Technical Overview


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CHAPTER 1

Introduction

This document provides a high-level overview of the technical features of the Ezairo 7150 SL Firmware Bundle V1, which is included as part of the Ezairo Preconfigured (Pre) Suite for hearing solutions.

FURTHER READING

For additional technical details and more information regarding firmware bundle algorithm parameters, please refer to the following resources:

- The Ezairo 7150 SL Datasheet
- Ezairo 7150 SL Firmware Bundle User Guide*
- Ezairo Sound Designer Software Development Kit (SDK) Programmer's Guide*
- Ezairo Sound Designer Software User's Guide*

*Can be requested through your local ON Semiconductor representative or authorized distributor

CHAPTER 2

Overview

Ezairo Pre Suite is a comprehensive, turnkey system solution for hearing aids consisting of Ezairo-based hybrid modules augmented with algorithm and firmware bundles ready to perform high-end, wireless-enabled, hearing aid functions. Also included is a set of advanced development tools, including a cross-platform SDK and evaluation and configuration software that enables rapid prototyping and flexible product deployment in the field.

EZAIRO 7150 SL HYBRID MODULE

Ezairo 7150 SL is an open-programmable DSP-based hybrid which enables wireless connectivity in hearing aids and cochlear implants.

The miniature hybrid is based on the Ezairo 7100 DSP platform, which is the industry's most integrated, flexible, and power efficient, single-chip solution.

Ezairo 7150 SL includes a multi-protocol radio integrated circuit (IC) ideally suited for Bluetooth® low energy technology and 2.4 GHz wireless applications.

To store important hearing aid parameters, Ezairo 7150 SL also features 2 Mb of EEPROM memory.

EZAIRO 7150 SL FIRMWARE BUNDLE V1

The Ezairo 7150 SL Firmware Bundle V1 is a complete, preconfigured hearing aid algorithm suite running on the Ezairo 7150 SL hybrid. The firmware bundle features a high quality, low-latency filterbank running at a sampling rate of 16 kHz (8 kHz audio bandwidth) with 24-bit precision audio processing.

Algorithms included in the Ezairo 7150 SL V1 firmware bundle are:

- Multi-channel Wide Dynamic Range Compression (WDRC) with In-Channel AGCo/Limiter
- Adaptive Feedback Cancellation
- Static and Adaptive Directional Microphone Processing
- Noise Reduction
- Environmental Classification
- Programmable Biquad Filters
- Wideband AGCo/Limiter
- 8 Configurable Profiles (or “Memories”)
- MMI Profile Switching and Volume Control
- Multi-Channel Equalization (EQ)
- Data Logging

The firmware bundle also contains the following 2.4 GHz wireless features:

- Remote Control over Bluetooth Low Energy (CoBLE)
- Stereo Audio Streaming using a Proprietary 2.4 GHz Protocol
- Wireless Adjustment and Tuning of Algorithm Parameters (Wireless Fitting) using Bluetooth Low Energy Technology

The novel architecture of Ezairo 7150 SL combined with an efficient, real-time framework allows all of these features to be realized with industry-leading, ultra-low power consumption, preserving the battery life of these extremely resource constrained hearing devices.

CHAPTER 3

System Overview

A block diagram of the Ezairo 7150 SL Firmware Bundle V1 is shown in Figure 1.

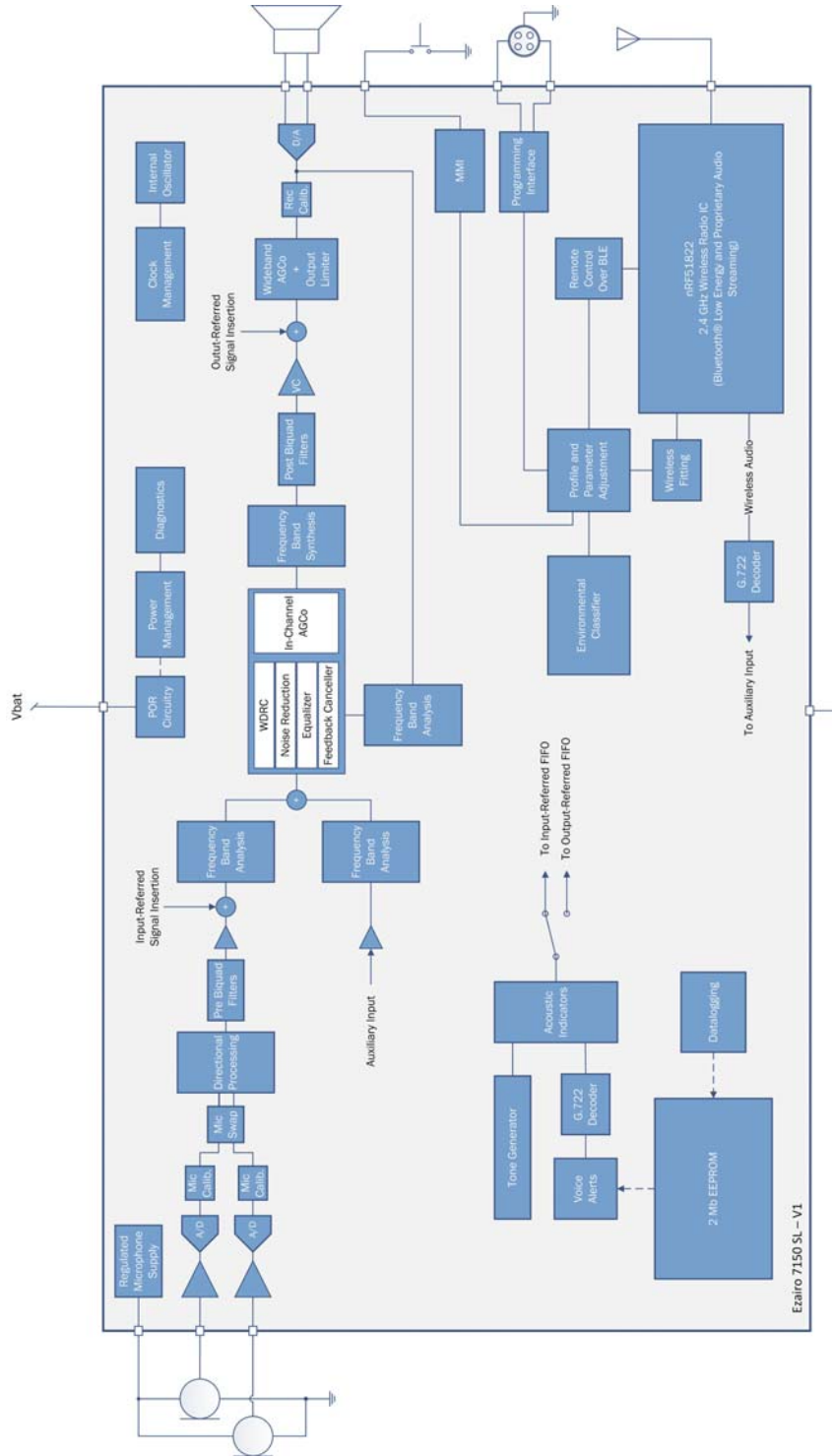
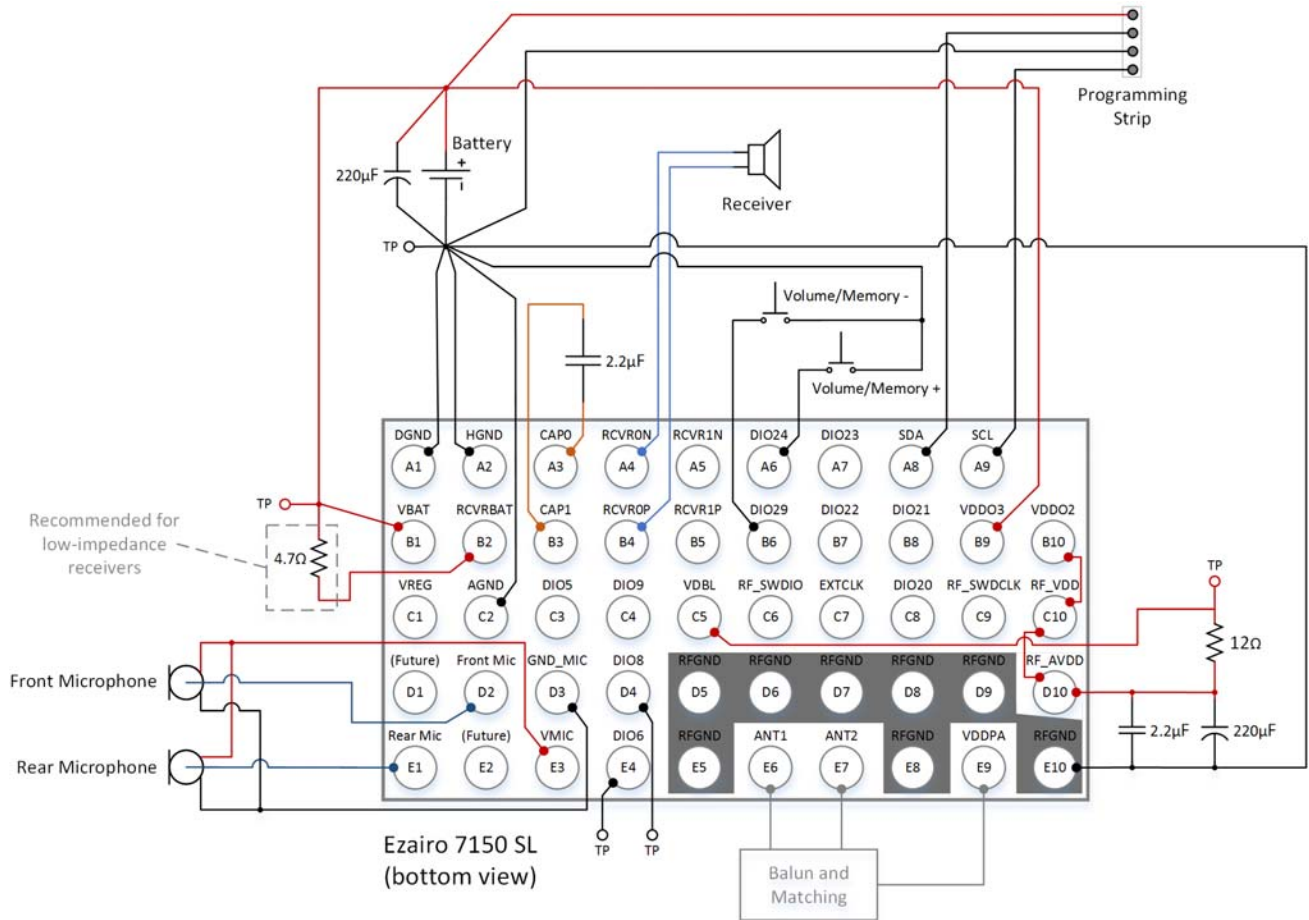


Figure 1. Ezairo 7150 SL Firmware Bundle V1 Block Diagram

The Ezairo 7150 SL – V1 firmware bundle is designed to work with the following external connections:



NOTE: For the purposes of wireless certification, it is recommended that the following signals are accessible or brought out to solderable test points: VBAT, GND, VDBL, DIO6, DIO8

Ezairo 7150 SL Pre Suite V1 Connection Diagram
December 11, 2017

Figure 2. Ezairo 7150 SL Typical Connection Diagram

CHAPTER 4

Ezairo Sound Designer Software

The Ezairo Sound Designer Software consists of two integrated software packages that allow hearing aid manufacturers to design, test and deploy hearing aids based on Ezairo Pre Suite. The two main software components are:

- The Ezairo Sound Designer Software Development Kit (Sound Designer SDK)
- A graphical user interface presented as the Sound Designer Software Application

SOUND DESIGNER SOFTWARE DEVELOPMENT KIT (SDK)

The Ezairo Sound Designer SDK provides an extensive Application Programming Interface (API) and accompanying sample code for configuring and modeling ON Semiconductor’s firmware bundles for hearing aids and assistive listening devices.

The Sound Designer SDK is intended to significantly shorten the software development time required to add new hearing devices to a fitting module or standalone fitting software. It simplifies or eliminates the repetitive tasks of programming different types of hearing aids. This allows more time for the fitting software development team to focus on innovative user interfaces, the audiologist/patient experience, and fitting methodologies. User-facing fitting software can be built on top of the Sound Designer Software API.

The Sound Designer Software API also provides interfaces that can be used to calibrate and configure hearing aids at the manufacturing level. Calibration allows for the compensation of gain mismatches due to process variability of microphones and receivers.

The Ezairo Sound Designer SDK is cross-platform and is available for Windows[®], Android[™], and iOS[®].

SOUND DESIGNER SOFTWARE

The Sound Designer Software Application is built on top of the Sound Designer SDK, and includes software tools needed for designing and evaluating Ezairo-based hybrid modules, and utilities for the calibration, configuration, and testing of hearing devices at the manufacturing level. Accurate acoustic modeling helps manufacturers develop new products more rapidly by making it unnecessary to measure acoustic curves for every parameter combination.

Figure 3 illustrates how the Sound Designer Software components relate to one another.

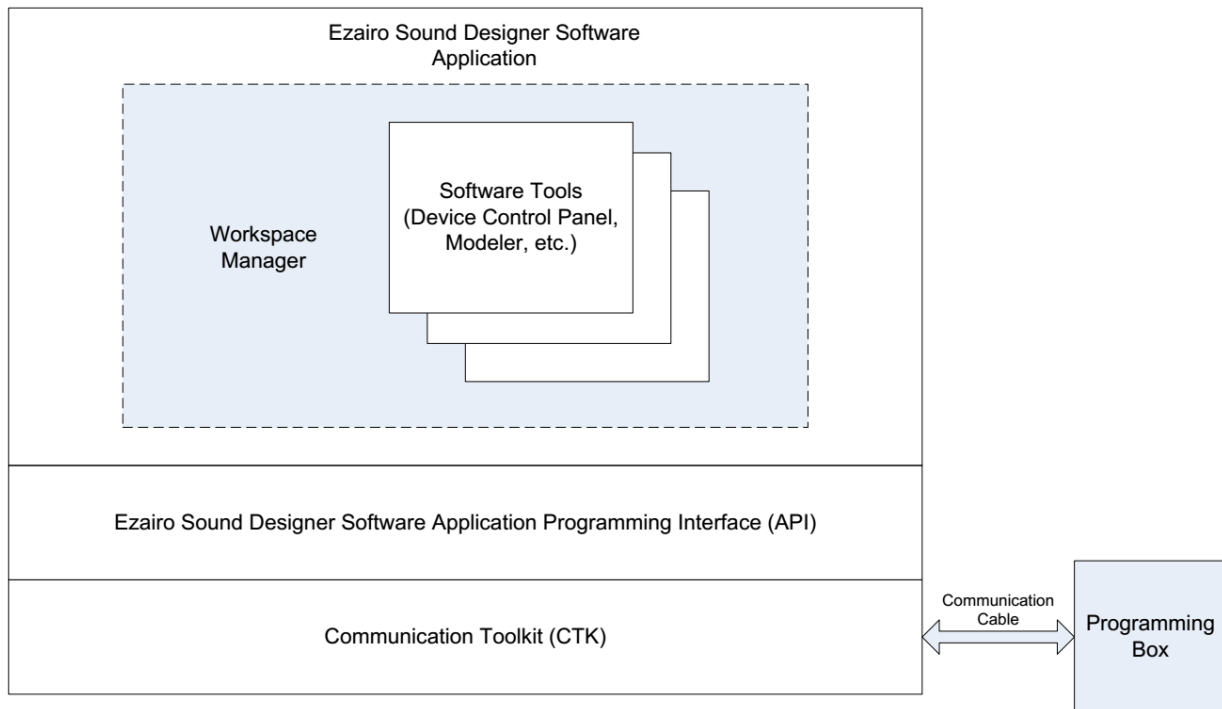


Figure 3. Sound Designer Software Components

The components of the Sound Designer Software Application are:

- Modeler
- Map Editor
- Library Manager
- Control Panel
- Calibration and Configuration (CalConfig)

Figure 4 shows the main screen of the Sound Designer Software Application.

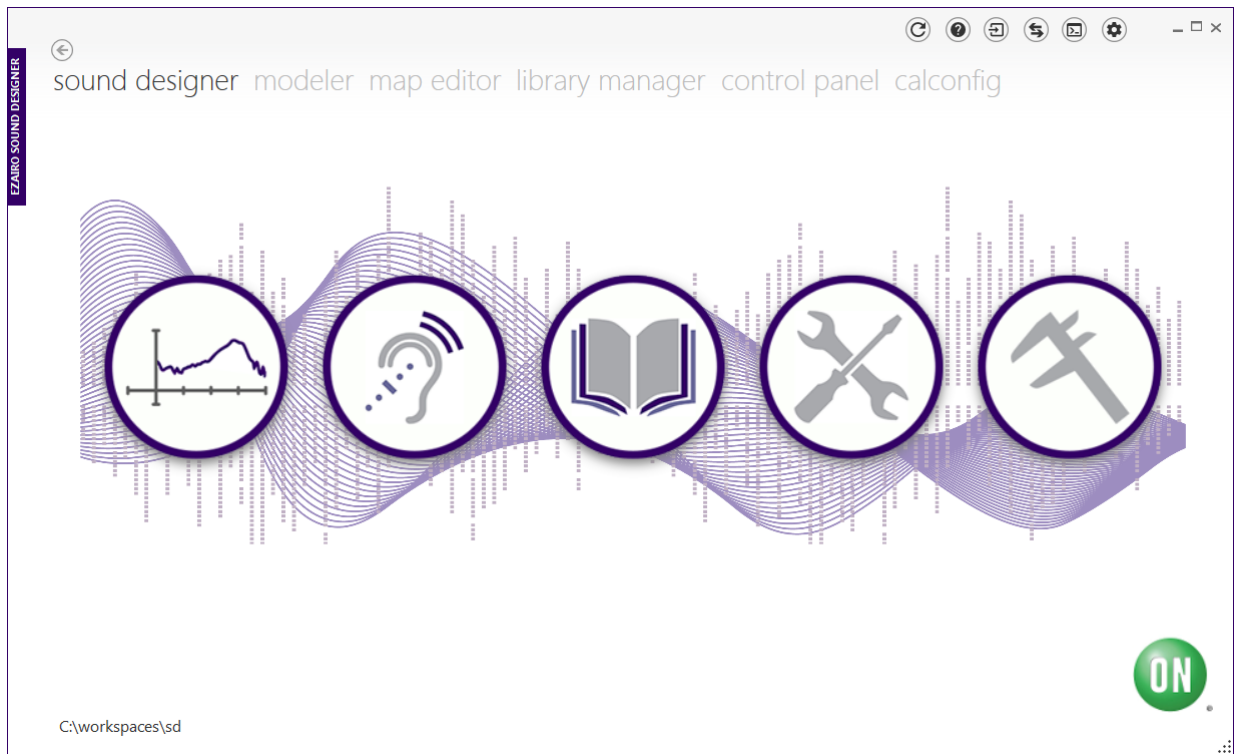


Figure 4. Sound Designer Software Application

The Ezairo Sound Designer Software can be used with or without a connected hearing aid device. Without a connected device, Sound Designer Software can still be used to design products, tailor their parameters, and view an accurate model of the resulting device response.

With a connected device, the Ezairo Sound Designer Software can be used to test designed products and observe their behavior. Depending on the connected device, live display options are available to view specific internal data, such as level detectors, streamed back in real-time.

CHAPTER 5

Wireless Features

The wireless features of the Ezairo 7150 SL Firmware Bundle V1 support connecting to the device via standard Bluetooth low energy technology providing control over the hearing aid and its parameters through standard Bluetooth low energy GATT services and characteristics. They also enable audio streaming from a transmitter to the hearing aid (or hearing aids in a stereo configuration) at a data rate of either 48 or 64 kbps.

Wireless features are enabled and controlled through the standard firmware bundle parameter interface, found in the RF firmware module. For a complete list of wireless parameters available for configuration refer to the *Firmware Bundle for Ezairo 7150 SL User Guide and Reference*.

ENABLING WIRELESS FEATURES

By default, wireless features are disabled. To enable wireless features, the global parameter **Wireless Enable** must be set to 1. Once enabled, you may connect to the device with any standard Bluetooth low energy client.

CHANGING THE DEVICE NAME

The wireless device name can be changed by modifying the **Device Name** parameters. The device name is formed by the concatenation of all five 24-bit device name parameter values and is interpreted as an array of ASCII characters. Each parameter encodes three characters, which means that the device name can be a maximum of 15 characters in length. Unused characters must be zero-padded (see example). The device must be power-cycled for a device name change to take effect.

For example, the device name “Ezairo 7150SL” is encoded as 0x45 0x7A 0x61 0x69 0x72 0x6F 0x20 0x37 0x31 0x35 0x30 0x53 0x4C. Combining the characters into parameter values gives:

- Device Name 0: 0x457A61
- Device Name 1: 0x69726F
- Device Name 2: 0x203731
- Device Name 3: 0x353053
- Device Name 4: 0x4C0000

CONNECTING TO THE DEVICE

Advertising on Boot

Upon boot (or reset), the device advertises with the configured device name for a configurable time period. If the bond table in the device is empty, the device advertises to all devices (in Pairing Mode) for the time period defined by the **Pairing Advertising Mode Timeout** parameter. If the bond table in the device is not empty (e.g., it has previously been paired and bonded to another device), the device advertises to whitelisted/bonded devices (in Whitelist Advertising Mode) for the time period defined by the **Whitelist Advertising Mode Timeout** parameter.

NOTE: A timeout value of 0 means to advertise indefinitely.

Initial Connection (Pairing)

Initially the device’s bond table contains no entries. The initial connection can be made by placing the device in pairing mode. This can be done by a super-long (> 10 second) press on DIO 24 (this can be remapped through the MMI module), or by rebooting the device. When this DIO is held for > 10 seconds, the device enters pairing mode for the period defined by the **Pairing Advertising Mode Timeout** parameter. During this period, any Bluetooth low energy client can connect and bond with the hearing aid.

While advertising, a client can discover and connect (pair) with the device to discover and use available services. The pairing method supported is unauthenticated (“Just Works”), which requires no passkey.

Disconnection

When a client disconnects from the hearing aid, the hearing aid again begins advertising to whitelisted/bonded devices for the period defined by the **Whitelist Advertising Mode Timeout** parameter.

Clearing the Bond Table

To clear the device's bond table, use a super-long press (more than 10 seconds) on DIO 29 (this configuration can be remapped through the MMI module). When the device's bond table is cleared, the device re-enters pairing mode and once again advertises to all clients for the period defined by the **Pairing Advertising Mode Timeout** parameter.

WIRELESS SERVICES AND CHARACTERISTICS

Once connected, the client can discover services made available by the hearing aid device. The supported services are:

Table 1. SUPPORTED SERVICES

Service	UUID
Control over BLE (CoBLE)	4F85A001-8262-11E5-84A0-0002A5D5C51B
Battery Service (BAS)	Standard BLE Service

For additional details on the characteristics available within these services refer to the *Firmware Bundle for Ezairo 7150 SL User Guide and Reference*.

Control over BLE (CoBLE) Service

The CoBLE service offers control over a few hearing aid features; specifically volume control, microphone and auxiliary attenuations, current profile (or memory), and event trigger which are exposed as characteristics.

Volume:

The volume characteristic is used to set the hearing aid volume to a specific level or to read back the current volume level. The data associated with this characteristic is a single byte, with a range that represents the volume level as a percentage and maps linearly onto the system volume level as $[0,100] \rightarrow [\text{Volume Minimum (dB)}, 0 \text{ dB}]$, where "Volume Minimum" is a global parameter that specifies the lower bound for system volume.

Example 1: If Volume Minimum is set to -50 dB and the volume is updated through the BLE characteristic with a value of 100, then the hearing aid system volume is set to 0 dB.

Example 2: If Volume Minimum is set to -50 dB and the volume is updated through the BLE characteristic with a value of 50, then the hearing aid system volume is set to -25 dB.

Local changes to the volume (e.g., through a pushbutton event) can be detected by reading the characteristic value on the client side or by enabling notifications.

Current Memory:

The current memory characteristic allows a BLE central device to set the current profile in use by the hearing aid. The data associated with this characteristic is a single byte, with a range of [1, 8]. Any values outside of this range that are written to the device are ignored.

Local changes to the memory (e.g., through a pushbutton event) can be detected by reading the characteristic value on the client side or by enabling notifications.

Event Trigger:

The event trigger characteristic allows a BLE central device to trigger an event on the hearing aid. Currently, the events supported are Volume Up, Volume Down, Next Memory and Previous Memory.

Memory Information:

The memory information characteristic describes the memories in the device in order to simplify user interaction from the client side. This characteristic is read-only and stores the number of memories in the device, enabled/disabled status and a 7-bit identifier for each memory.

Auxiliary Attenuation:

The auxiliary attenuation characteristic is used to set and read the auxiliary channel (wireless audio) attenuation level. The data associated with this characteristic is a single byte, with a range that represents the attenuation as a percentage and maps linearly onto the auxiliary volume level as [0,100] → [-128 dB, 0 dB]. Any values outside of this range that are written to the device are ignored.

Example 1: If the Auxiliary Attenuation is updated through the Bluetooth low energy characteristic with a value of 100, the hearing aid Auxiliary Attenuation is set to 0 dB.

Example 2: If the Auxiliary Attenuation is updated through the Bluetooth low energy characteristic with a value of 50, the hearing aid Auxiliary Attenuation is set to -64 dB.

Changes to this characteristic from the software can be detected by reading the characteristic value on the client side or by enabling notifications.

Microphone Attenuation:

The microphone attenuation characteristic is used to set and read the microphone channel attenuation level. The data associated with this characteristic is a single byte, with a range that represents the attenuation as a percentage and maps linearly onto the microphone volume level as [0,100] → [-128 dB, 0 dB]. Any values outside of this range that are written to the device are ignored.

Example 1: If the Microphone Attenuation is updated through the Bluetooth low energy characteristic with a value of 100, the hearing aid Microphone Attenuation is set to 0 dB.

Example 2: If the Microphone Attenuation is updated through the Bluetooth low energy characteristic with a value of 50, the hearing aid Microphone Attenuation is set to -64 dB.

Changes to this characteristic from the software can be detected by reading the characteristic value on the client side or by enabling notifications.

Other Characteristics:

The Control Over BLE service also contains a set of characteristics used by the Sound Designer SDK to interact with the device and read and write parameters. These characteristics should not be accessed directly by the user.

Battery Service (BAS)

The Battery Service is a Bluetooth standard service with a single characteristic, which is the current battery charge represented as a percentage value from 0 to 100.

The current implementation uses a fixed discharge profile with three regions to convert battery voltage to percentage. Percentage values are linearly interpolated based on the region into which they fall. The regions are described by the following points:

- 1.05 V = 0%
- 1.24 V = 10%
- 1.31 V = 90%
- 1.45 V = 100%

The battery service implementation in the device refreshes the battery level value every 5 seconds (only if the level has changed since the last update). Changes to the battery level can be detected by reading the characteristic value on the client side or by enabling notifications.

Notifications

Any characteristics whose properties are set to allow notifications support having their values updated automatically on the client. To enable this, the device must be bonded and then have these characteristics enabled for notifications.

CHAPTER 6

Audiology Features

FRONT END AND HARDWARE CONTROL

Overview

The Front End (FE) and Hardware Control (HC) modules oversee and control the front end signal processing (including directional, and pre- and post- biquads) and routing. As can be seen in Figure 1, two separate input signal paths (or input channels) exist in the firmware: the main microphone(s) input channel, and an auxiliary input channel. The microphone input channel (also known as the **Front-End Mode**) cannot be disabled, while the auxiliary input channel is optional.

Front-End Mode

Six front end modes are currently available:

- Single Mic
- Front-Mic OMNI
- Rear-Mic OMNI
- Two-Mic OMNI
- Static Directional
- Adaptive Directional Microphone (ADM)

Depending on the mode there are several controls available to adjust the algorithm accordingly.

Single Mic mode is intended to be used in devices with only one microphone or in devices with two microphones but perhaps deliberately chosen to have limited functionality in certain profiles (to minimize power consumption). In this mode only the front microphone signal is used and the rear microphone ADC is disabled to save power. While it is available in all other front-end modes, wind detection (part of the Environmental Classifier) is not available in Single Mic mode.

The default **Front-End Mode** is Front-Mic OMNI. In this mode, the front microphone signal is routed to the output of the hearing aid.

Rear-Mic OMNI mode is similar to the Front-Mic OMNI mode with the exception that it routes the rear microphone signal to the output.

The Two-Mic OMNI algorithm provides noise reduction by combining the rear and front microphones' signals in a constructive pattern. This type of noise-reduction is particularly useful when the noise at each microphone is independent (wind, microphone sensor noise).

Static Directional mode provides directional- sensitive noise reduction by steering a null toward a pre-determined direction. The **Directional Null Angle** parameter sets the angle at which the incident sound level is maximally attenuated.

ADM mode uses the same technique to reduce directional noise; however, rather than using a pre-determined null angle, it employs live adaptation to find the direction of the noise and steers the null toward that direction.

In Two-Mic OMNI, Static Directional and ADM modes it is critical to have magnitude and phase matched front and rear microphones. Small mismatches in the microphones can result in significant performance degradation. It is also critical to have the same pre-amplifier gain setting (**Front Mic Preamp** and **Rear Mic Preamp** parameters) selected to the front and rear microphones in these three modes.

Directional Null Angle

This parameter allows you to choose a null angle direction between 90 and 180 degrees. This is only used in the Static Directional mode. For example, the following pattern is formed for a null angle of 180° (Cardioid):

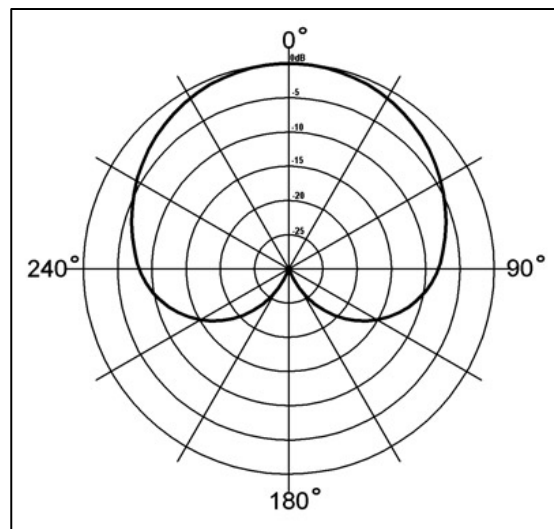


Figure 5. Example Polar Pattern for a Null Angle of 180°

Sound waves arriving from 0° (directly in front) are passed through without attenuation, and the ones arriving from 180° (directly behind) are attenuated accordingly.

The main polar patterns are defined by the following null angles:

- Figure 8: 90°
- Hyper Cardioid: 109°
- Super Cardioid: 125°
- Cardioid: 180°

Microphone Spacing Distance

The microphone port spacing is needed by the Two-Mic OMNI, Static Directional and ADM modes. The microphone spacing can be anything from 5 to 25 mm.

LFB Filter Enabled

This is the enable/disable control for the low-frequency boost filter. The filter is only used in the Static Directional and ADM modes; it does not filter the signal in other modes even if it is enabled. The boost is designed to compensate for the inherent low-frequency drop in the Static Directional and ADM algorithms to provide a flat response in the pass-band region.

LFB Cut-Off Frequency

This parameter allows adjustment of the cut-off frequency for the LFB filter. The frequency response plot is updated according to the chosen cut-off frequency when the LFB filter is enabled and the Front-End Mode is Static Directional or ADM.

Mic Swap

During manufacturing, the front and rear microphones may be accidentally soldered to the wrong pads. Enabling this parameter allows the routing of the microphone signals to be reversed to compensate for this. If the microphones are soldered to the correct pads, then the directional noise reduction patterns (otherwise known as polar patterns) will be reversed when the Mic Swap option is enabled.

Auxiliary Input Selection

As can be seen in Figure 1, two separate input signal paths (or input channels) exist in the firmware: the main microphone(s) input channel, and an auxiliary input channel. Currently the auxiliary channel can be set to one of:

- Wireless – the wireless signal is chosen as the auxiliary signal through the system
- None – no auxiliary signal is selected

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This is a profile-specific parameter, allowing wireless audio to be optionally mixed in with the microphone channel audio on a per-memory basis. Further, both input channels have an independent attenuation parameter, allowing adjustment of the ratio of microphone(s) signal level to auxiliary input (wireless audio) signal level.

Microphone and Auxiliary Input Attenuation

These parameters can be used to set the mixing ratio between the microphone and auxiliary input (wireless audio) input signals.

Pre- and Post-Biquad Filters

The HC module also contains pre- and post-biquad filtering. Four sets of coefficients for pre-biquad filters and four sets of coefficients for post-biquad filters are available for user configuration, enabling arbitrary frequency shaping of the main microphone signal before the WDRC block (via the pre-biquads), and the combined microphone and auxiliary signal just before the volume control (via the post-biquads).

For other parameters in the Front End and Hardware Control modules, refer to the *Firmware Bundle for Ezairo 7150 SL User Guide and Reference*.

WIDE DYNAMIC RANGE COMPRESSOR (WDRC)

The WDRC is a highly-configurable multi-channel (up to 8) compressor featuring adjustable crossover frequencies. Each channel features both lower- and upper-level kneepoints, an available expansion region, and in-channel AGCo. Three level detectors with independently- configurable attack and release times are available to adjust the dynamic performance of the algorithm in each channel.

Attack and Release Times

Attack and release times are based on the ANSI Specification of Hearing Aid Characteristics (ANSI/ASA S.22–2009).

Level Detectors

Each channel features in-channel AGCo, compression, and expansion level detectors with configurable attack and release times. Only one level detector takes priority and is used as the active signal level estimator in that channel at any given time, depending on the operating conditions described below:

- The in-channel AGCo level detector takes priority if it exceeds the in-channel output limit. In general, configure this level detector with the fastest attack time. By adjusting the attack time, you can apply harder or softer limiting
- The compressor level detector takes priority if it exceeds the expansion threshold and the in-channel AGCo level detector has not already taken priority
- The expansion level detector only takes priority if the in-channel AGCo level detector has not taken priority, and the compressor level detector is below the expansion threshold

Inter-Channel Effects

The WDRC makes use of the WOLA filterbank to maximize signal fidelity and minimize processing delay. As a result, some signal energy in the processing domain might appear in adjacent channels. This means that the configuration in one channel can have a subtle influence on the behavior of the neighboring channels. All of these effects are captured and predicted in the frequency response and input/output modelling.

Configuration Rules

The Ezairo Sound Designer Software prevents you from configuring the WDRC in a way that would violate any of the following rules:

- Crossover frequencies must be selected in ascending order
- The lower threshold cannot exceed the upper threshold
- If expansion is enabled, the expansion threshold cannot exceed the lower threshold
- The high level gain cannot exceed the low level gain
- The low level output cannot exceed the high level output. In other words, the sum of the lower threshold and low level gain cannot exceed the sum of the upper threshold and high level gain

ACOUSTIC FEEDBACK CANCELLER

The Feedback Canceller (FBC) is used to control acoustic feedback from the device, and comes with a number of adjustable parameters, including an optional gain management feature that allows the FBC to actively limit the gain in the forward path.

Startup Time

During the startup time of between 1 and 60 seconds, the FBC prioritizes feedback cancellation after power-up, to establish an initial feedback path estimate before, or while, the device is being inserted into the ear. The FBC operates at the fastest possible speed with the current gain management settings.

Adaptation Speeds

In general, there is a tradeoff between adaptation speed and entrainment performance: while a faster speed can ensure that feedback goes away quickly, the algorithm is more susceptible to entrainment in this case.

To minimize this tradeoff, the FBC features both active and idle speeds, where a lower number is a faster speed. The FBC's feedback detection mechanism allows it to run at idle speed when feedback is not detected to improve entrainment, switching to active speed when feedback is detected. There is a brief delay before the onset of feedback and switching to a faster speed.

Gain Management

Normally, the FBC has little power to manage feedback, under conditions where the forward gain is so high and/or the acoustic feedback path is so severe that it is beyond the Added Stable Gain (ASG). The gain management feature can improve entrainment, ASG, and feedback management under some conditions by actively limiting the forward gain in each band to no more than a specified amount above the estimated Maximum Stable Gain (MSG) in real-time in each frequency band.

Depending on how the gain management limit is set, it is possible to realize different performance benefits with different tradeoffs. The more negative the gain management limit, the more aggressively the FBC controls the forward gain of the device. At a minimum, it is recommended to set the gain management limit to a positive value below the pre-determined ASG of the FBC.

To maximize feedback control under changing feedback conditions, it might be desirable to set a negative gain management limit. This effectively limits the forward gain of the device to be less than the currently estimated MSG, creating a buffer where a change in feedback path is unable to generate feedback, because the conditions for feedback have been removed in the first place.

General Tuning Guidelines

The following sections describe the recommended tuning ranges and effects of key parameters.

Active Gain Management:

Active Gain Management can provide enhanced stability in difficult feedback situations. We recommend tuning this between 0 and 8 dB if feedback control is a priority over achieving the maximum possible ASG. The higher this setting is, the less effect active gain management will have, and the more the FBC will be relied upon for stability. A setting of 0 dB limits the forward gain of the device to the current estimated feedback level in each band. A setting of 8 dB limits the forward gain of the device to no more than 8 dB above the current estimated feedback level in each band. A setting less than 0 dB prevents the gain from approaching the estimated feedback limit.

Active Speed:

Active Speed is used when full-on feedback is present. In general, we recommend tuning this between 1 and 3. A lower number for active speed results in more rapid FBC adjustments and therefore better feedback control when full-on feedback is present. However, this may also lead to increased entrainment.

Active Sensitivity:

Active Sensitivity acts as a regulator between switching from idle to active speeds. A lower value means that the FBC is more sensitive to switching to active speed. In general, a value of 5 should provide good overall feedback control and entrainment performance. However, if the FBC is switching to active speed too frequently (increasing entrainment), it may be desirable to increase this value.

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Active Time:

Active Time is the time the FBC remains at active speed. In general, we recommend tuning this between 300 and 1500 ms. A shorter active time ensures that FBC is more robust to entrainment. However, discrete feedback events might be heard as the FBC switches between idle and active speeds in the presence of a slowly varying feedback path.

Idle Speed:

The **Idle Speed** parameter is used when full-on feedback is not present. In general, we recommend tuning this between 6 and 10. A lower number for idle speed results in more rapid FBC adjustments and therefore better feedback control when full-on feedback is not present. This may help to reduce or eliminate “tinny”-ness that can be heard at higher gain levels, and to minimize the need for the FBC to switch to active speed when subtle feedback path changes occur. However, this may also lead to increased entrainment.

For detailed information on the FBC algorithm and more in-depth tuning guidelines refer to [AND9660/D](#) – Ezairo Preconfigured Suite Adaptive Feedback Canceller.

WIDEBAND AGCO

Overview

The Wideband AGCo (W-AGCo) is primarily used to limit the acoustic output level of the device. This module features both an adjustable output limit, and wideband gain.

Output Limit

The **Output Limit** is a global parameter and defines the maximum acoustic output level at 1 kHz. Note that other frequencies might be limited above or below the output limit by an amount approximately equal to the difference in receiver sensitivity between those frequencies and 1 kHz. Signal levels might briefly exceed the output limit during sudden increases in signal level to preserve signal quality (see [Response Time](#) below).

Wideband Gain

The **Wideband Gain** is a profile-specific parameter that can be configured to provide gain or attenuation. However, the applied wideband gain will be limited for high level signals that would otherwise exceed the **Output Limit**, which always takes priority. The actual gain applied by the W-AGCo module is therefore always equal to or less than the wideband gain depending on the level of the signal being processed.

Response Time

The attack and release times of the W-AGCo have been set to approximately 1 and 32 ms, respectively, based on the standard 1/e formulas. This is to ensure safe limiting without compromising signal quality. If the output limit is set sufficiently low such that it is always engaging, pumping artifacts might occur.

NOISE REDUCTION

Overview

The Noise Reduction (NR) algorithm is a long term steady state noise reduction algorithm that analyzes the incoming audio signal and removes unwanted background noise while maintaining the desired audio signal. It is a frequency- domain algorithm and offers adjustment over frequency.

Maximum Attenuation

The actual noise reduction applied will change based on the input conditions. The maximum attenuation setting controls the amount by which the algorithm reduces the noise in worst case (poor SNR) conditions. It is adjustable over frequency, with a single, maximum attenuation setting of up to 15 dB for each of the 33 frequency bands. Higher maximum attenuation settings increase the risk of some impact to speech in some situations. The adjustment over frequency may be used to obtain the desired balance of speech naturalness and noise attenuation.

EQUALIZER

Overview

The 33-Band Equalizer applies user-defined gains to the 33 WOLA frequency bands. Gains are allocated based on user-defined configuration of equalizer channel crossover frequencies. The number of user-defined channels is variable. This module applies gains within the range $[-36, 0]$ dB. The Wideband AGCo applies any required positive wideband gain when this module is enabled, providing a total range of $[-18, 18]$ dB of gain in each frequency band. Due to WOLA filterbank sub-band interactions the gains applied to a given WOLA band are not exact; rather, it is a function of the gains applied to adjacent WOLA bands. These effects are reflected accurately in the modeled response of the system.

ENVIRONMENT CLASSIFIER

Overview

The Environment Classifier (EC) feature allows hearing aid operation using a single program memory that functions in any listening situation. When enabled, the Environment Classifier continually analyzes the acoustic environment and automatically adjusts the selected algorithms without the need for user intervention. The Environmental Classifier simplifies user controls while improving audibility, comfort and listening experience according to specific user preferences.

The Environmental Classifier's operation can be broken into two main stages: detection and memory management. The detection mechanism is responsible for determining the type of environment, while the memory management mechanism controls the process that applies the new parameter values in response to environment changes.

Detection

The detection mechanism can be enabled through assigning the **EC Activation Memory** parameter on the EC tab in Sound Designer to one of the available MMI-enabled memories. The assigned memory can be called the *automatic memory*. Once the automatic memory is assigned, the detection algorithm remains active at all times regardless of the memory the device is in.

Five separate environments are supported by the Environment Classifier module:

- Quiet
- Speech
- Noise
- Noisy Speech
- Wind

Parameters are available to help to tune the detection mechanism. For example, the **Speech Detection Sensitivity** parameter in combination with the **Quiet Threshold** parameter can impact how narrow or wide the noise, noisy-speech and speech environment detection ranges can be.

Refer to the *Firmware Bundle for Ezairo 7150 SL User Guide and Reference* for the full list of the Environmental Classifier parameters, and how each parameter can impact the algorithm.

Memory Management

While new environment classifications can happen regardless of the memory the device is in, the Environment Classifier only takes actions in response to classification changes when the device is in the automatic memory. There are two ways to enter or exit the automatic memory: a manual memory switch (e.g., pressing a pushbutton) or via the Bluetooth wireless interface. While in the automatic memory, upon detecting a new environment, the Environmental Classifier might switch from the current memory to a pre-programmed memory assigned to the detected environment.

It is important to note that the device would still officially remain in the automatic memory even though the new environment-mapped memory has been activated; in other words, you can consider the automatic memory to be the top level EC memory and the environment-mapped memories to be the sub-categories. There are no restrictions on how the memories can be programmed for each specific environment and any available user parameter can be adjusted as needed. Once the memories are programmed for each environment, they can be linked to their corresponding environments on the EC tab. For

example, if Memory F is programmed to be used in windy environments, the **Windy Environment Memory** field would be assigned to Memory F. Multiple environments can be assigned to the same memory if desired.

In addition to the environment-mapping parameters, another memory management related parameter called **Minimum Amount of Time Between Environments** enforces a minimum amount of time that must pass by before switching to another memory. This avoids a possible scenario where overly frequent memory switches can happen in a short span of time.

Cross-Functional Dependencies and Impacts

You can manually switch to only the memories that have been checked off as enabled (e.g., the MMI-enabled memories) on the Framework tab in the Sound Designer software. However, the Environment Classifier algorithm can switch to any available program memory on the device, regardless of whether it is accessible through the MMI. The **EC Activation Memory** field must always be assigned to an MMI-enabled memory to allow entry to the automatic memory.

Upon exiting the automatic memory, the MMI-enabled memory to which the device will switch, is relative to the automatic memory regardless of the environment-mapped memory at the time of the switch. For example, if Memories A through E are MMI-enabled, the automatic memory is Memory A, the device is in the automatic memory, and the current active environment-mapped memory is Memory D. Upon a memory switch button press, the device switches to Memory B rather than Memory E. Upon returning to the automatic memory, the device might immediately switch to the latest classified environment rather than the previous active environment-mapped memory at the time of exit. This is shown in Figure 6. In the previous example, if the device returns to the automatic memory and the latest classification is now linked to Memory C, the device immediately switches to Memory C rather than Memory B.

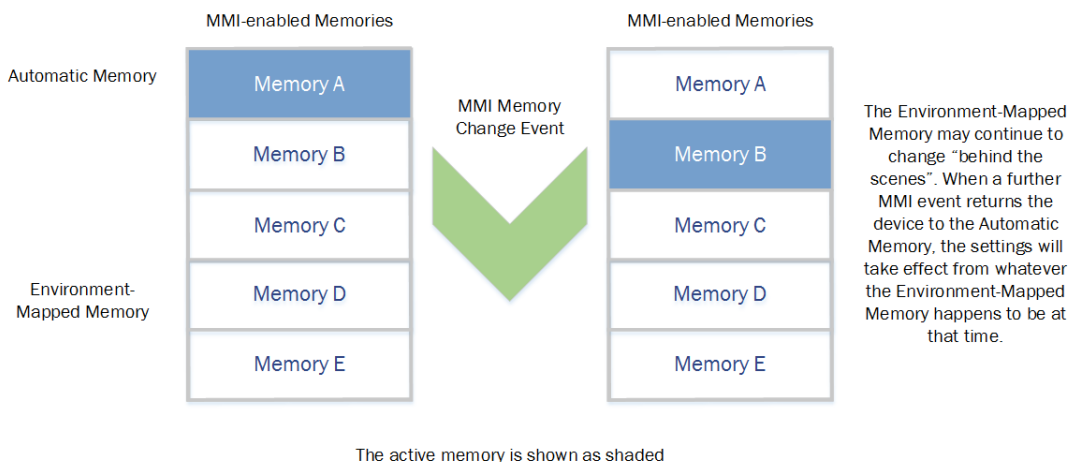


Figure 6. EC Memory Transitions Example

Live Monitoring

The state of the Environment Classifier can be tracked through the Environment Classifier live monitoring tool that is provided with the Sound Designer software:

Tentative Classification:

This field shows the active live classification that is continuously updated regardless of the MMI-enabled memory the device is in.

Established Classification:

This field is only updated when certain conditions are met after a new Tentative Classification is made and only when the device is in the automatic memory.

EC Memory:

If the device is in the automatic memory this field shows the currently enforced environment-mapped memory. If the device is not in the automatic memory, EC Memory shows the last active environment-mapped memory prior to the switch out of the automatic memory.

Additional Parameter Information

Refer to the *Firmware Bundle for Ezairo 7150 SL User Guide and Reference* for the full list of the Environmental Classifier parameters and how each parameter can impact the algorithm.

Single Microphone Operation

While two microphones are recommended, it is possible to use the Environment Classifier with a single microphone hearing aid. In this case, all profiles must use the *Single Mic* Front-End Mode, and the Wind Detection classification state will be unavailable.

CHAPTER 7

Volume Control and Memory Select

The Volume and Memory controls relate to both Man-Machine Interface (MMI) and command-based changes to volume and memory profile settings. The focus of this section will be those changes initiated from physical controls though much extends to other methods of control.

PHYSICAL CONTROLS (MMI)

The main method of physical user interaction to change the volume or memory profile is through push-buttons. These push-buttons are connected to the available DIOs on the hybrid pads.

CONFIGURATION OPTIONS

Many parameters are available to allow configuration of push-button functionality. Currently, all are found on the MMI tab in the Sound Designer Software. A subset of digital I/O pins (DIOs) can be used for changing volume or changing profile memory (increment or decrement in both cases). Users can control these DIOs with either a short press, a long press, or an even longer press (known as a super-long press). The durations of the long and super-long press are also configurable.

The available options for configuration are as follows:

- Event Type (volume up/down, memory up/down, enter pairing mode, clear bond table)
- Event I/O Pad (DIO 5, 6, 8, 9, 22, 23, 24, or 29)
- Event Trigger Type (short, long, super-long)
- Detection Edge (applies to all events)
- Long Press Duration
- Super-Long Press Duration

MEMORY/PROFILE SWITCHING

A total of eight memories (or profiles) are available. All but the first memory can be disabled individually via the Framework (FWK) module parameters. You can cycle through the memories in increasing or decreasing order by configuring the MMI event types appropriately.

VOLUME CONTROL

The volume control minimum, step size, and consequently its range are all programmable.

Note that the **Volume** parameter is a global parameter, whereas **Wideband AGCo Gain** is a profile-specific parameter and can be used for profile-specific wideband gain adjustments.

RETENTION OF SETTINGS

Any changes to the **Volume** or **Memory** parameter can be stored immediately in the EEPROM, allowing the hearing aid to start up with the same volume setting and in the same memory as when the power was turned off. This feature is disabled by default, but can be enabled using the **Volume Retention Enable** and **Memory Retention Enable** settings.

CHAPTER 8

Power Management

The Power Management module monitors the average battery voltage to disable features that consume a significant amount of current at specific voltage thresholds, and gradually power down the hearing aid in a controlled manner as the battery drains and eventually becomes unusable.

As the voltage falls, the power management module performs the following in this order:

1. At $1.1\text{ V} \pm 50\text{ mV}$, disables wireless
2. At $1.05\text{ V} \pm 50\text{ mV}$, disables any writes to non-volatile memory
3. At $0.95\text{ V} \pm 50\text{ mV}$, applies gain reductions periodically until the audio is muted

Disabling wireless means that you cannot communicate between the hearing device and a controlling device (i.e., a smartphone, wireless remote control or remote streamer).

Disabling writes to non-volatile memory includes disabling data logging, volume retention and memory retention. When the threshold is crossed for volume and memory retention, the device remembers the last setting before the voltage crossed the threshold.

If the average supply voltage drops below 0.95 V , the device reduces the gain by 1 dB every 10 seconds until either the average supply voltage rises above 0.95 V , or a total of 18 gain reductions have been applied, at which point the audio path is muted. If the average supply voltage returns to a level above 1.1 V , the audio path is first un-muted, if required. The gain is then increased by 1 dB every 10 seconds until either the average supply voltage drops below 1.1 V , or all gain reductions have been removed. No action is taken while the average supply voltage resides between 0.95 V and 1.1 V .

A low battery alert can be triggered periodically starting from a user configurable voltage threshold between 1 V and 1.2 V . This threshold is configured on the Diagnostics tab in the Sound Designer Software and determines the first time the low battery alert sounds; subsequently, it occurs every 5 minutes.

In order for the low battery alert to be played, ensure you configure a *Low Battery* acoustic indicator (see the next section).

CHAPTER 9

Acoustic Indicators

The Acoustic Indicators (AI) module configures either a Tone Sequence or a Voice Alert to play when any of the available events assigned to one of these indicators is triggered. A Tone Sequence is a sequence of up to four tones configurable via a series of parameters (some common to all tone sequences). A Voice Alert is an audio file stored in the EEPROM of the device as encoded audio data, and decoded and played in real-time when an event assigned to that Voice Alert is triggered.

The events assignable to an acoustic indicator are:

- Volume Up
- Volume Down
- Volume Set
- Volume Maximum
- Volume Minimum
- Memory A
- Memory B
- Memory C
- Memory D
- Memory E
- Memory F
- Memory G
- Memory H
- Low Battery
- Enter Pairing (Wireless)
- Delete Bonds (Wireless)
- Boot
- Error

CONFIGURING AN ACOUSTIC INDICATOR

Refer to the *Firmware Bundle for Ezairo 7150 SL User Guide and Reference*, the *Ezairo Sound Designer Software Development Kit (SDK) Programmer's Guide* and the *Ezairo Sound Designer User's Guide* for more information on configuring acoustic indicators.

AUDITIONING ACOUSTIC INDICATORS

The **Acoustic Indicator Audition** parameter can be used to trigger the acoustic indicator currently assigned to any event. This can be useful for testing an event's indicator, checking what is already configured on a device, or calibrating audio recording equipment levels by configuring an event to use a Tone Sequence of a desired frequency and amplitude.

CHAPTER 10

Data Logging and Non-volatile Storage

DATA LOGGER

Overview

The Data Logger module periodically writes information to EEPROM to track long-term device usage and diagnostic statistics. Two forms of data logging are used to track different types of information: long-term and short-term. Both forms of logging are enabled when data logging is enabled. The data logger has a configurable logging period, which is the time between logging events.

Long-Term Data Logging

In long-term data logging, counters are used to track the amount of time spent in each memory. Each memory has a counter, and each time the data logging period elapses, the counter associated with the current memory is incremented and the values updated in EEPROM. To determine the amount of time (in seconds) that the device has been in a memory, multiply the counter for the desired memory by the logging period. This form of logging captures a small amount of information, but over a very long period of time.

Short-Term Data Logging

In short-term data logging, device statistics and system events are encoded into records and written into a circular buffer in EEPROM. This form of logging captures detailed information, but over a shorter period of time due to the fixed length of the data buffer.

For details on the various device statistics that are logged in the short-term data log refer to the *Ezairo Sound Designer Software Development Kit (SDK) Programmer's Guide*.

SCRATCH MEMORY

Overview

The Scratch Memory (SM) module is a “data-only” module that provides an area to store any additional data (up to 512 bytes) that might be used to identify or configure the device, beyond what is provided in other modules. The data in this module has no effect on other algorithms and modules, but can be used by external software for the purposes listed above.

CHAPTER 11

Security

The Ezairo 7150 SL firmware bundle provides multiple levels of security, giving manufacturers the flexibility to select how much protection they would like to deploy into their hearing aid products.

RESTRICTED MODE AND ENCRYPTION

The Ezairo 7150 SL firmware bundle leaves the Ezairo 7100 DSP executing in Restricted Mode. When in this mode, the debug port of the device is effectively locked preventing any unwanted external access. In addition to Restricted Mode, all code in non-volatile memory has been encrypted using Level 1 encryption.

USER-SPECIFIED SECURITY

The Ezairo 7150 SL firmware bundle provides two additional methods of security, both of which are optional, and specified by the user. There is the option of locking any Product Library with a user-specified key, as well as locking all parameter access with a user-specified parameter access lock.

Product Library Key

When creating a Product Library, the user has the option of specifying a key that effectively acts as a password on the Product Library. Without this password it is impossible to open the Product Library and without access to a Product Library, interaction with the device via the SDK is impossible.

This password also prevents third parties from opening your Product Library inside Sound Designer and viewing the transducer models and parameter map files found inside.


Parameter Access Lock

Manufacturers can also employ the use of a parameter access lock. During the configuration stage of CalConfig, a 96-bit key can be set that effectively restricts the read/write access of all parameters within the device. If the device has been configured with this parameter lock key set, the device must be unlocked using the `UnlockParameterAccess()` method of the `CommunicationAdaptor` interface before you will be able to read or write any parameters from or to the device.

ADDITIONAL SECURITY OPTIONS

Manufacturers are free to write any other information they wish into the Scratch Memory area of the device. Scratch Memory can be used in concert with the Product Library Key and Parameter Access Lock to implement additional, proprietary security algorithms in fitting software.

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