RHYTHM™ R3920

User's Guide

AND9168/D
Rev. 1, May−2014

ON Semiconductor®

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CHAPTER 1

Introduction

PURPOSE

This guide provides an overview of the applications, software, and features of the RHYTHM™ R3920. This guide is intended to provide a high-level overview of the product, as well as where other resources relating to the product are available.

INTENDED AUDIENCE

This guide is for engineers, developers, and anyone else who requires technical information about the R3920.

CONVENTIONS

This manual does not use any special typographical conventions.

MANUAL ORGANIZATION

The RHYTHM R3920 User's Guide contains the following chapters and appendices:

- **Chapter 1: Introduction**, describes the purpose of the manual, outlines how the manual is organized, and lists suggested documents for more information
- **Chapter 2: Overview**, provides a high-level overview of the R3920 and its architecture
- **Chapter 3: Applications**, provides an overview of the features available on R3920
- **Chapter 4: Power Management**, describes the available power management schemes and functions of the R3920
- **Chapter 5: Volume Control and Switches**, describes the volume control and switch settings
- **Chapter 6: Software and Security**, provides an overview of the security features incorporated into the R3920 to protect the device against cloning and software piracy
- **Chapter 7: Input Connection and Layout Considerations**, provides wiring and layout recommendations, and strategies to minimize noise and interference

FURTHER READING

The following supporting documentation can be found at www.onsemi.com:

- Rhythm R3920 Datasheet
- ARK User's Guide
- Rhythm R3920 Parameters Information Note
- Impulse Noise Reduction Application Note
- Adaptive Feedback Cancellation for the Rhythm R3920 Application Note
- Automatic Directional Microphone Application Note
- iScene Detect Application Note
- iLog Datalogging Application Note
- Feedback Path Measurement Tool Application Note
- Biquad Filters in ON Semiconductor Preconfigured Digital Hybrids Application Note
- EVOKE Application Note
- Component Selection in Hearing Aids Application Note
- GU6701-E DSP Programmer Guide Application Note
- How to Store, Reflow and Solder ON Semiconductor Hybrids Application Note
- Hybrid Jig User's Guide Application Note
- Recommendations for Handling ON Semiconductor Hybrids Application Note

CHAPTER 2

Overview

INTRODUCTION

Rhythm R3920 is a preconfigured DSP designed for hearing aid applications that is based on the WOLVERINE™ multi-processor DSP platform. Wolverine is a 90 nm Silicon-on-Chip platform enabling design of highly-efficient and flexible hearing aid solutions. The multi-processor DSP system maximizes MIPS/W with a unique reconfigurable architecture. An integrated high-resolution dual ADC and a single DAC are included in one of the industry's smallest package sizes. Unmatched DSP processing capability and flexibility is offered in an ultra small footprint with industry-leading power consumption and easy integration into a wide range of hearing products.

The Wolverine DSP core implements all necessary advanced algorithms needed to develop a high-end hearing aid, including: WDRC compression, FrontWave® directional processing, programmable filters, adaptive algorithms, wideband gain, and MMI volume control. The adaptive algorithms include Impulse Noise Reduction, Adaptive Noise Reduction, Adaptive Feedback Cancellation and Automatic Adaptive Directional Microphones.

Impulse Noise Reduction (INR) is a new algorithm feature exclusive to the R3920. The INR algorithm will actively monitor the acoustic signal for loud impulsive noises in the environment such as slamming doors, dropped items, or cutlery rattling in a drawer. These noises can become uncomfortably or dangerously loud in a traditional hearing aid. The INR algorithm processes the signal to ensure that the sound at the output remains descriptive of the environment without being uncomfortably loud.

The Adaptive Noise Reduction reduces sustained audible noise in a low distortion manner while preserving perceived speech levels. The Adaptive Feedback Canceller reduces acoustic feedback while offering robust performance against pure tones. The Adaptive Directional Microphone algorithm automatically reduces the level of sound sources that originate from behind or from the side of the hearing-aid wearer without affecting sounds from the front.Additionally, the Automatic Adaptive Directional Microphones algorithm automatically reduces current by turning off the second input channel if it is not needed.

The iLog™ 4.0 Datalogging feature in R3920 records various parameters at programmable intervals (from 4s to 60 m) during use of the device. Once these parameter values are read from the device, they can be used to counsel the user and fine tune the fitting. iSceneDetect™ is the R3920's environmental classification algorithm that senses the users environment and automatically optimizes the hearing aid to maximize user comfort and audibility without any user interaction. The R3920 supports iSceneDetect in one-mic omni, static directional or adaptive directional modes.

R3920 comes with EVOKE™ advanced acoustic indicators. Evoke allows manufacturers to provide more complex, multi-frequency tones, in addition to traditional programmable tones for memory changes and low battery indication, which can simulate musical notes or chords for a more enjoyable and comfortable user experience.

Another new feature exclusive to the R3920 is the Automatic Receiver Detection (ARD) function. ARD is designed specifically for hearing aids with field replaceable receivers such as Receiver in Canal (RIC) devices. Via a software command in the R3920 API manufacturers can automatically detect which receiver is connected to the hearing aid, eliminating the possibility for an incorrect setup.

R3920 is equipped with a noise source that can be used in treating tinnitus. The Tinnitus Treatment noise can be shaped and attenuated and then summed into the audio path either before or after the volume control.

The Narrow-band Noise Stimulus feature allows the user to generate stimuli from the device that can be used for in situ audiometry. R3920 utilizes the power and capabilities of Wolverine to deliver advanced features and enhanced performance previously unavailable to a product in its class. As well, R3920 contains security features to protect clients' intellectual property against device cloning and software piracy.

SYSTEM OVERVIEW

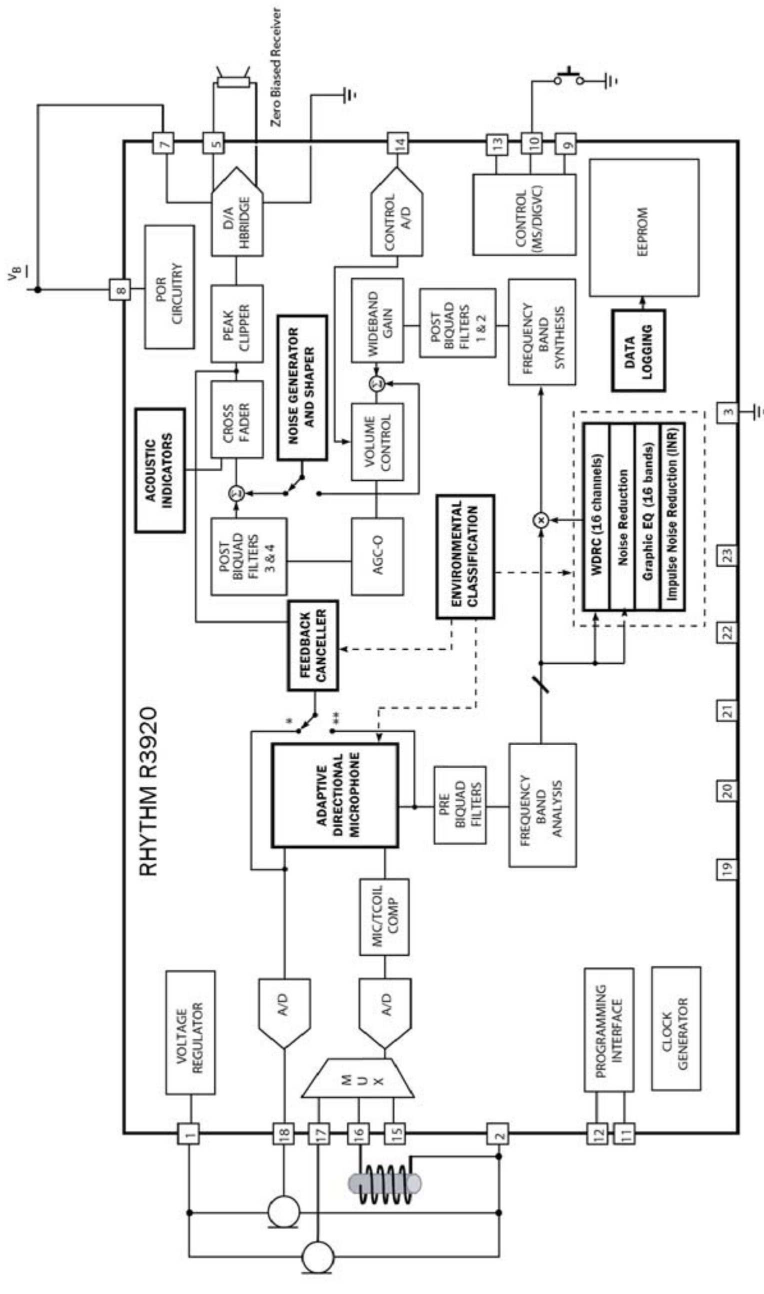

Figure 1. Hybrid Application Diagram

CHAPTER 3

Applications

16-CHANNEL WIDE DYNAMIC RANGE COMPRESSION (WDRC) CHANNEL PROCESSING

Figure 2 represents the I/O characteristic of each of the 16 independent AGC channels. The I/O curve can be divided into the following main regions:

- Low input level expansion (squelch) region
- Low input level linear region
- Compression region
- High input level linear region (return to linear)

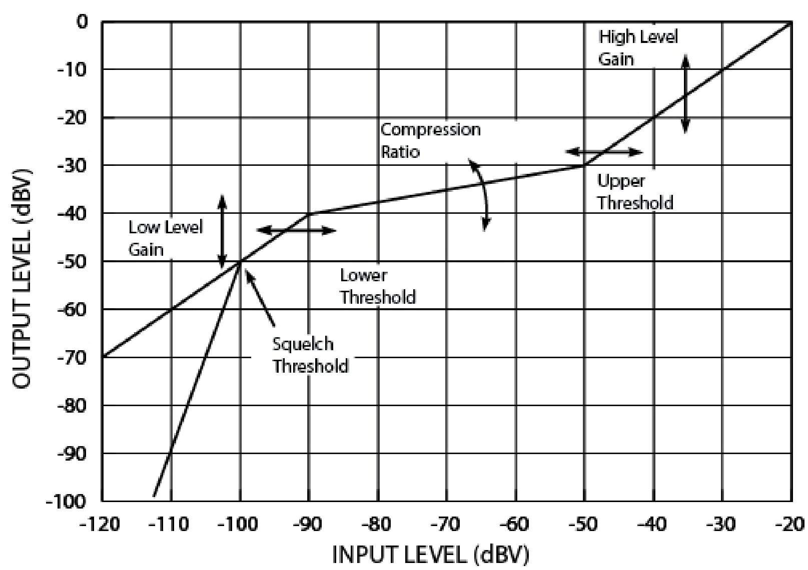

Figure 2. Independent Channel I/O Curve Flexibility

The I/O characteristic of the channel processing can be adjusted in the following ways:

- Squelch threshold (SQUELCH TH)
- Squelch expansion ratio
- Low level gain (LLGAIN)
- Lower threshold (LTH)
- High level gain (HLGAIN)
- Upper threshold (UTH)
- Compression ratio (CR)

To ensure that the I/O characteristics are continuous, it is necessary to limit adjustment to a maximum of four of the last five parameters. During Parameter Map creation, it is necessary to select four parameters as user adjustable, or fixed, and to allow one parameter to be calculated.

The squelch region within each channel implements a low level noise reduction scheme (1:2 or 1:3 expansion ratio) for listener comfort. This scheme operates in quiet listening environments (programmable threshold) to reduce the gain at very low levels. When the Squelch and AFC are both enabled it is highly recommended that the Squelch be turned on in all channels and that the Squelch thresholds be set above the microphone noise floor (see "Adaptive Feedback Canceller").

IMPULSE NOISE REDUCTION

Loud impulsive sounds in the environment such as slamming doors, dropped items or even cutlery rattling in a drawer can become uncomfortably loud in a traditional hearing aid. Hearing aids incorporating the Impulse Noise Reduction (INR) algorithm will actively monitor the acoustic signal for such impulsive sounds, and process the signal to ensure that the sound at the output remains descriptive of the environment without being uncomfortably loud.

The INR algorithm is specifically designed not to interfere with speech and other slow changing sounds, and to pass these audio signals transparently. The incoming audio signals are divided into 16 channels, each with their own center frequency and compression settings. The INR algorithm uses a select number of these input channels to further process the audio. Figure 3 and Figure 4 show a speech signal with multiple impulses; Figure 3 shows the processed signal with INR off while Figure 4 shows the processed signal with INR on, attenuating the loud impulsive sounds while keeping the remaining signal intact.

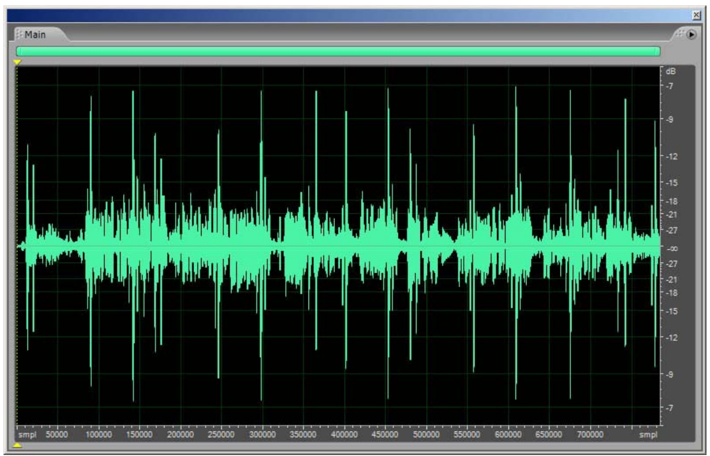

Figure 3. Audio Without INR

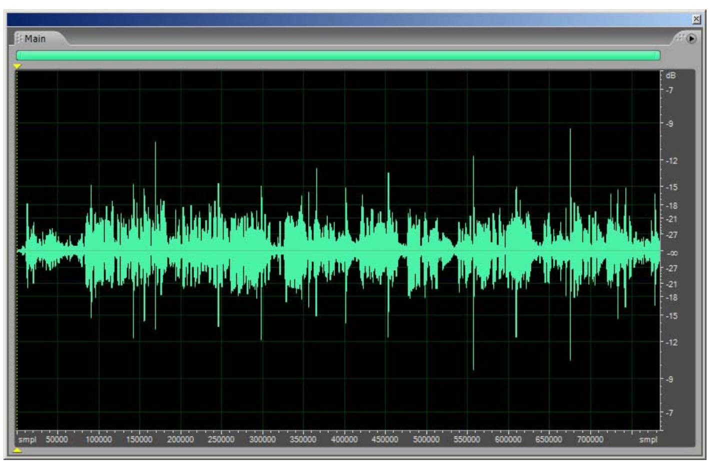

Figure 4. Audio With INR

As with all hearing aid algorithms, the INR algorithm requires parametric adjustment to accommodate different listening environments and hearing aid transducers. The adjustments are detailed in Table 1 below.

Table 1. AVAILABLE ADJUSTABLE PARAMETERS FOR THE INR ALGORITHM

Parameter	Description
Start Channel	The INR algorithm will be applied to this WDRC channel and all higher ones
Transient Threshold	The minimum input rate of change required for the INR to engage
Gain Profile Slope	The amount of reduction that will be applied to transients when the INR engages
INR WBGain Level	Adjusts how loud a signal must be before the algorithm decides whether it is a transient to reduce

ADAPTIVE NOISE REDUCTION

The noise reduction algorithm is built upon a high resolution 128-band filter bank enabling precise removal of noise. The algorithm monitors the signal and noise activities in these bands, and imposes a carefully calculated attenuation gain independently in each of the 128 bands.

The noise reduction gain applied to a given band is determined by a combination of three factors:

- Signal-to-Noise Ratio (SNR)
- Masking threshold
- Dynamics of the SNR per band

The SNR in each band determines the maximum amount of attenuation to be applied to the band - the poorer the SNR, the greater the amount of attenuation. Simultaneously, in each band, the masking threshold variations resulting from the energy in other adjacent bands is taken into account. Finally, the noise reduction gain is also adjusted to take advantage of the natural masking of ‘noisy’ bands by adjacent speech bands over time.

Based on this approach, only enough attenuation is applied to bring the energy in each ‘noisy’ band to just below the masking threshold. This prevents excessive amounts of attenuation from being applied and thereby reduces unwanted artifacts and audio distortion. The Noise Reduction algorithm efficiently removes a wide variety of types of noise, while retaining natural speech quality and level. The level of noise reduction (aggressiveness) is configurable to 3, 6, 9 and 12 dB of reduction.

ADAPTIVE FEEDBACK CANCELLER

The Adaptive Feedback Canceller reduces acoustic feedback by forming an estimate of the hearing aid feedback signal and then subtracting this estimate from the hearing aid input. The forward path of the hearing aid is not affected. Unlike adaptive notch filter approaches, the AFC in R3920 does not reduce the hearing aid’s gain. The AFC is based on a time-domain model of the feedback path.

Additional flexibility exclusive to the R3920 allows manufacturers to tune the algorithm for maximum performance. Adjustments can be made to the algorithms adaptation speed in order to adjust the speed at which the algorithm reacts to detected feedback signals.

The third-generation AFC (see: Figure 5) allows for an increase in the stable gain of the hearing aid while minimizing artefacts for music and tonal input signals. As with previous products, the feedback canceller provides completely automatic operation.

NOTE: Added stable gain will vary based on hearing aid style and acoustic setup. Please refer to the Adaptive Feedback Cancellation note for more details

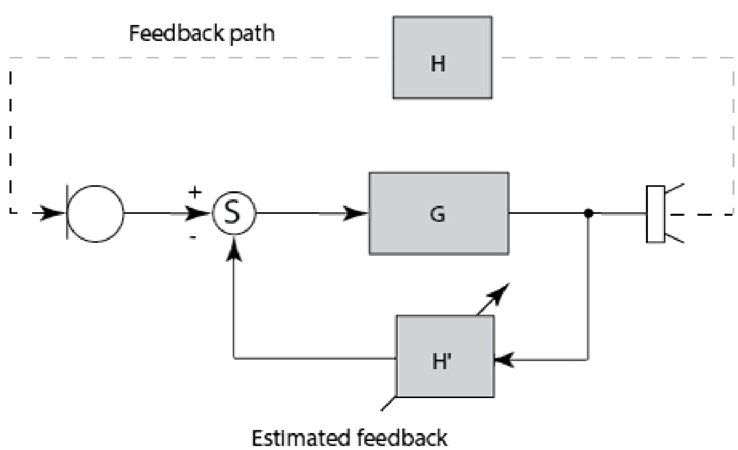

Figure 5. Adaptive Feedback Canceller (AFC) Block Diagram

AUTOMATIC RECEIVER DETECTION (ARD)

An automatic receiver detection (ARD) feature has been implemented on the R3920 to be used for devices with removable and replaceable receivers in the field, including Receiver in the Canal (RIC) hearing aids. ARD utilizes ground and ID pins embedded within a receiver link and connected to the hybrid, to code the replaceable receivers in such as way as to allow for 5 unique receiver types (including none). Through a command generated within the software (Read Detected Receiver in the API), the R3920 can detect which receiver code is connected. The software can then make the necessary changes to ensure proper configuration within the hearing aid.

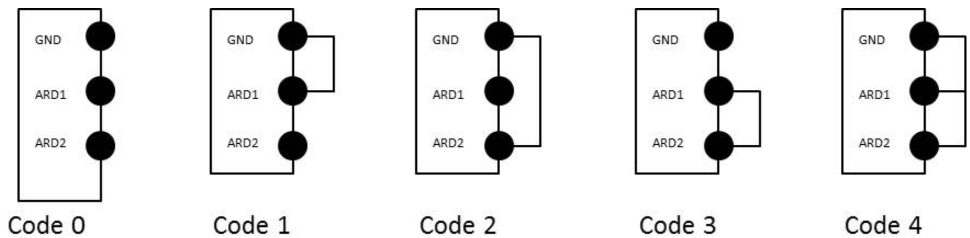

Figure 6. Wiring Diagrams for Automatic Receiver Detection

DIRECTIONAL MICROPHONES

In any directional mode, the circuitry includes a fixed filter for compensating the sensitivity differences between microphones. The filter parameters are adjusted during product calibration.

A dedicated biquad filter following the directional block has been allocated for low frequency equalization to compensate for the 6 dB/octave roll-off in frequency response that occurs in directional mode. The amount of low frequency equalization that is applied is programmable.

ON Semiconductor recommends using matched microphones. The maximum spacing between the front and rear microphones cannot exceed 20 mm (0.787 in).

Adaptive Directional Microphones

The Adaptive Directional Microphone (ADM) algorithm is a two-microphone processing scheme for hearing aids. It is designed to automatically reduce the level of sound sources that originate from behind or the side of the hearing-aid wearer without affecting sounds from the front. The algorithm accomplishes this by adjusting the null in the microphone polar pattern to minimize the noise level at the output of the ADM. The discrimination between desired signal and noise is based on the direction of arrival with respect to the hearing aid: sounds from the front hemisphere are passed unattenuated whereas sounds arriving from the rear hemisphere are reduced.

The angular location of the null in the microphone polar pattern is continuously variable over a range of 90 to 180 degrees where 0 degrees represents the front.

The location of the null in the microphone pattern is influenced by the nature of the acoustic signals (spectral content, direction of arrival) as well as the acoustical characteristics of the room. The ADM algorithm steers a single, broadband null to a location that minimizes the output noise power. If a specific noise signal has frequency components that are dominant, then these will have a larger influence on the null location than a weaker signal at a different location. In addition, the position of the null is affected by acoustic reflections. The presence of an acoustic reflection may cause a noise source to appear as if it originates at a location other than the true location. In this case, the ADM algorithm chooses a compromise null location that minimizes the level of noise at the ADM output.

Automatic Adaptive Directional Microphones

When Automatic ADM mode is selected, the adaptive directional microphone remains enabled as long as the ambient sound level is above a specific threshold and the directional microphone has not converged to an omni-directional polar pattern. On the other hand, if the ambient sound level is below a specific threshold, or if the directional microphone has converged to an omni-directional polar pattern, then the algorithm will switch to single microphone, omni-directional state to reduce current consumption. While in this omni-directional state, the algorithm will periodically check for conditions warranting the enabling of the adaptive directional microphone.

FrontWave Directionality

The FrontWave block provides the resources necessary to implement directional microphone processing. The block accepts inputs from both a front and rear microphone and provides a synthesized directional microphone signal as its output. The directional microphone output is obtained by delaying the rear microphone signal and subtracting it from the front microphone signal. Various microphone response patterns can be obtained by adjusting the time delay.

TINNITUS TREATMENT

R3920 has an internal white noise generator that can be used for Tinnitus Treatment. The noise can be attenuated to a level that will either mask or draw attenuation away from the user's tinnitus. The noise can also be shaped using low-pass and/or high-pass filters with adjustable slopes and corner frequencies.

As shown in Figure 1, the Tinnitus Treatment noise can be injected into the signal path either before or after the volume control (VC) or it can be disabled. If the noise is injected before the VC then the level of the noise will change along with the rest of the audio through the device when the VC is adjusted. If the noise is injected after the VC then it is not affected by VC changes.

The Tinnitus Treatment noise can be used on its own without the main audio path in a very low power mode by selecting the Tinnitus Treatment noise only. This is beneficial either when amplification is not needed at all by a user or if the user would benefit from having the noise supplied to them during times when they do not need acoustic cues but their sub-conscious is still active, such as when they are asleep.

The ARK software has a Tinnitus Treatment tool that can be used to explore the noise shaping options of this feature. This tool can also be easily incorporated into another software application.

If the noise is injected before the VC and the audio path is also enabled, the device can be set up to either have both the audio path and noise adjust via the VC or to have the noise only adjust via the VC. If the noise is injected after the VC, it is not affected by VC changes (see Table 2).

Table 2. NOISE INJECTION EFFECT ON VOLUME CONTROL

Noise Insertion Mode	VC Controls	Noise Injected
Off	Audio	Off
Pre VC	Audio + Noise	Pre VC
Post VC	Audio	Post VC
Noise Only Pre VC	Noise	Pre VC
Noise Only Post VC	-	Post VC
Pre VC with Noise	Noise	Pre VC

NARROW-BAND NOISE STIMULUS

R3920 is capable of producing Narrow-band Noise Stimuli that can be used for in situ audiometry. Each narrow-band noise is centered on an audiometric frequency. The duration of the stimuli is adjustable and the level of the stimuli are individually adjustable.

IN-SITU DATALOGGING – ILOG 4.0

R3920 has a datalogging function that records information every 4 seconds to 60 minutes (programmable) about the state of the hearing aid and its environment to non-volatile memory. The function can be enabled with the ARK software and information collection will begin the next time the hybrid is powered up. This information is recorded over time and can be downloaded for analysis.

The following parameters are sampled:

- Battery level
- Volume control setting
- Program memory selection
- Environment
- Ambient sound level
- Length of time the hearing aid was powered on

The information is recorded using two methods in parallel:

Short-term method: A circular buffer is serially filled with entries that record the state of the first five of the above variables at the configured time interval.

Long-term method: Increments a counter based on the memory state at the same time interval as that of short-term method. Based on the value stored in the counter, length of time the hearing aid was powered on can be calculated

There are 750 log entries plus 6 memory select counters which are all protected using a checksum verification. A new log entry is made whenever there is a change in memory state, volume control, or battery level state. A new log entry can also be optionally made when the environmental sound level changes more than the programmed threshold, thus it is possible to log only significantly large changes in the environmental level, or not log them at all.

The ARK software iLog graph displays the iLog data graphically in a way that can be interpreted to counsel the user and fine tune the fitting. This iLog graph can be easily incorporated into other applications or the underlying data can be accessed to be used in a custom display of the information.

SIGNAL PATH

There are two main audio input signal paths. The first contains the front microphone and the second contains one of the rear microphone, telecoil or direct audio input, as selected by a programmable MUX. The front microphone input is intended as the main microphone audio input for single microphone applications.

Analog input signals should be ground referenced to MGND (microphones, telecoils, DAI). MGND is internally connected to GND to minimize noise, and should not be connected to any external ground point.

In iSceneDetect, FrontWave, ADM or Automatic ADM operation, the incoming signal from the two microphones is used to produce a directional hearing aid response. The two audio inputs are buffered, sampled and converted into digital form using dual A/D converters. The digital outputs are converted into a 32 kHz or 16 kHz, 20-bit digital audio signal. Further IIR filter blocks process the front microphone and rear microphone signals. One biquad filter is used to match the rear microphone's gain to that of the front microphone. Next, other filtering is used to provide an adjustable group delay to create the desired polar response pattern during the calibration process. In iSceneDetect, ADM and Automatic ADM, the two microphone inputs are combined in an adaptive way while in FrontWave operation the combination is static.

In the telecoil mode gains are trimmed during Cal/Config process to compensate for microphone/telecoil mismatches.

The FrontWave block is followed by four cascaded biquad filters: pre1, pre2, pre3 and pre4. These filters can be used for frequency response shaping before the signal goes through channel and adaptive processing.

The channel and adaptive processing consists of the following:

- Frequency band analysis
- 16-channel WDRC
- 16 frequency-shaping bands (spaced linearly at 500 Hz intervals, except for first and last bands)
- 128 frequency band adaptive noise reduction
- Impulse Noise Reduction
- Frequency band synthesis

After the processing the signal goes through two more biquad filters, post1 and post2, which are followed by the AGC-O block. The AGC-O block incorporates the wideband gain and the volume control. There are also two more biquad filters, post3 and post4, and the peak clipper. The last stage in the signal path is the D/A H-bridge.

White noise for tinnitus masking can be shaped, attenuated and then added into the signal path at two possible locations: before the volume control (between the wideband gain and the volume control) or after the volume control (between post 4 and the peak clipper) as shown in Figure 1. This can be used as a pure tinnitus masker or as part of a complete hearing aid solution.

iSceneDetect ENVIRONMENTAL CLASSIFICATION

When enabled, the iSceneDetect feature will sense the environment and automatically control the enhancement algorithms without any user involvement. It can detect up to six environments including: speech in quiet, speech in noise, wind, music, quiet and noise environments, and make the necessary adjustments to the parameters in the audio path, such as ADM, ANR, WDR, BC, and FBC in order to optimize the hearing aid settings for the specific environment.

iSceneDetect makes these adjustments gradually so the change in settings is smooth and virtually unnoticeable. This feature will enable the hearing aid wearer to have an aid which will work in any environment with a single memory. iSceneDetect enables manufacturers to build a fully-featured hearing aid that performs well in any environment, without the need of a memory select push button and with only one memory.

EVOKE ADVANCE ACOUSTIC INDICATORS

Advanced acoustic indicators provide alerting sounds that are more complex, more pleasing and more meaningful to the end user than the simple tones used on previous products. The feature is capable of providing pulsed, multi-frequency pure tones with smooth on and off transitions and also damped, multi-frequency tones that can simulate musical notes or chords.

A unique indicator sound can be assigned to each of the ten system events: memory select (A, B, C, D, E or F), low battery warning, digital VC movement, digital VC minimum/maximum, and startup. Each sound can consist of a number of either pure tones or damped tones but not both.

A pure tone sound can consist of up to four tones, each with a separate frequency, amplitude, duration and start time. Each frequency component is smoothly faded in and out with a fade time of 64 ms. The start time indicates the beginning of the fade in. The duration includes the initial fade-in period. By manipulating the frequencies, start times, durations and amplitudes various types of sounds can be obtained (e.g., various signalling tones in the public switched telephone network).

A damped tone sound can consist of up to six tones, each with a separate frequency, amplitude, duration, start time and decay time. Each frequency component starts with a sudden onset and then decays according to the specified time constant. This gives the audible impression of a chime or ring. By manipulating the frequencies, start times, durations, decays and amplitudes, various musical melodies can be obtained.

Acoustic indication can be used without the need to completely fade out the audio path. For example, the low-battery indicator can be played out while the user still hears an attenuated version of the conversation.

FEEDBACK PATH MEASUREMENT TOOL

The feedback path measurement tool uses the onboard feedback cancellation algorithm and noise generator to measure the acoustic feedback path of the device. The noise generator is used to create an acoustic output signal from the hearing aid, some of which leaks back to the microphone via the feedback path. The feedback canceller algorithm automatically calculates the feedback path impulse response by analyzing the input and output signals. Following a suitable adaptation period, the feedback canceller coefficients can be read out of the device and used as an estimate of the feedback-path impulse response.

This tool is an excellent resource for mechanical designers to help determine the source of electrical, acoustic and mechanical feedback. Allowing for quick diagnosis of potential feedback issues in a particular design, the tool can also be incorporated into the manufacturers fitting software in order to allow audiologists and dispensers a method to evaluate the fit of a hearing aid on the patient.

A/D AND D/A CONVERTERS

The system's two A/D converters are second order sigma-delta modulators operating at a 2.048 MHz sample rate. The system's two audio inputs are pre-conditioned with antialias filtering and programmable gain pre-amplifiers. These analog outputs are over-sampled and modulated to produce two, 1-bit Pulse Density Modulated (PDM) data streams. The digital PDM data is then decimated down to Pulse-Code Modulated (PCM) digital words at the system sampling rate of 32 kHz.

The D/A is comprised of a digital, third order sigma-delta modulator and an H-bridge. The modulator accepts PCM audio data from the DSP path and converts it into a 64-times or 128-times over-sampled, 1-bit PDM data stream, which is then supplied to the H-bridge. The H-bridge is a specialized CMOS output driver used to convert the 1-bit data stream into a low-impedance, differential output voltage waveform suitable for driving zero-biased hearing aid receivers.

HRX HEAD ROOM EXPANDER

R3920 has an enhanced Head Room Extension (HRX™) circuit that increases the input dynamic range of R3920 without any audible artifacts. This is accomplished by dynamically adjusting the pre-amplifier's gain and the post-A/D attenuation depending on the input level.It is recommended that HRX be left on in all situations.

TELECOIL PATH

The telecoil input can be on independently or summed together with the front microphone.The telecoil gains are calibrated during the Cal/Config process. To compensate for the telecoil/microphone frequency response mismatch, a first order filter with 500 Hz corner frequency is implemented. Through ARKonline®, it is possible to implement a telecoil compensation filter with an adjustable corner frequency. To accommodate for the gain mismatch, the telecoil gain is adjusted to match the microphone gain at 500 Hz or 1 kHz (default) and is selectable in ARKonline.

There is also a telecoil gain adjustment parameter that can be enabled in ARKonline and set in the Interactive Data Sheet (IDS), enabling manual adjustment of the telecoil gain compensation.

AUTOMATIC TELECOIL

R3920 is equipped with an automatic telecoil feature, which allows the hybrid to switch to a specific memory upon the closing of a switch connected to MS2. This feature is useful when MS2 is connected to a magnetic field-sensing switch, such as a reed switch, that changes state depending on the presence of a static magnetic field like those found in common telephones. The last valid memory (memory F in a 6 memory device) can be programmed to be the telecoil or mic + telecoil memory. This is done so that, when a telephone handset is placed near the hearing aid, its static magnetic field closes the switch and causes the hybrid to change to the last memory with the telecoil input enabled. However, it is possible that the hearing aid wearer may move his or her head away from the telephone handset momentarily, in which case, it is undesirable to immediately change out of telecoil mode and then back in moments later. To prevent this needless switching, the R3920 has a debounce circuit implemented.

This debounce circuit delays the device from switching out of the last memory when MS2 is configured as a static switch in 'D-only' mode. The debounce time is programmable to be 1.5, 3.5 or 5.5 seconds after the switch opens (i.e., the handset is moved away from the hearing aid) or this feature can be disabled.

DAI PATH

The DAI input can be adjusted using a first order filter with a variable corner frequency similar to the telecoil compensation filter. Through ARKonline, it is possible to implement this DAI filter to set either a static or adjustable corner frequency.

The Mic plus DAI mode mixes the Mic1 and DAI signals. The Mic1 input signal is attenuated by 0, -6 or -12 dB before being added to the DAI input signal. The DAI input also has gain adjustment in 1 dB steps to assist in matching it to the Mic1 input level.

GRAPHIC EQUALIZER

R3920 has a 16-band graphic equalizer. The bands are spaced linearly at 500 Hz intervals, except for the first and the last band, and each one provides up to 24 dB of gain adjustment in 1 dB increments.

BIQUAD FILTERS

Additional frequency shaping can be achieved by configuring generic biquad filters. The transfer function for each of the biquad filters is as follows:

$$H(z) \equiv \frac{b0 + b1 \times z^{-1} + b2 \times z^{-2}}{1 + a1 \times z^{-1} + a2 \times z^{-2}}$$

Note that the a0 coefficient is hard-wired to always be 1. The coefficients are each 16 bits in length and include one sign bit, one bit to the left of the decimal point, and 14 bits to the right of the decimal point. Thus, before quantization, the floating point coefficients must be in the range 2.0 x < 2.0 and quantized with the function:

$$\mathrm{round}\left(x \times 2^{14}\right)$$

After designing a filter, the quantized coefficients can be entered into the PreBiquads or PostBiquads tab in the Interactive Data Sheet. The coefficients b0, b1, b2, a1, and a2 are as defined in the transfer function above. The parameters meta0 and meta1 do not have any effect on the signal processing, but can be used to store additional information related to the associated biquad.

The underlying code in the product components automatically checks all of the filters in the system for stability (i.e., the poles have to be within the unit circle) before updating the graphs on the screen or programming the coefficients into the hybrid. If the Interactive Data Sheet receives an exception from the underlying stability checking code, it automatically disables the biquad being modified and display a warning message. When the filter is made stable again, it can be re-enabled.

Also note that in some configurations, some of these filters may be used by the product component for microphone/telecoil compensation, low-frequency EQ, etc. If this is the case, the coefficients entered by the user into IDS are ignored and the filter designed by the software is programmed instead. For more information on filter design refer to the Biquad Filters in ON Semiconductor Preconfigured Digital Hybrids Application Note application note.

AGC-O AND PEAK CLIPPER

The output compression limiting block (AGC-O) is an output limiting circuit whose compression ratio is fixed at: 1. The threshold level is programmable. The AGC-O module has programmable attack and release time constants.

The AGC-O on the R3920 has optional adaptive release functionality. When this function is enabled, the release time varies depending on the environment. In general terms, the release time becomes faster in environments where the average level is well below the threshold and only brief intermittent transients exceed the threshold.

Conversely, in environments where the average level is close to the AGC-O threshold, the release time applied to portions of the signal exceeding the threshold is longer. The result is an effective low distortion output limiter that clamps down very quickly on momentary transients but reacts more smoothly in loud environments to minimize compression pumping artifacts. The programmed release time is the longest release time applied, while the fastest release time is 16 times faster. For example, if a release time of 128 ms is selected, the fastest release time applied by the AGC-O block is 8 ms.

R3920 also includes the Peak Clipper block for added flexibility.

MEMORY SWITCH FADER

To minimize potential loud transients when switching between memories, R3920 uses a memory switch fader block. When the memory is changed, the audio signal is faded out, followed by the memory select acoustic indicators (if enabled), and after switching to the next memory, the audio signal is faded back in. The memory switch fader is also used when turning the Tone Generator on or off, and during SDA programming.

CHAPTER 4

Power Management

R3920 has three user-selectable power management schemes to ensure the hearing aid turns off gracefully at the end of battery life. shallow reset, deep reset and advanced reset mode. It also contains a programmable power on reset delay function.

POWER ON RESET DELAY

The programmable POR delay controls the amount of time between power being connected to the hybrid and the audio output being enabled. This gives the user time to properly insert the hearing aid before the audio starts, avoiding the temporary feedback that can occur while the device is being inserted. During the delay period, momentary button presses are ignored.

NOTE: The values set in IDS are relative values from to 11 seconds; not absolute. The POR delay is relative to the configuration loaded on the Wolverine platform.

LOW BATTERY NOTIFICATION

Notification of the low battery condition via an acoustic indicator is optionally performed when the battery voltage drops below a configurable low battery notification threshold. The low battery indicator is repeated every five minutes until the device shuts down.

POWER MANAGEMENT FUNCTIONALITY

As the voltage on the hearing aid battery decreases, an audible warning is given to the user indicating the battery life is low. In addition to this audible warning, the hearing aid takes other steps to ensure proper operation given the weak supply. The exact hearing aid behavior in low supply conditions depends on the selected POR mode. The hearing aid has three POR modes:

- Shallow Reset Mode
- Deep Reset Mode
- Advanced Mode

Shallow Reset Mode

In shallow reset mode, the hearing aid will operate normally when the battery is above 0.95 V. Once the supply voltage drops below 0.95 V the audio will be muted and remain in that state until the supply voltage rises above 1.1 V. Once the supply voltage drops below the control logic ramp down voltage, the device will undergo a hardware reset. At this point, the device will remain off until the supply voltage returns to 1.1 V. When the supply voltage is below the control logic voltage, but above 0.6 V and rises above the 1.1 V turn on threshold, the device will activate its output and operate from the memory that was active prior to reset. If the supply voltage drops below 0.6 V, and rises above the 1.1 V turn on threshold, the device will re-initialize, activate its output and operate from memory A.

Deep Reset Mode

In deep reset mode, the hearing aid will operate normally when the battery is above 0.95 V. Once the supply voltage drops below 0.95 V the audio will be muted. The device remains in this state until the supply voltage drops below the hardware reset voltage of 0.6 V. When this occurs, the device will load memory A and operate normally after the supply voltage goes above 1.1 V.

Advanced Reset Mode

Advanced reset mode on R3920is a more sophisticated power management scheme than shallow and deep reset modes. This mode attempts to maximize the device's usable battery life by reducing the gain to stabilize the supply based on the instantaneous and average supply voltage levels. Instantaneous supply fluctuations below 0.95 V can trigger up to two 3 dB, instantaneous gain reductions. Average supply drops below 0.95 V can trigger up to eighteen, 1 dB average gain reductions.

While the average supply voltage is above 0.95 V, an instantaneous supply voltage fluctuation below 0.95 V will trigger an immediate 3 dB gain reduction. After the 3 dB gain reduction has been applied, the advanced reset model holds off checking the instantaneous voltage level for a monitoring period of 30 second in order to allow the voltage level to stabilize. If after the stabilization time the instantaneous voltage drops a second time below 0.95 V during the next monitoring period, the gain will be reduced an additional 3 dB for a 6 dB total reduction and a 30 second stabilization time is activated. The advanced reset mode continues to monitor the instantaneous voltage levels over 30 second monitoring periods. If the instantaneous voltage remains above 1.1 V during that monitoring period, the gain will be restored to the original setting regardless of whether one or two gain reductions are applied. If two gain reductions are applied and the instantaneous voltage level remains above 1.0 V for a monitoring period, the gain will be restored to a 3 dB reduction.

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

NOTE: Instantaneous and average gain reductions are adjusted independently

When the instantaneous voltage falls below the hardware shutdown voltage, the device will undergo a hardware reset. When it turns back on because the voltage has risen above the turn-on threshold, it will behave the same as it would in shallow reset mode.

POWER SUPPLY CONSIDERATIONS

R3920 was designed to accommodate high power applications. AC ripple on the supply can cause instantaneous reduction of the battery's voltage, potentially disrupting the circuit's function. R3920 hybrids have a separate power supply and ground connections for the output stage. This enables hearing aid designers to accommodate external RC filters to minimize any AC ripple from the supply line. Reducing this AC ripple greatly improves the stability of the circuit and prevents unwanted reset of the circuit caused by spikes on the supply line.

For more information on properly designing a filter to reduce supply ripple, refer to the Using DSP Hybrids in High Power Applications Initial Design Tips information note.

CHAPTER 5

Volume Control and Switches

EXTERNAL VOLUME CONTROL

The volume of the device can either be set statically via software or controlled externally via a physical interface. R3920 supports both analog and digital volume control functionality, although only one can be enabled at a time. Digital control is supported with either a momentary switch or a rocker switch. In the latter case, the rocker switch can also be used to control memory selects.

Table 3. MS SWITCH MODES

MS Switch Mode	MS1 Switch	MS2 Switch	Max # of Value Memories	Donly	MSSMode	Use
Mode 1	Momentary	None	6	Off	Momentary	Simplest Configuration
Mode 2	Momentary	Static	6	On	Momentary	Jump to last memory
Mode 3	Static	Static	4	Off	Static	Binary selection of memory
Mode 4	Static	Static	3	On	Static	Jump to last memory

The flexibility of the MS switches is further increased by allowing the MS switches to be wired to GND or VBAT, corresponding to an active low or active high logic level on the MS pins. This option is configured with the MSPullUpDown/MS2PullUpDown setting in the IDS settings tab as shown in Table 4 below.

Table 4. MS SWITCH LOGIC LEVELS VS. IDS PULLUPDOWN SETTINGS

PullUpDown Settings in IDS	MS Switch State	MS Input Logic Level	Switch Connection
Pulldown	CLOSED	HI	TO VBAT
Pulldown	OPEN	LOW	TO VBAT
Pullup	CLOSED	LOW	TO GND
Pullup	OPEN	HI	TO GND

In the following mode descriptions, it is assumed that the PullUpDown setting has been properly configured for the MS switch wiring so that a CLOSED switch state is at the correct input logic level.

Mode 1: Momentary Switch on MS1

This mode uses a single momentary switch on MS1 (Pin10) to change memories. When using this mode the part starts in memory A, and whenever the button is pressed, the next valid memory is loaded. When the user is in the last valid memory, a button press causes memory A to be loaded.

This mode is set by programming the ‘MSSMode’ parameter to ‘Momentary’ and ‘Donly’ to ‘disabled’.

Mode 1 Example:

If 6 valid memories: ABCDEFABCDEF…
If 5 valid memories: ABCDEABCDE…
If 4 valid memories: ABCDABCDA…
If 3 valid memories: ABCABCA…
If 2 valid memories: ABABA…
If 1 valid memory: AAA…

Mode 2: Momentary Switch on MS1, Static Switch on MS2 (Jump to Last Memory)

This mode uses a static switch on MS2 (Pin 9) and a momentary switch on MS1 (Pin 10) to change memories. If the static switch is OPEN, the part starts in memory A and behaves like momentary, with the exception that the highest valid memory (F if 6 memories selected) is not used. If the static switch on MS2 is set to CLOSED, the part automatically jumps to the highest valid memory location (occurs on startup or during normal operation). In this setup, the momentary switch's state is ignored, preventing memory select beeps from occurring. When MS2 is set to OPEN, the part loads in the memory location selected before MS2 was closed.

This mode is set by programming the 'MSSMode' parameter to 'Momentary' and 'Donly' to 'enabled'.

Mode 2 Example:

If MS2 = OPEN and there are 6 valid memories: ABCEFABCEF…
If MS2 = OPEN and there are 5 valid memories: ABCEABCE…
If MS2 = OPEN and there are 4 valid memories: ABCABCA…
If MS2 = OPEN and there are 3 valid memories: ABABA…
If Pull-up/Pull-down = Pull-down and MS2 = HIGH: D…
If Pull-up/Pull-down = Pull-up and MS2 = LOW: D…

Table 5. DYNAMIC EXAMPLE WITH FOUR VALID MEMORIES (T=MOMENTARY SWITCH IS TOGGLED; 0=OPEN; 1=CLOSED)

MS2	0	0	1	1	1	0	0	0	1	0	0	0	0	0	0	0
MS1	0	T	T	0	T	T	0	T	T	0	0	T	T	T	T	T
Memory	A	B	C	D	D	DC	C	A	B	D	B	C	A	B	C	A

Mode 3: Static Switch on MS1 and MS2

This mode uses two static switches to change memories. Table 6 describes which memory is selected depending on the state of the switches.

In this mode, it is possible to jump from any memory to any other memory simply by changing the state of both switches. If both switches are changed simultaneously, then the transition is smooth. Otherwise, if one switch is changed and then the other, the part transitions to an intermediate memory before reaching the final memory. The part starts in whatever memory the switches are selecting. If a memory is invalid, the part defaults to memory A.

This mode is set by programming the 'MSSMode' parameter to 'static'.

Table 6. MEMORY SELECTED BY STATIC SWITCH ON MS1 AND MS2 MODE (EXAMPLE WITH FOUR VALID MEMORIES)

MS1	MS2	Memory
OPEN	OPEN	A
CLOSED	OPEN	B (if valid, otherwise A)
OPEN	CLOSED	C (if valid, otherwise A)
CLOSED	CLOSED	D (if valid, otherwise A)

Mode 4: Static Switch on MS1, Static Switch on MS2 (Jump to Last Memory)

This mode uses two static switches to change memories.Unlike in the previous example, this mode will switch to the last valid memory when the static switch on MS2 is OPEN or CLOSED depending on the configuration of MS2. This means that this mode will only use a maximum of three memories (even if four valid memories are programmed). Table 7 describes which memory is selected depending on the state of the switches.

This mode is set by programming the ‘MSSMode’ parameter to ‘static’ and ‘Donly’ to ‘enabled’.

In this mode, it is possible to jump from any memory to any other memory simply by changing the state of both switches. If both switches are changed simultaneously, then the transition is smooth.When MS2 is set CLOSED, the state of the switch on MS1is ignored. This prevents memory select beeps from occurring if switching MS1 when MS2 is CLOSED. The part starts in whatever memory the switches are selecting. If a memory is invalid, the part defaults to memory A. The part starts in whatever memory the switches are selecting. If a memory is invalid, the part defaults to memory A. Otherwise, if one switch is changed and then the other, the part transitions to an intermediate memory before reaching the final memory.

Table 7. MEMORY SELECTED BY STATIC SWITCH ON MS1, STATIC SWITCH ON MS2 (JUMP TO LAST MEMORY) MODE

MS1	MS2	Memory
OPEN	OPEN	A
CLOSED	OPEN	B (if valid, otherwise A)
OPEN	CLOSED	D
CLOSED	CLOSED	D

CHAPTER 6

Software and Security

R3920 incorporates the following security features to protect the device from cloning and against software piracy:

- DLL protection by password - prevents a third party from using IDS to reconfigure parts
- Hybrid authentication by 128-bit fingerprint to identify parts in application software - prevents a third party from cloning a device's EEPROM because the fingerprint cannot be overwritten. Special functions can be used in fitting software to reject parts that do not match the expected fingerprint. This would prevent the piracy of fitting software
- DLL to hybrid pairing by using a software key in ARK to match product libraries with client software - a part can be 'locked' at manufacturing time so that it only communicates with the library it was programmed with. This prevents a third party from potentially upgrading a device with a different library in IDS or other application software.

Full software support is provided for every stage of development from design to manufacturing to fitting. For details, refer to the ARK User's Guide.

SDA AND I^2C COMMUNICATION

R3920 can be programmed using the SDA or I^2C protocol. During parameter changes, the main audio signal path of the hybrid is temporarily muted using the memory switch fader to avoid the generation of disturbing audio transients. Once the changes are complete, the main audio path is reactivated. Any changes made during programming are lost at power-off unless they are explicitly burned to EEPROM memory.

Improvements have been made to the ARK software for R3920 resulting in increased communication speed. Certain parameters in ARKonline can be selected to reduce the number of pages that need to be read out.

In SDA mode, R3920 is programmed via the SDA pin using industry standard programming boxes. I^2C mode is a two wire interface which uses the SDA pin for bidirectional data and CLK as the interface clock input. I^2C programming support is available on the HiPro (serial or USB versions) and ON Semiconductor's DSP Programmer 3.0.

CHAPTER 7

Input Connection and Layout Considerations

PCB AND COMPONENT LAYOUT CONSIDERATIONS

Careful PCB design and component placement in an R3920 design is important to reduce the possibility of audible artifacts and potential feedback paths. Any digital outputs from the R3920 are a potential source of interference through radiated or conducted signal paths.

It is recommended to connect unused audio input pins directly to MGND to minimize the possibility of noise pickup. Inputs are internally AC coupled, so there is no additional leakage current when inputs are connected directly to ground.

In order to further minimize noise at the inputs, MGND should be used as reference ground plane for input signals only. All input components should be ground referenced to MGND. This ground plane should be isolated from all other ground connections in the system.

H-BRIDGE INTERFERENCE

In order to minimize radiated H-bridge noise from coupling into the telecoil and analog inputs, use twisted pair wires from the hybrid pins OUT+/OUT− to the receiver. The conducted H-Bridge interference is transmitted primarily through the power and ground traces. In high power designs, keep the noisy current flow within the H-bridge circuit by connecting VBP and PGND to a decoupling cap directly and connect these nets to the battery terminals with a very short interconnect.

CHOICE OF DECOUPLING CAPACITOR

A decoupling capacitor is not a requirement in a R3920 design, however, if a high power, low impedance receiver is being used, then power supply decoupling will be necessary.

The choice of decoupling capacitor value is dependent upon the inductance of the power supply signal traces, with a long signal trace increasing the inductance in a resonant circuit. A capacitor value between 1 μF and 2.2 μF is sufficient for higher power receiver designs.

PCB LAYOUT

A suggested PCB layout includes star routing power supply lines back to the power source or battery. This is recommended to minimize noise and crosstalk in the system.

Separate ground planes for PGND and GND is not necessary but okay when connected together with narrow width traces. Connect the battery negative terminal as close to PGND as possible.

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