Chapter 6. Performance Analysis

Although the performance of the spread spectrum radio frequency transmission from the AudioLink to receiver cell sites is critical to overall IVDS system success, it was not the direct responsibility or concern of the researchers involved in the efforts described in this thesis. The interested reader is hereby referred to published research that investigated the wireless communications link used and its performance statistics [26]. This chapter, therefore, will restrict attention to the performance of the audio coding scheme alone.

Section 6.1. Early Testing During Development

During the evolution of the AudioLink numerous tests were conducted, at various stages of development, to evaluate its performance. These tests were used to indicate whether a recent change made to the device actually improved performance or degraded it. Most often the experiments took anywhere from a few minutes to a few hours, and were performed informally, with scratch notes recorded on scrap paper. Once the results were obtained from a specific test, decisions were made and little formal record of the test was kept. Although presenting all of the many tests and their results here would be impractical if not impossible, several of the more relevant tests warrant a brief general description, which is generated below.

Section 6.1.1. Signal Level Versus Microphone Distance

Since the IVDS communication scheme depends on acoustic signal transmission from the television speaker to the AudioLink microphone, the placement of the microphone with respect to the speaker is critical. In the final version of the AudioLink the embedded microphone is not used, and the signal is obtained acoustically through a clip-on microphone placed directly on the speaker. (The signal can also be obtained electrically via an RCA jack connected to the TV.) Earlier versions of the AudioLink, however, relied on the embedded microphone and the internal audio amplifier/filter circuitry. For these versions it was imperative that a sufficiently strong signal be obtained from the microphone, and methods of automatic gain control were examined to help ensure such adequate reception.

To learn more about the signal environment we investigated how the strength of the received signal varied with respect to microphone distance. The normalized results of a particular experiment are plotted in Figure 6.1 below, and indicate that a drastic rolloff in power occurs within the first twelve inches of distance. (The plot is normalized in that the power has been scaled to be one at a distance of one foot.) However, the decay becomes more gradual after a few feet, and the received power flattens somewhat to a value of about twenty percent of that experienced at the twelve-inch distance. Note that the rapid decay of power over the initial twelve inches provides strong justification for the use of a clip-on microphone instead of an embedded one, which would operate at a much larger distance and receive much less acoustic power in general. The signal corruption due to extraneous room noise is also usually larger for the embedded than the clip-on microphone.

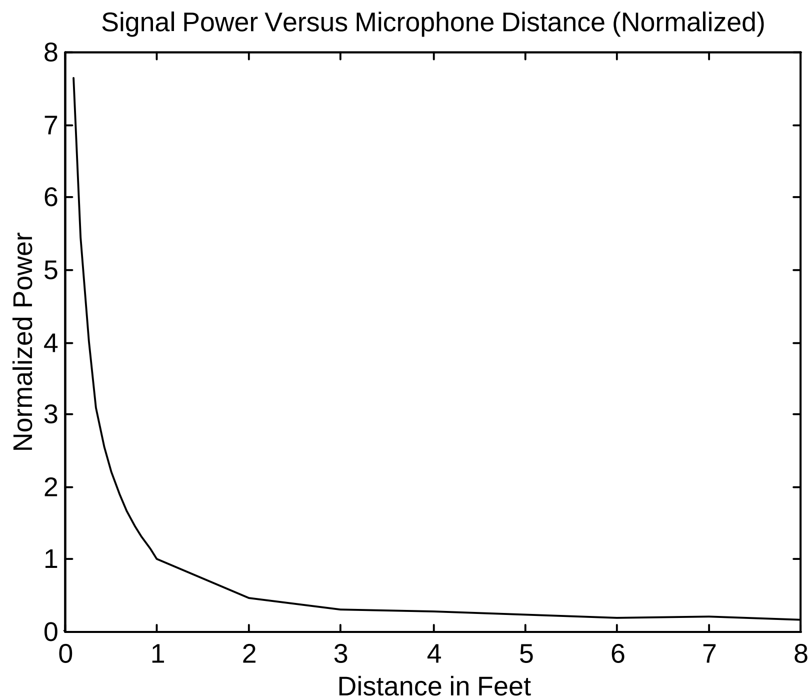

Figure 6.1 Power of Received Signal Versus Microphone Distance

Section 6.1.2. Robustness to Extraneous Room Noise

Since the AudioLink receives its signal acoustically (unless the RCA input option is used), extraneous room noises can corrupt the communications channel and compete with the television audio. The room noises are typically human voices and music, although miscellaneous household appliances such as hair dryers and vacuum cleaners can also add to the problem. During AudioLink development many experiments were performed to investigate qualitatively and quantitatively the amount of interference that could be tolerated. These experiments were often informal, where speech and/or music from the radio was played at various volumes while the AudioLink attempted reception of its codes. Many of these experiments were done formally, however, and digital versions of speech and music were used as standard interferences during code receptions. The results of these numerous experiments were provided in Chapter 4. The experiments showed that the communication scheme is relatively robust to human voice interference but less so to music, which tends to have frequency components that compete directly with the code sinusoids. The performance is very dependent, however, on the particular song and the local frequency content of the music.

Section 6.2. Performance Analysis of the Final Prototype

As part of the last phase of the IVDS research contract we were to evaluate the performance of the final AudioLink prototype once it was delivered from the manufacturer. The purpose of the performance analysis was primarily to establish a baseline to be compared against in later tests. By comparing to the baseline we could determine if future changes to the AudioLink enhanced or degraded performance.

To prepare for the performance analysis we produced a document outlining the planned tests and procedures. Once we received the final prototype, we performed the necessary tests and appended a *results* section to each test description in the proposal document. This amended document was given to the research sponsors as the contract deliverable. A reformatted copy of the “Test Procedures and Results for the New AudioLink” document is provided below. It contains detailed descriptions of the proposed tests and the results obtained thereof.

Test Procedures and Results for the New AudioLink

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Introduction

The new AudioLink will be tested in several phases. First the audio detection capability must be verified through coarse tests in the DSP Research Laboratory (DSPRL). Then proper radio frequency (RF) transmission must be verified via testing in the Center for Wireless Telecommunications (CWT) lab. Finally, once the basic capabilities have been verified, more detailed testing can proceed and qualitative and quantitative results can be compiled. An estimate for the total testing time, assuming no major hardware deficiencies exist, is 1.5-2.0 weeks.

We will test the hardware detection capability and software modules by detecting commands zero through fifteen. The detection of a secondary audio code will be tested for commands six through eight, and the release from mode one to mode zero is to be verified by secondary detection after commands fourteen and fifteen. A test tape similar to the May 21, 1997 tape will be created, and additional copies will be made to facilitate testing at locations other than the DSPRL. The format of the May 21, 1997 test tape was as listed below in Table 6.1.

Table 6.1 Audio Segments on the VHS Test Tape Produced on May 20-21, 1997

command = 00, message = 1234000
command = 01, message = 1234001
command = 02, message = 1234002
command = 03, message = 1234003
command = 04, message = 1234004
command = 05, message = 1234005
command = 06, message = 1234006, then repeated
command = 07, message = 1234007, then repeated
command = 08, message = 1234008, then repeated
command = 09, message = 1234009
command = 10, message = 1234010
command = 11, message = 1234011
command = 12, message = 1234012
command = 14, message = 1234014
command = 15, message = 1234015
command = 13, message = 0000000
command = 13, message = 0000001
command = 13, message = 0000002

One difference will be that the inserted audio codes will be hidden by insertion into small audio blocks. Another change will be that different message numbers will be chosen to better represent typical usage conditions. The messages of 12340xx listed above never utilize about ten bits out of the 31 available (see Table 6.2 below), and therefore about thirty sinusoids are always zero. This can slightly bias testing in that, after power normalization in the audio code insertion process, each sinusoid will contain more power. Using a larger base number containing approximately the same number of binary ones and zeros will incorporate power levels more likely in the typical case. The use of 1,987,654,300 as a base number is suggested, which will contain binary ones in its representation as listed in Table 6.3 below. Creation of such a tape will require approximately six hours, plus work to be performed by VA Tech Video Broadcasting Services.

Table 6.2 Number of Binary Ones in the Representation of (Old) Base Number 1,234,000

There are 08 ones in the representation of	0,001,234,000.
There are 09 ones in the representation of	0,001,234,001.
There are 09 ones in the representation of	0,001,234,002.
There are 10 ones in the representation of	0,001,234,003.
There are 09 ones in the representation of	0,001,234,004.
There are 10 ones in the representation of	0,001,234,005.
There are 10 ones in the representation of	0,001,234,006.
There are 11 ones in the representation of	0,001,234,007.
There are 09 ones in the representation of	0,001,234,008.
There are 10 ones in the representation of	0,001,234,009.
There are 10 ones in the representation of	0,001,234,010.
There are 11 ones in the representation of	0,001,234,011.
There are 10 ones in the representation of	0,001,234,012.
There are 11 ones in the representation of	0,001,234,013.
There are 11 ones in the representation of	0,001,234,014.
There are 12 ones in the representation of	0,001,234,015.

Table 6.3 Number of Binary Ones in the Representation of (New) Base Number 1,987,654,300

There are 17 ones in the representation of 1,987,654,300.
There are 18 ones in the representation of 1,987,654,301.
There are 18 ones in the representation of 1,987,654,302.
There are 19 ones in the representation of 1,987,654,303.
There are 15 ones in the representation of 1,987,654,304.
There are 16 ones in the representation of 1,987,654,305.
There are 16 ones in the representation of 1,987,654,306.
There are 17 ones in the representation of 1,987,654,307.
There are 16 ones in the representation of 1,987,654,308.
There are 17 ones in the representation of 1,987,654,309.
There are 17 ones in the representation of 1,987,654,310.
There are 18 ones in the representation of 1,987,654,311.
There are 16 ones in the representation of 1,987,654,312.
There are 17 ones in the representation of 1,987,654,313.
There are 17 ones in the representation of 1,987,654,314.
There are 18 ones in the representation of 1,987,654,315.

Unfortunately there is no way to automate the testing procedures without possibly corrupting the software we are attempting to test. Therefore human operators will be necessary on all tests to play and rewind tapes, press remote control buttons, and collect statistics.

Preparatory Work Results:

A test tape was produced with the audio segments and formatting as specified below in Table 6.4 and Table 6.5. Two copies of the test tape were made in order to prevent unnecessary wear on the original from repeated playbacks. An interference tape was also produced, and contains thirty seconds of audio resulting from the summation of twelve human voices. This thirty-second block was placed on the tape repeatedly until the tape was full. Very little or no silence was placed between the repeated thirty-second blocks.

Table 6.4 Audio Segments of the August 22, 1997 Test Tape

The test tape produced on August 22, 1997 contains the following audio segments:

command = 00, message = 1987654300
command = 01, message = 1987654301
command = 02, message = 1987654302
command = 03, message = 1987654303
command = 04, message = 1987654304
command = 05, message = 1987654305
command = 06, message = 1987654306, then repeated
command = 07, message = 1987654307, then repeated
command = 08, message = 1987654308, then repeated
command = 09, message = 1987654309
command = 10, message = 1987654310
command = 11, message = 1987654311
command = 12, message = 1987654312
command = 14, message = 1987654314
command = 15, message = 1987654315
command = 13, message = 0000000000
command = 13, message = 0000000001
command = 13, message = 0000000002

Parameters specified parameters during file creation:

Original file: AJO2_44E.WAV
Source/Destination:
    7) P5-100XL @ 44.1 kHz ----> ADSP2181 @ 16.0 kHz (SRCF=1.0, new)
command = x
message = bb + x (bb = 1987654300) (exceptions for message=13)
offset = 0.7 seconds
duration = 0.3 seconds
ADR = 27 dB

Table 6.5 Format of the August 22, 1997 Test Tape

The test tape produced on August 22, 1997 has the following format:
(WAV file name and (approximate) number of seconds of silence to follow)

c00.wav	5		
c01.wav	5		
c02.wav	5		
c03.wav	5		
c04.wav	5		
c05.wav	5		
c06.wav	2,	c06.wav	5
c07.wav	2,	c07.wav	5
c08.wav	2,	c08.wav	5
c09.wav	5		
c10.wav	5		
c11.wav	5		
c12.wav	5		
c14.wav	5		
c15.wav	5		
c13m00.wav	5		
c13m01.wav	5		
c13m02.wav	5		

Then this entire sequence was placed on the test tape repeatedly until the tape was full, but with approximately ten seconds of silence between the blocks.

Section 6.2.1. Test 1.

The first test is designed to indicate whether or not the new system works and whether it performs about equally well as the existing system. In order to ascertain this, we will bypass the audio amplifier and filtering stages, use a clip-on microphone, and run a series of tests to compare with the existing system. The tests will consist of detection of the codes from the new VHS tape and directly from the computer at various volume levels in the DSPRL. Only after we are convinced that the new unit performs about as well as the existing one under the current operating conditions should the analog audio preprocessor hardware (the audio amplifier and filtering stages and the embedded microphone) be enabled and further testing performed.

Also, it must be verified that the RF section is truly off when the AudioLink is in listening mode. We will test this with a shorted microphone input. The level and spectrum of the internal and circuit noise will be compared with that measured on the present board. It is imperative that the RF section does not interfere with the listening mode.

Test 1 Results:

Early tests readily indicated that the audio amplifier and filter circuitry enhanced detection performance, and so the analog audio preprocessor hardware was enabled for all subsequent tests. Testing also verified that the RF section was disabled whenever the AudioLink entered listening mode. Furthermore it was decided that the embedded microphone would not be used. A clip-on microphone or an RCA audio-input connector would be the two signal input options.

Section 6.2.2. Test 2.

After the brief initial tests in the DSPRL, the next series of tests will be performed using the entire detection-transmission path, i.e., including the RF link. All results arriving at the receiver cell should be verified. These tests should be performed in the CWT lab with the TV and VCR(s) already there. If the RF tests fail, little additional audio testing is appropriate until the RF problems are resolved; we have seen in the past that solutions to RF problems can affect and corrupt the audio subsystem. It is senseless to proceed with the more time-consuming tests specified below if changes are to be made to the system.

Test 2 Results:

Proper operation of the AudioLink through the entire detection-transmission path was verified, including the RF link. No problems were encountered.

Section 6.2.3. Test 3.

With the analog audio preprocessor hardware enabled, a second series of tests will be appropriate to determine the detection sensitivity with respect to AudioLink position. These too will be performed in the DSPRL, and will involve moving the AudioLink and detecting audio codes from various locations at several volume levels. These tests will not be comprehensive. If it is found that the performance is quite sensitive to position, subsequent tests can proceed with the analog audio preprocessor hardware disabled. Likewise, if performance seems relatively independent of device position, most further testing can be performed with the analog audio preprocessor hardware enabled. If the sensitivity level is found to be intermediate, however, more detailed analysis at this point is warranted to determine the impact. A decision should be made whether to operate with or without the analog audio preprocessor hardware.

Test 3 Results:

Early tests readily indicated that the audio amplifier and filter circuitry enhanced detection performance, and so the analog audio preprocessor hardware was enabled for all subsequent tests. Furthermore it was decided that the embedded microphone would not be used. A clip-on microphone or an RCA audio-input connector would be the two signal input options. Therefore performance tests with respect to device position were no longer necessary.

Section 6.2.4. Test 4.

There are many parameters that determine the detection performance of the AudioLink: the choice of target television commercial, the placement of the audio code in the commercial, the relative power level of insertion, the duration of the inserted code, the television and VCR hardware used during playback, the television volume level, the positioning of the microphone, the room environment and acoustic transfer characteristics, the ambient room noise, etc. It will not be possible to provide performance statistics for the units in general terms, but rather we can test them under well-defined and controlled conditions.

A baseline performance for the system should be established under such a set of controlled conditions. A TV-VCR station will be set up in the “quiet room” on the fifth floor of Whittemore Hall. Here the performance of the system can be quantified for:

- the specific VCR tape created for these tests (a specific commercial with specific code placements, relative power levels, and durations).
- a specific TV-VCR hardware combination
- a quiet environment with negligible acoustic interference from multipath effects
- negligible acoustic interference from external sources
- operation without the analog audio preprocessor hardware (the audio amplifier and filter circuitry should be disabled and a clip-on microphone should be used)

Three television volume levels will be used for testing. Quiet, medium, and loud levels should be chosen, with the digital readout of the volume levels noted. It is important to always reach each of the set volume levels from the same direction (e.g. from louder to softer). We have seen that the actual power corresponding to a given digital level will be different depending on from which direction it was approached. The quiet and loud levels should be chosen to correspond to the points where the device begins to fail occasionally.

We propose to detect the 21 code insertions at three volume levels, with ten repeats for each test. Thus there will be 21 * 3 * 10 = 540 tests. The results will be entered into spreadsheets for statistical analysis. Since each round of 21 detections requires about fifteen minutes, an estimate of the test time is 7.5 hours. Repeating each test five times instead of ten can cut this time in half. This may be especially important if comparison with the existing system is desired, since all tests should be performed for each system.

A second TV-VCR station will be used to obtain some knowledge of the effect of hardware variation on the detection performance. It is recommended that an old TV and VCR be used for these tests, to indicate performance in the worst hardware cases. The testing sequence outlined above will be repeated to gather statistics for comparison with the established baseline.

The results of Test 3 will determine whether these baseline experiments will also be repeated with the analog audio preprocessor hardware of the AudioLink enabled, and with varying placement of the AudioLink (and embedded microphone).

Test 4 Results:

It was determined that the “quiet room” on the fifth floor of Whittemore was in fact rather noisy due to a fan above the ceiling. Because of this and accessibility issues, we decided to perform the tests in the new DSP Research Laboratory in Whittemore 422, using the available Zenith VR4125 VHS VCR and the Mitsubishi CS-20203 television. Sufficient acoustic isolation was achieved by placing the microphone and speaker(s) in a cardboard box lined with Styrofoam and foam rubber insulation. The audio preprocessor hardware was enabled for all testing.

The results of the detection performance tests are provided in Table 6.6 below. As expected, copying VHS tapes reduced the signal-to-noise (SNR) ratio and increased the range of the frequency shifts that result from tape wow and flutter. Because of these signal degradations, detection performance from copied tapes was poor. Detection performance improved when the original VHS tape was used, and performance was best when the computer was used to generate the audio signals. The “u2” and “d2” abbreviations in the “Volume Level” column of the spreadsheet represent the direction from which the final TV volume level was approached (as in “up to” and “down to”). As mentioned above, we have seen that the actual power corresponding to a given digital level will be different depending on from which direction it was approached.

Table 6.6 Performance Test - Code Detection

IVDS Performance Tests: Detection																								
Audio Source	Volume Level	# of Trials	Number of Valid Detections of Command Number:																					Detection Rate (%)
			0	1	2	3	4	5	6a	6b	7a	7b	8a	8b	9	10	11	13	14	15	13a	13b	13c	
Orig VHS	d2 12	5	5	4	4	3	3	5	5	5	3	4	4	4	5	5	5	3	5	3	5	4	5	84.76
Orig VHS	u2 18	5	3	2	4	4	5	4	5	4	4	4	4	5	4	4	4	4	5	5	5	5	5	84.76
Orig VHS	u2 25	5	5	2	5	3	5	4	5	4	4	3	4	4	5	3	4	2	5	5	5	5	5	82.86
Copy VHS	d2 12	5	4	4	3	3	2	5	1	3	5	2	4	2	4	4	3	4	4	4	4	2	1	64.76
Copy VHS	u2 18	5	3	3	3	3	3	3	3	3	3	2	5	2	2	4	1	3	3	3	2	4	2	57.14
Copy VHS	u2 25	5	0	2	2	2	3	0	1	2	2	1	2	1	1	2	0	3	0	0	3	4	1	30.48
P5-200	0.25	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	100.00
P5-200	0.5	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	100.00
P5-200	0.75	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	10	100.00
Percentage Detection			83	78	85	80	85	85	83	85	85	77	88	80	85	87	78	82	87	83	90	90	82	

It was decided that testing with a second TV-VCR combination would not provide the insight originally expected, since controlling and matching volume levels would be impossible. Therefore a single TV-VCR station was used for the testing, which should enhance the repeatability of the results.

Using the interference tape described in the introduction we also tested detection performance in the presence of human voices. Table 6.7 below tabulates the results of these tests. A 200 MHz Pentium computer acted as the audio signal source, with the microphone clipped onto the computer speaker. The interference tape was played on the VCR with the sound emanating through the television speaker at a distance of six inches from the computer speaker. The acoustic isolation box enclosed the test area to the extent possible. As expected the audio code detection was relatively robust to human voice interference, but did degrade with increasing interference power level.

Table 6.7 Performance Test - Interference

IVDS Performance Tests: Interference								
Audio Source	Soundcard Settings	Speaker Volume	Television Volume	Distance (inches)	Command Number	Number of Trials	Number of Detections	Percentage Detection
P5-200	default	0.5	u2 12	6	0	100	99	99.00
P5-200	default	0.5	u2 14	6	0	100	87	87.00
P5-200	default	0.5	u2 16	6	0	100	64	64.00
P5-200	default	0.5	u2 17	6	0	100	40	40.00

Section 6.2.5. Test 5.

In order to verify that the alarm detection threshold is not set too low we will try to cause false detections with various household appliances and other sources of periodic acoustic signals. This will entail placing an AudioLink in proximity to various appliances such as refrigerators, fans, air conditioners, hair dryers, kitchen blenders and mixers, microwaves, alarm clocks, etc. Since the AudioLink can enter an infinite loop, by detecting a relatively strong frequency component that does not satisfy the alarm beep rate constraint, and thus not return to listening mode, these tests must also include concurrent detection of valid audio codes from the tape or from a computer. If it is impractical for an appliance to be brought to the DSPRL for testing, home tests can be performed by bringing AudioLinks and copies of the test tape to the test sites.

Test 5 Results:

In order to estimate the rate at which false detections occur, AudioLinks were installed in six offices and labs in Whittemore Hall. These AudioLinks were modified so that upon any reception of a valid audio code or detection of an alarm, the results would be transmitted immediately without verbal prompting or IR verification. Several days of false alarm and false detection data were collected and analyzed.

Audio codes were falsely detected approximately once every 24 hours per AudioLink unit. Since the audio signal input is processed every 12.8 msec, there are 6.74E+6 opportunities for each AudioLink to falsely detect a code each day. Thus the false detection rate is on the order of 1.5E-7. This rate can be significantly reduced if necessary by requiring the triplication code result to be valid, including the CRC. Currently, if the triplication result is invalid, detection is given a “second chance” by considering the results in each of the three frequency bands. Although this enhances detection performance when a true code is to be detected, it also increases the false detection rate significantly. Other options to lower the false detection rate include raising the threshold for valid sinusoid classification in the FFT bins and/or requiring that valid code be present in more than a single FFT block (requiring persistence in time).

False alarm detections occurred more frequently than the false code detections. As expected there are many situations in which a single strong frequency component exists in the candidate range for a period of time and is also modulated at a rate consistent with common alarms. This has been observed with music, background sounds in movies, and with acoustic signals produced by computer hard drives and monitors. The false alarm detection rate can be significantly lowered if more than a single two-second block is required to have an alarm present. For example, we could require that the alarm signal be present between 0 - 2 seconds, 10 - 12 seconds, and 20 – 22 seconds. Most of the false alarm sources we encountered did not have the persistence in time to trigger with such augmented criteria.

It was decided that the alarm detection should take precedence over the audio code detection, and so the infinite loop situation described above was deemed acceptable. Therefore the proposed testing was no longer necessary, and no system modifications were required.

It is imperative to remember that reducing the false alarm and false code detection rates will inevitably degrade the detection performance when true alarms and audio codes are present. This is a fundamental tradeoff that must be considered when choosing threshold levels and other operating parameters.

Section 6.2.6. Test 6.

It is possible that a certain ordering of detected commands could trigger a software bug. Thus we will run a test where each command is followed by every other possible successor. This would entail 16 * 16 = 256 tests. Since the audio for these tests can be generated and played from a computer in the DSPRL, the testing should not require too much time (two or three hours). A related series of tests will be used to confirm proper operation when an alarm is present during active reception of each of the sixteen valid audio commands. (A proposed modification to the detection software will require that audio code detection completes before the presence of an alarm is considered.)

Test 6 Results:

The sequencing tests were all successful, and no problems were encountered. The results are tabulated in Table 6.8 below. Since it was decided that the alarm detection should take precedence over the audio code detection, the tests involving simultaneous alarm and audio code detection were not required.

Table 6.8 Performance Test - Command Sequencing

IVDS Performance Tests: Sequencing																
First	Second Command Number (1=success, 0=failure)															
Command	0	1	2	3	4	5	6	7	8	9	0	11	12	13	14	15
0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
2	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
3	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
4	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
5	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
6	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
7	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
8	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
9	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
10	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
11	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
12	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
13	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
14	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
15	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1

Section 6.2.7. Test 7.

With the analog audio preprocessor hardware and the embedded microphone enabled on the new system, a more detailed performance analysis with respect to device position may be warranted. Here too general performance is impossible to measure, but some knowledge can be gained by a series of tests with well-defined and controlled conditions and parameters. The baseline performance statistics established under Test 4 can be used for comparison. All parameters should be held constant except for those being tested. [details will be determined later]

Tests 7 Results:

Early tests readily indicated that the audio amplifier and filter circuitry enhanced detection performance, and so the analog audio preprocessor hardware was enabled for all the tests listed in this document. Furthermore it was decided that the embedded microphone would not be used. A clip-on microphone or an RCA audio-input connector would be the two signal input options. Therefore performance tests with respect to device position were irrelevant.