NASA/TM—2014-216664

Power, Avionics and Software—Phase 1.0: Subsystem Integration Test Report

William D. Ivancic, Obed S. Sands, Casey J. Bakula, Daniel R. Oldham, Ted Wright, and Martin A. Bradish
Glenn Research Center, Cleveland, Ohio

Joseph M. Klebau
Science Applications International Corporation, Cleveland, Ohio

August 2014

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National Aeronautics and
Space Administration

Glenn Research Center
Cleveland, Ohio 44135

August 2014

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William D. Ivancic[a,*], Obed S. Sands[a], Casey J. Bakula[a], Daniel R. Oldham[a], Ted Wright[a], Martin A. Bradish[a], Joseph M. Klebau[b]

[a]*NASA Glenn Research Center, 21000 Brookpark Road, Cleveland, Ohio 44135 USA*
[b]*Science Applications International Corporation (SAIC) , Cleveland, Ohio USA*

Abstract

This report describes Power, Avionics and Software (PAS) 1.0 subsystem integration testing and test results that occurred in August and September of 2013. This report covers the capabilities of each PAS assembly to meet integration test objectives for non-safety critical, non-flight, non-human-rated hardware and software development. This test report is the outcome of the first integration of the PAS subsystem and is meant to provide data for subsequent designs, development and testing of the future PAS subsystems. The two main objectives were to assess the ability of the PAS assemblies to exchange messages and to perform audio testing of both inbound and outbound channels. This report describes each test performed, defines the test, the data, and provides conclusions and recommendations.

Keywords: Communication Architecture, Protocols, Telemetry, Avionics

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*Corresponding author
Email addresses: `william.d.ivancic@nasa.gov` (William D. Ivancic), `obed.s.sands@nasa.gov` (Obed S. Sands), `casey.j.bakula@nasa.gov` (Casey J. Bakula), `daniel.r.oldham@nasa.gov` (Daniel R. Oldham), `ted.wright@nasa.gov` (Ted Wright), `martin.a.bradish@nasa.gov` (Martin A. Bradish), `joseph.m.klebau@nasa.gov` (Joseph M. Klebau)

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1. Background

The current Extravehicular Mobility Unit (EMU) suites (space suits) are still using technology from the 1970's and 1980's, particularly regarding the communications system. These systems work well, and there is always a reluctance to change space systems. There is nearly always the desire to be backwards compatible with proven systems, to reduce risk. Unfortunately, this backwards compatibility often occurs to the detriment of new technologies infusion, the same technologies that may actually reduce cost while dramatically improving capability.

In order to infuse new technologies into the EMUs and prepare for future National Aeronautics and Space Administration (NASA) missions, NASA is currently developing an Advanced Extravehicular Mobility Unit (AEMU)[1]. This is being performed within the Advanced EVA Systems Development Project under the Advanced Exploration Systems (AES) Program. A key part of this development is the spacesuit Primary Life Support Subsystem (PLSS) technology unit for long-duration microgravity or planetary missions and vacuum or low-pressure environments. NASA Glenn Research Center (GRC)'s role is to develop the Power, Avionics and Software (PAS) subsystem.

The PAS subsystem being developed at GRC supports the AEMU program at NASA's Johnson Space Center (JSC). GRC's role is to develop a prototype suite avionics subsystem and to research new technologies for the AEMU. The results will be used to refine requirements for the AEMU as well as to potentially integrate new technologies into the AEMU.

2. Introduction

This report describes PAS 1.0 subsystem integration testing and test results that occurred in August and September of 2013. This report covers the capabilities of each PAS assembly to meet integration test objectives for non-safety critical, non-flight, non-human-rated hardware and software development. The reported upon activities were performed with regard to applicable requirements to ensure design goals and objectives are satisfied. For an in-depth look at the PAS system network architecture for this test refer to the Power, Avionics and Software Communication Network Architecture publication [1]. This test report is the outcome of the first integration of the PAS subsystem and is meant to provide data for subsequent designs, development and testing of the future PAS subsystems. The test results presented in this document are from a developmental subsystem that consists of an integration of assembly level developmental breadboards. Note that this is developmental hardware and does not represent flight hardware or its final configuration. The two main objectives were to assess the ability of the PAS assemblies to exchange messages and to perform audio testing of both inbound and outbound channels. These two objectives were meet. This report discusses each test performed, defines the test, the data, and provides conclusions and recommendations.

[1] A spacesuit that provides environmental protection, mobility, life support, and communications for astronauts performing an Extravehicular Activity (EVA)

3. Test System

The PAS 1.0 subsystem interface test configuration is shown in Figure 1: Test Configuration. It consists of a caution warning and control assembly, a graphical user interface for the caution warning and control subsystem, a high definition video camera, informatics (INFO), the communication assembly, an audio processing unit and a Measurement, Test and Validation (MTV) computer. All assemblies are connected through a NetGear ProSafe® GS724T Gigabit Ethernet (GigE) managed switch. Although, technically, the PAS bus used in the test is GigE, the Avnet board (outbound audio), all of the netbooks, Radio, and the Informatics assembly are only FastEthernet compatible (10/100 Mbps). Fortunately, the Netgear GS724T is 10/100/1000 compatible. The switch was configured such that all connected ports transmit signals were mirrored to a single monitoring port. The MTV was connected to that mirrored port.

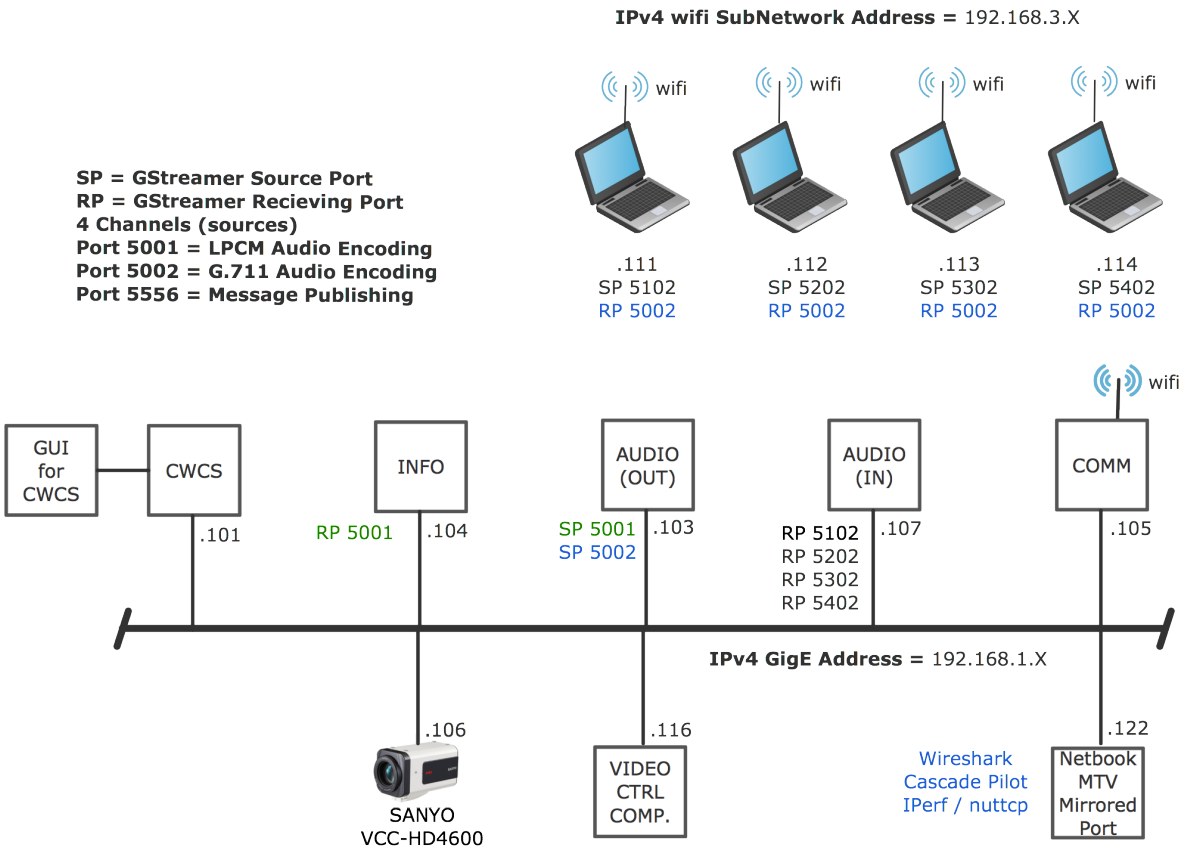


Figure 1: Test Configuration

All communication for these tests used Internet Protocol version 4 (IPv4) addressing. All subsystems are on class-C subnet 192.168.1.0. All external audio sources are on class-C subnet 192.168.3.0. All routing was statically configured. The Communication Subassembly (COMM) is attached to both networks. Thus, all routes off suit use COMM address 192.168.1.105 as the default gateway while all external systems use COMM address 192.168.3.105 as the default gateway. It is important to note that this static routing technique was done simply to test the messaging between assemblies. This would not be done for a multi-homed radio system such as would be required on an AEMU deployed for terrestrial sorties. Furthermore, if one had multiple AEMUs deployed, the routing for example from the informatics system on suit A to the informatics system on suit B, would be from the internal subnet on suit A to the internal subnet on suit B. It would not be from the internal subnet on suit A to the common radio subnet of which both

suit A and suit B belong.

The amount of traffic generated was insignificant relative to the bandwidth available on the GigE bus, even when including our High Definition Television (HDTV) video. All assemblies were connected over GigE through a GigE switch. An Ethernet switch ensures that only data destined to a particular device will be seen by that device (unless, one overrides this by port mirroring). All other traffic with the exception of multicast and broadcast traffic is filtered.

Request/Respond is discouraged as it requires maintaining state. Instead, subsystem communications is via a Publish/Subscribe mechanism using ZeroMQ [2]. One feature of ZeroMQ is that there is no "daemon" or broker process that runs in the background. ZeroMQ messages are defined using Google Protocol Buffers (GPB)or simply protocol buffers [3] in a common aemu.proto file. This file can be found in the appendix of Power, Avionics and Software Communication Network Architecture publication [1]. Each message type has specific parameters associated with it. Those parameters are also defined in the aemu.proto file. There are two basic message types: TELEMETRY and EVENT. TELEMETRY provides status whereas an EVENT reports that something has happened or to relay that an assembly needs some attention. The term "command" is intentionally avoided as "command" implies request/respond which requires systems to maintain states. Only publish/subscribe is implemented which *does not* require systems to maintain states.

4. Assembly Descriptions:

The following section provides a brief description of each assembly. It is important to note that many of the assemblies in PAS 1.0 are implemented using development boards. More often than not with such boards, the processors and supporting tools are reasonably adequate, but peripheral support is lacking. For example, buffers may be rather small on Ethernet interfaces resulting in reduced throughput between devices. Many of the test results related to throughput are a direct result of limited peripheral support.

4.1. Communications Assembly (Router/Radios)

The Communication Subassembly (COMM) for this testing consists of a wireless interface between various PAS assemblies and the rest of the off-suit EVA network. It is essentially a router between the on-suit network and the external wireless network. The COMM Assembly hardware consists of a Beaglebone processor board (manufactured by Circuitco) which uses a Texas Instruments AM335x "Sitara" CPU (1 GHz ARM Cortex A8 core) and a Commercial-Off-The-Shelf (COTS) 802.11n adapter. The COMM assembly performs the routing of information between the PAS Subsystems and between AEMUs via the radio network. The COMM assembly also performs configuration control of the attached 802.11n radio.

All audio streams are unicast because the chosen Linux operating system version on the current netbooks does not support multicast. The PAS subsystem is on the 192.168.1.xxx subnet (subnet mask 255.255.255.0) and the Wireless Fidelity (a.k.a. Wireless Internet) (WiFi) radios are on the 192.168.3.xxx subnet (subnet mask 255.255.255.0). Internet Protocol (IP) addresses are statically routed during this test. For simplicity during initial testing, the GStreamer audio [4] was transmitted between assemblies using User Datagram Protocol (UDP) and unicast addressing using different destination ports to identify the different sources. The drawback is that n number of streams from each source requires n number of destinations. Since there is only one inbound destination this was not as issue on the inbound side. For the outbound audio, there were four off suit destinations (plus the Linear Pulse Code Modulation (LPCM) stream to Informatics). Thus, four unicast streams were sent over the wireless link instead of one multicast stream. The GStreamer audio can also be run over UDP using a simple multicast address with different ports to identify different sources. The COMM assembly provides this result in one multicast stream per source instead of mxn unicast streams where m are the number of sources and n the number of destinations.

4.2. Audio Assembly

Originally the Audio assembly was envisioned as consisting of one, Avnet Spartan®-6/OMAP development kit. However, when attempting to process one outbound audio stream and 4 inbound audio streams using GStreamer and G.711 encoding and processing these in software (rather than utilizing the DSP support chip), it was determined that this Avnet card did not have sufficient processing capability to move between encoding the outbound streams and

decoding the four inbound streams[2]. In order to meet the test schedule, a decision was made to split the outbound processing and inbound processing by using one Avnet development board for outbound audio and one Avnet development board for processing the four inbound streams. Unfortunately the Avnet development kit did not have sufficient processing to simultaneously decode four inbound streams. It could barely decode two inbound streams in real-time. Thus, the second Avnet development kit was replaced with an Acer Ze7 netbook computer having an Intel N2600 Atom 1.6 GHz Dual-core processor running Linux. The netbook was able to handle four inbound streams. Thus, all the Audio tests were performed using on Avnet Spartan®-6 development board for outbound traffic and one netbook for inbound traffic.

4.3. Informatics Assembly

The Informatics assembly provides human interfacing to the PAS subsystem through a graphics display. The display presents functions such as consumable tracking, metabolic rate, assembly health and status, electronic procedures reading, and more. For this test cycle, the Informatics assembly was built using an Advanced RISC Machines (ARM) 7 architecture and a PowerVR graphics coprocessor found on the COTS Beaglebone computer board. Also, the current system uses an *LCD7* 800 by 480 resolution 7 inch Liquid Crystal Display (LCD) display that mounts on top of the Beaglebone. The *LCD7* has five push-buttons that emulate keystrokes for operator input. Angstrom Linux was chosen as the operating system because it has drivers that support the hardware accelerated Graphics Processing Unit (GPU). Qt version 4.8 [5] was installed from the Angstrom software repository and used for Informatics Graphical User Interface (GUI) development. Figure 2 shows an example of the Consumables Screen.

Figure 2: Informatics Consumables Screen-Shot

4.4. Caution, Warning and Control Assembly

The Caution, Warning and Control (CWC) assembly contains processing capability to provide control to and monitoring of the Primary Life Support Subsystem (PLSS). It is integrated into the PLSS backpack and directly interacts with various motors and sensors. The sensor data and equipment states are relayed to other PAS assemblies and off-suit for monitoring. The Caution, Warning and Control System (CWCS) assembly is connected to the PAS

[2]Note, the computational load for G.711 is around 0.01 to 0.02 MIPS per channel . This explains why the ARM CPU was overwhelmed when attempting to run the G.711 codec. G.729 would have faired worse as G.729 requires 20 – 30 MIPS.

subsystem via the PAS Ethernet bus and communicates with other assemblies uses ZeroMQ over Transmission Control Protocol (TCP).

The CWCS GUI provides the ability to monitor the telemetry from the CWCS as well as emulate switches the CWCS. Communication between the CWCS and CWCS GUI is over a separate Ethernet bus using TCP/IP protocols in a server/client arrangement.

For these tests, a CWCS GUI PAS Audio page was added to emulate switches and controls related to the audio subsystem. Examples of the types of events being sent are CWCS status, master volume, inbound channel volume, the caution and warning tone event messages, and the in-suit pressure. A screen-shot of the CWCS GUI page can be seen in Figure 3.

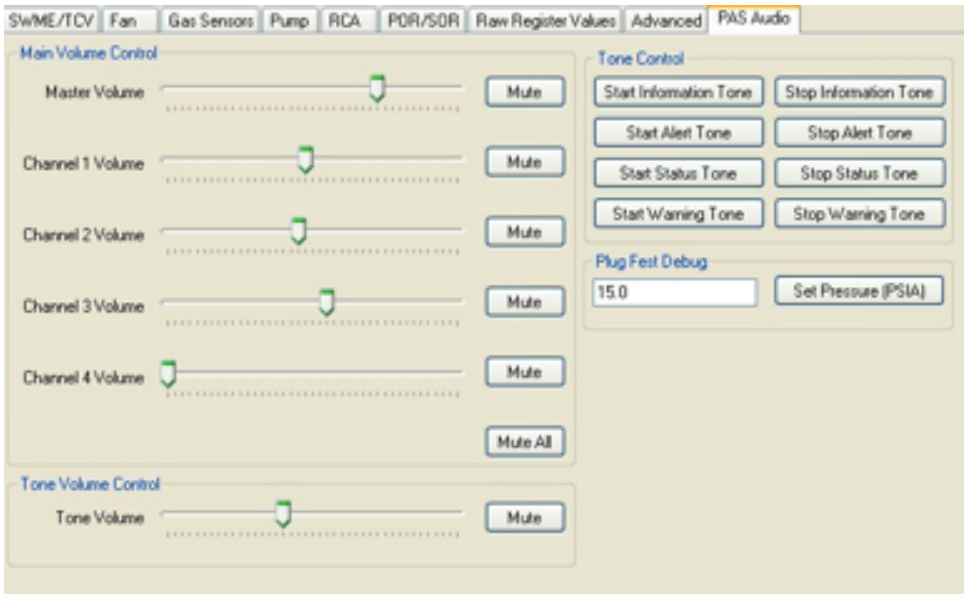


Figure 3: CWCS GUI Screen-Shot

4.5. Video Assembly

The current PAS 1.0 architecture calls for a suit mounted video camera that will interface directly with the PAS system data bus. The purpose of the suit camera is to provide record keeping as well as situational awareness to ground support and other mission assets during EVA. Thus with a wide range of uses, the camera must have broad flexibility in resolution, frame rate, and the ability to generate multiple video streams simultaneously. Video stream rates are expected to be as high as 15 Mbps for high quality High Definition (HD) at 30 fps. These rates are orders of magnitude higher than all other bus traffic that may exist in the currently envisioned PAS 1.0 subsystem. As such, video has a greater potential to interfere with other system traffic. In order to identify possible interactions, a video stream was sent over the GigE bus to ensure there is sufficient bandwidth on the bus to allow all necessary communication between assemblies occurs and that the quality of communications remains intact. For PAS 1.0 the video source was a SANYO VCC-HD4600 high definition IP Camera. The Sanyo camera was chosen as a close functional and performance approximation to the camera currently envisioned for the suit. The Sanyo camera was attached to the NetGear Gigabit switch, the PAS bus, on one of the available ports. It was given address 192.168.1.106. The video sink (or destination) was displayed on a computer attached to the NetGear Gigabit Ethernet switch on one of the ports. It was given address 192.168.1.116. The SANYO camera provided client software to both configure the camera and display the video via a Web-based application interface.

5. Interface Test Summary

The section describes the various tests performed in August and September of 2013 as well the results and conclusions. The detail test procedures and data can be found in Appendix A

5.1. Network Connectivity

The most basic function of the COMM assembly is to connect the assemblies on the suit to the EVA wireless network allowing the suit to communicate with other suits and a spacecraft. The network connectivity test demonstrated that each assembly on the PAS bus can communicate through the radio to a computer or device that is connected to the EVA wireless network, and vice versa. This test consists of a series of network pings[3] [6] between every PAS assembly and every wireless node to insure all combinations of connectivity are intact. The network connectivity test was performed many times during the course of the test campaign, both as a test in and of itself and as a precursor test to many other tests to ensure that the subsystem is operating properly. This test consistently passed during nominal test conditions with the exception of the failures identified under the following conditions:

- A PAS device was “locked up” due to circumstances unrelated to this test and needed to be rebooted
- During the Network Performance Tests, when the network was flooded with more traffic than the COMM hardware could handle, causing the network interfaces to fail. This condition was corrected by power cycling the devices being tested.

In preliminary PAS testing, the Informatics software would quit unexpectedly with a ZeroMQ error message when the CWCS was restarted. Ideally this should not happen. After updating to the latest ZeroMQ software, the Informatics software would no longer quit unexpectedly and would continue to receiving messages after a CWCS restart, but the Informatics assembly would not reestablish communication with CWCS (receiving its published messages) automatically as it should have. To start receiving CWCS messages again, the Informatics software would have to be restarted. It is believed that the Audio software exhibited the same behavior (rebooting the CWCS required restarting software on the Audio assembly in order to receive new messages from CWCS). Both assemblies were never "locked up", but they were not receiving all the messages they should have been until the application using the ZeroMQ socket was restarted. The consensus is that this is most likely caused by the Ethernet software on the CWCS VxWorks operating system and some incompatibility with the Portable Operating System Interface for Unix (application programming interface) (POSIX) assumptions made by ZeroMQ causing socket reinitialization failures when CWCS is restarted. There is supposed to be some automatic handshaking that allows sockets to be connected in any order and reconnect properly after disconnects, but that handshaking was not working correctly when the CWCS assembly was involved. Also, note that VxWorks is not officially supported by ZeroMQ, but the PAS avionics team managed to compile it and it seemed to work well otherwise.

5.2. Network Time Protocol

Network Time Protocol (NTP) [7] was used to maintain clock synchronization between the PAS assemblies. The Radio Testbed was used as the master clock, and all other PAS assemblies were configured to use the Radio Testbed as their sole time source. Since the Radio Testbed did not have another device available to use as its master clock, an arbitrary time was provided to the testbed during its boot process. Because NTP generally has a very slow convergence rate, to avoid significant time-stepping in the assemblies all PAS assemblies were forced to step to the Radio Testbed’s time during their boot process. Without this forced step, the PAS assembly clocks will not synchronize to the Radio Testbed clock due to the large difference in system times. The convergence rate is adjustable, but EVA’s are projected to be last 8 hours or less, which is not enough time to properly synchronize the assemblies’ clocks. Thus, if NTP is deployed, there will be a need to pre-synchronize the assemblies’ clocks at power-up.

The convergence properties of NTP were tested by the following process:

1. Obtain time synchronization among all PAS assemblies

[3] Ping is a computer network administration utility used to test the reachability of a host on an Internet Protocol (IP) network

2. Manually step the system time on the Radio Testbed by various time increments
3. Observe the behavior of the other PAS assemblies as they attempt to resynchronize with the Radio Testbed

The following general observations were made:

- After the system time on the Radio Testbed was stepped, the NTP slave devices temporarily refused to attempt re-synchronization for periods on the order of 30-60 minutes. This was due to logic in the NTP software that profiles each master clock source and assigns a trusted value to it based on observations of the clock. In this case, the time-stepping on the Radio Testbed appeared to the slave devices as erratic, untrustworthy behavior, and therefore the slave devices refused to re-synchronize until the behavior of the Radio Testbed clock stabilized and the trust factor was improved. This functionality is very useful to maintain network clock stability when large banks of master clock sources are used across the globe, but a better way to synchronize clocks may need to be investigated for a distributed PAS architecture.
- Even for sub-second discrepancies between assembly clock times, NTP can often take hours to obtain re-synchronization. Although many of NTP's performance parameters can be adjusted via configuration files, NTP remains heavy-weight for a PAS-like subsystem. Another, solution may be to add a battery backed clock and to preset each assemblies' initial time which would lessen the time delta thereby reducing the synchronization time. Lack of time synchronization between telemetry sources will complicate inter-subsystem communications. Thus, one conclusion is that assembly clocks will need to be synchronized in some fashion when assemblies are networked.
- The behavior of NTP can be and was modified to suit the needs of PAS, but most of these modifications simply disabled a lot of the advanced functionality of NTP.

Therefore, if NTP is deployed, periodic, forced synchronizations with the Communication timestamp might be sufficient to maintain clock synchronization between the PAS assemblies over the duration of an EVA. The same is true between the COMM assembly and its higher-tier clock (Access Point, Mission Control, etc.).

5.3. Assembly Status

The Assembly Status Test verifies that all assemblies are operational and are publishing information. It also verifies that the status message adheres to both the format and the transport protocol used by the PAS assemblies. Google Protocol Buffers (GPB) [3] were used to define the message formats, and ZeroMQ [2] was used to facilitate the transfer of these messages over the PAS bus. The verification was performed by examining the Informatics COMM screen, which is divided into four quadrants (one for each assembly) as shown in Figure 4. The background color of the quadrant changes to green indicating that current status information has been received by the Informatics assembly (for PAS 1.0), status is updated once per second). There is also a timestamp field in each assembly quadrant that is updated with the time of the latest received status message. The Informatics, Audio, and CWCS assemblies showed green status and had updating timestamps throughout the integration testing. Initially, the COMM assembly was not showing up as operational on the Informatics display. But an examination of the "Cyclical Redundancy Check (CRC) error" field on the Informatics COMM screen indicated that the number of network messages being rejected for CRC failures was being incremented every second that the COMM assembly was turned on. A short investigation determined that the COMM assembly was sending data using an outdated version of the wire format. After updating the packet formatting code on the COMM assembly, it showed green on the Informatics display. All assemblies were then verified as operational, and no more CRC errors were found during the integration testing.

5.4. Configuration and Performance Measurement Telemetry

Often telemetry refers to ALL data coming from a spacecraft to the ground. For the AEMU, telemetry only refers to system and subsystem status, configuration and performance measurement data (e.g. suit internal pressure, suit internal temperature, power supply voltage, power supply current and radio transmission rate). All other data flows such as file transfers, and streaming video and audio are considered data flow and not telemetry.

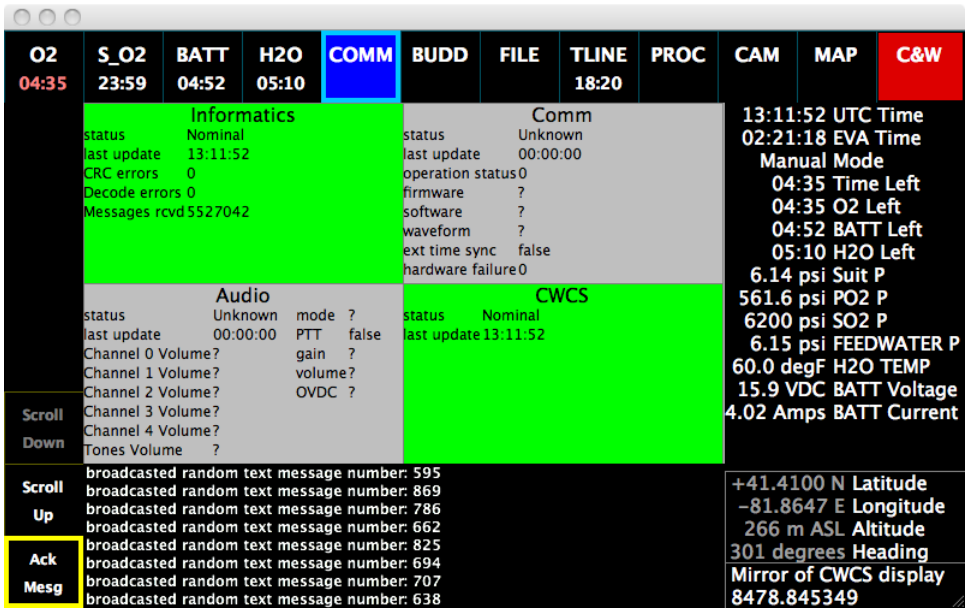


Figure 4: Informatics Communication Screen

The Configuration and Performance Telemetry tests verify that the CWCS assembly is sending required telemetry messages and that they are being received and displayed properly by the Informatics assembly. These telemetry messages include consumables measurements, physiological readings (i.e. heart rate and metabolic rate), basic telemetry (i.e. mode setting, suit pressure, etc.) and messaging that reflects what will be displayed on the PAS CWCS high-critical display. A display sample can be seen in Figure 2: Informatics Consumables Screen-Shot.

5.4.1. Consumables

There are four consumable levels that are computed on the CWCS and monitored by Informatics: Primary Oxygen (PO2), Secondary Oxygen (SO2), Battery Life (BATT), and Water (H2O). There are four consumable display screens on the Informatics display – one for each monitored consumable. Consumables telemetry is verified by checking that the consumable levels as displayed by Informatics are decreasing. All four consumable displays showed decreasing values during the tests.

5.4.2. Physiological

Heart Rate and Metabolic Rate information is sent by the CWCS and displayed on each of the Informatics consumables screens. These displays showed slightly varying values updated once per second during the testing, indicating that telemetry was being sent and received.

5.4.3. Basics

The Basics telemetry consists of suit information that appears on all Informatics screens: EVA time, CWCS mode, suit pressure, feed water pressure, water temperature, battery voltage, and battery current. All displayed values on Informatics were verified to match the values sent by CWCS. The suit pressure value was changed during audio testing, and the Informatics display of this value was verified to track the changes.

5.4.4. CWCS Two Line Display

The Informatics display has a small section that is intended to mirror the contents of the two-line text display driven by the CWCS. There was no two-line display attached to the CWCS during the integration tests, but telemetry showing a count (incremented once per second) was sent for testing. The two-line text mirror on Informatics was verified to be displaying an incrementing count - shown in the bottom right corner of Figure 4.

5.5. Audio Tests

These tests demonstrated movement of compressed streaming audio between AEMUs through the COMM assembly. The Audio assembly functions for this test include: 1) transmit one outbound G.711 audio stream to four external systems; 2) transmit one outbound LPCM audio stream to Informatics; 3) receive and process G.711 encoded inbound audio streams from four external entities (off-suit assets like other EMUs, Mission Control, etc.); and 4) activate the necessary caution and warning tones. External off-suit nodes were emulated using netbooks running Ubuntu 12.04 with GStreamer 0.10.36. They each used a Wireless Local Area Network (WLAN) adapter to join an 802.11 Ad Hoc WLAN network that included the COMM assembly and four netbooks directly addressed to the WLAN network, 192.168.3.0. The COMM assembly routed the streaming audio packets between the two networks, the internal PAS network, 192.168.1.0 and the WLAN, 192.168.3.0.

In order to understand the test results it is important to understand the basic workings of GStreamer [4].

> "GStreamer's development framework makes it possible to write any type of streaming multimedia application. The GStreamer framework is designed to make it relatively easy to write applications that handle audio or video or both or any kind of data flow. The applications it supports range from simple Ogg/Vorbis playback, audio/video streaming to complex audio (mixing) and video (non-linear editing) processing. GStreamer is made up of Elements, Pads, Bins and pipelines, and Communication. An element is the most important class of objects in GStreamer. You will usually create a chain of elements linked together and let data flow through this chain of elements. By chaining together several such elements, you create a pipeline that can do a specific task. Pads are element's input and output, where you can connect other elements. A bin is a container for a collection of elements. A pipeline is a top-level bin. It provides a bus for the application and manages the synchronization for its children. Once started, pipelines will run in a separate thread until you stop them or the end of the data stream is reached.
>
> The data flowing through a pipeline consists of a combination of buffers and events. Buffers contain the data that will flow through the pipeline you have created. A source element will typically create a new buffer and pass it through a pad to the next element in the chain. The simple case is that a buffer is created, memory allocated, data put in it, and passed to the next element. That element reads the data, does something (like creating a new buffer and decoding into it), and unreferences the buffer. This causes the data to be freed and the buffer to be destroyed.
>
> The rtpjitterbuffer (jitter buffer) will wait for missing packets up to a configurable time limit using the "latency" property. Packets arriving too late are considered to be lost packets. The lost packet events are usually used by a depayloader or other element to create concealment data or some other logic to gracefully handle the missing packets. The jitterbuffer will use the Decode Time Stamp (DTS) or Presentation Time Stamp (PTS) if no DTS is set of the incoming buffer and the rtptime inside the Real-time Transport Protocol (RTP) packet to create a PTS on the outgoing buffer."

Buffer size is basically the size of memory allocation whereas Block size is the size of the chunks of data acted upon within the buffer.

The particular GStreamer pipeline implemented for this series of tests is shown in Figure 5. In this architecture, each of the four input channels can be muted or audio adjusted individually via four separate pipelines. They are then audio-mixed in the summing function. Here the four pipes are combined time-aligned and everything synchronized. This is all done in software and requires significant processing capability. A second master volume control is implemented along the new input chain and sent to a second combiner where the Audio Tones and Combined Audio Channels are mixed, again synchronized and time-aligned. The output of the second audio mixer is passed on to the Linux audio system, Advanced Linux Sound Architecture (ALSA) [4]

Architecturally, this implementation mirrors the audio requirement for mixing four independent audio communications streams and having the Audio Tones volume and controls independent of the communication channels as well

[4] ALSA is part of the Linux kernel that provides an Application Program Interface (API) for sound card device drivers.

as ensuring the Audio Tones are 13 dB above the background noise. However, due to the processing necessary to support the interaction of the Ethernet buffer drivers, the four audio communication channels and the mixing functions, the software was unable to keep up. The results are presented in 5.5.1 through 5.5.7.

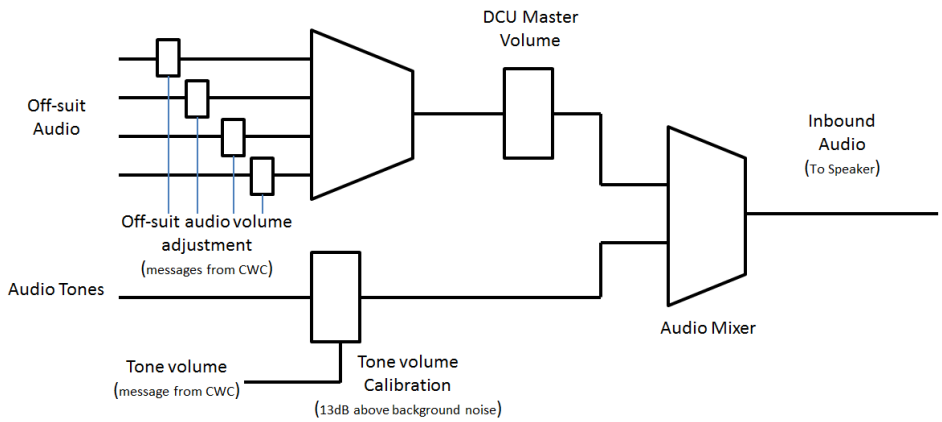


Figure 5: GStreamer Inbound Audio Processing Pipeline

5.5.1. Streaming Audio to Off-suit Nodes

The purpose of this test was to demonstrate effective movement of compressed streaming audio from the PAS Audio assembly to four off-suit radio nodes through the COMM assembly. Compressed streaming audio data created in the PAS Audio assembly, moves across the PAS bus through the PAS radio, across the Radio Frequency (RF) interface and into four radio nodes where it is decoded and played through the speakers of the off-suit nodes (192.168.3.111 - 3.114). This test demonstrated only the out bound direction of the bidirectional audio requirement of R2.1.420. Inbound is tested in another test.

- R2.1.420: (PAS2009) Full Duplex Voice Communication. The PAS Audio Assembly shall provide full-duplex voice communications to the crewmember.

Subsystem components (including PAS Audio assembly, COMM assembly, radio nodes, etc.) were powered-on and set-up per standard procedures including using a commanded suit pressure of 10 psia, 500 ms for jitter buffer size and 4096 bytes for GStreamer Block size. Wireshark software [8] was used for monitoring network traffic. Speech was input to the outbound PAS Audio assembly and was heard at each of the off-suit nodes. The speech produced at each off-suit node was intelligible. No discernible artifacts or distortion were observed. The test was considered successful and no issues were identified by the test. Successful test of outbound PAS audio assembly was based on GStreamer software and it is recommended that continued development should leverage this software.

5.5.2. Streaming Audio to Informatics Assembly

This test demonstrated effective movement of uncompressed streaming audio from PAS Audio assembly to the Informatics assembly. Uncompressed streaming audio data created in the PAS Audio assembly, moves across the PAS bus to the Informatics assembly, where it is archived. Uncompressed streaming audio data provides assurance of the highest possible level of audio fidelity is being sent to informatics for storage. This test demonstrated that the assembly is capable of meeting requirement R3.1.634 which pertains to voice field notes.

- R3.1.634: Field notes. The Informatics assembly shall record a crew field note consisting of: audio, still image or video data, location coordinates and timestamp.

The Subsystem components (including PAS Audio assembly, COMM assembly, radio nodes, etc.) were powered-on and set-up per standard procedures including using a commanded suit pressure of 10 psia, 500 ms for jitter buffer

size and 4096 bytes for GStreamer Block size. Wireshark® software [8] was used for monitoring network condition. Speech was input to the outbound PAS Audio assembly and captured by the Informatics assembly in full-fidelity LPCM format [9]. Successful testing of the outbound PAS Audio assembly was based on GStreamer software and it is recommended that continued development should leverage this software. Audio spoken into the outbound PAS Audio assembly was recorded on Informatics assembly and successfully played back.

5.5.3. Streaming Audio Latency

Audio latency testing characterizes the amount of end-to-end delay in the inbound and outbound PAS audio channels. Latency encompasses all delays in the channel including those associated with audio processing, coding and decoding of streaming audio data in the outbound and the inbound assemblies as well as the movement of the data through the PAS bus and the RF network. Applicable requirements for streaming audio latency include:

- R2.1.308: (PAS1218) Latency tolerance. This requirement limits overall PAS audio latency to be less than 100 ms delay for surface-to-surface crew-to-crew voice. This requirement is further decomposed into the following requirements:
- R3.1.512: Latency tolerance—radio. This requirement limits latency associated with the radio to < 20 ms.
- R3.1.513: Latency tolerance—system bus. This requirement limits latency associated with the PAS bus to < 50 ms.
- R3.1.514: Latency tolerance—Audio assembly. This requirement limits latency associated with the Audio assembly bus to < 30 ms.

One-way latencies of 200ms or less are considered acceptable for normal conversation. An allocation of 100ms total to each PAS subsystem on a voice loop will support a total voice-loop latency of 200ms, not including speed-of-light travel times associated with radio wave propagation. The test conducted here measured the amount of end-to-end Audio assembly latency including PAS bus and RF network. Latencies associated with the different components of the audio stream (i.e. Communication assembly, Audio assembly and the PAS bus) were not separately measured. End-to-end latency was measured using a computer with two microphones and a device that creates a distinctive pulse. To conduct a latency test a distinctive pulse was made at the microphone of one end of the voice loop and simultaneously observed by one of the microphones of the latency measurement system. The second microphone of the latency measurement system was placed at the speaker on the receiving end of the voice loop where the delayed pulse was observed. End-to-end latency was measured by observing the amount of time that elapsed between the pulses observed by the two microphones. The latency measurement system was performed using a netbook computer running Audacity, a multi-track audio editor and recorder. Pulses were observed graphically. The distinctive pulse was created by snapping fingers and the latency measurements appeared to be accurate to within a few milliseconds. This is shown in Figure 6 with the source tone on the bottom trace and the received pulse on the top trace. The delay is the measure time between to bottom and top pulse. The observed latencies, themselves, appeared to be subject to a significant amount of variation over time. A brief test procedure is provided below.

1. The Measurement, Test and Validation (MTV) netbook was used to measure latency. Microphones were connected to the MTV netbook, software was started and the ability to measure latency was validated
2. The subsystem (including PAS audio sub-assembly, COMM assembly, radio nodes, etc.) was powered-on and set-up per standard procedures including using a commanded suit pressure of 10 psia.
3. Latency measurements were made for both inbound and outbound channels for each of the radio nodes. In each case, six distinct latency measurements were made for purposes of averaging and collecting statistics on the measurements. Nominal values of 500 ms for GStreamer Jitter Buffer Size (gJBS) and 4096 bytes for GStreamer Block Size (gBS) were used in this test.
4. Latency measurements were made for different combinations of gBS and gBS to determine effect of these quantities on latency. These measurements were taken for a single radio node.

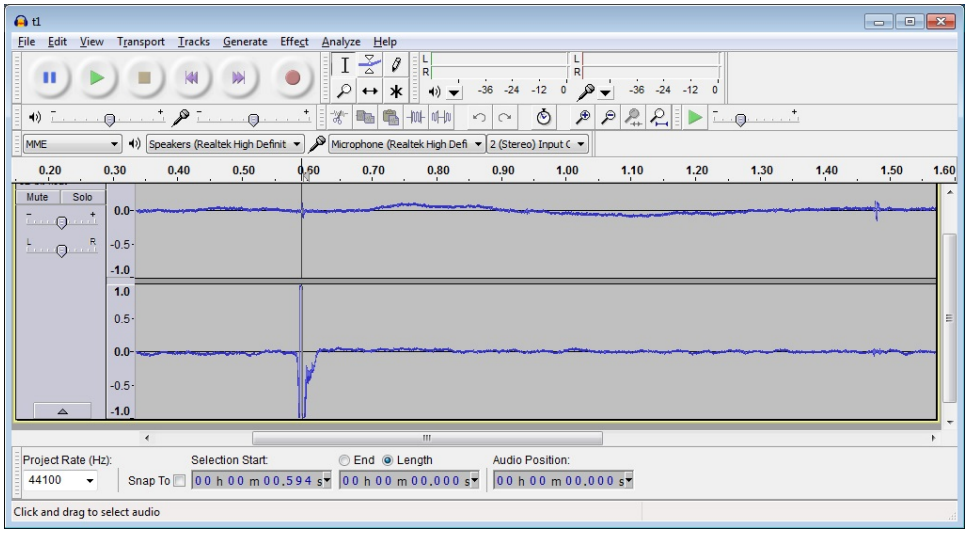


Figure 6: Audacity: Latency Measurement

Measured latency data from Step 3 are given in Table 1(a) for inbound channels and in Table 1(b) for outbound channels. Measurements of latency were made in direct succession in which all six measurements associated with a particular inbound or outbound channel were made sequentially (e.g. measurement 2 was made right after measurement 1 and measurement 3 was made right after measurement 2, etc.). In each case all six measurements associated with a particular channel were made over a period of about five minutes. There was no video or other high rate data going through the PAS bus during the time when these measurements were made. In Table 1(a) and Table 1(b) the average latency and standard deviation of the latency measurements are calculated in the penultimate and final rows of these tables, respectively.

Table 1: Latency measurements for inbound 1(a) and outbound 1(b) channels. Jitter buffer size and GStreamer block size are fixed at 500 ms and 4096 bytes, respectively. Measurements are given in milliseconds. Off-suit nodes are identified by the final octet of their network IP address.

Due to the large discrepancy between other data sets and the inbound channel to the PAS receive assembly from off-suit node 3.111 a second set of six datum were collected. This second data collection yielded an average latency of 914 ms with a standard deviation of 2.2 ms. Measured latency data from step 2 of the test procedure are displayed graphically in Figure 7(a) for the inbound channel of the PAS Audio assembly and in Figure 7(b) for the outbound channel of the PAS Audio assembly.

Figure 7: Latency measurements for inbound (a) and outbound (b) channels. Data represent single latency measurements for different levels of jitter buffer size and GStreamer block size. Data are for off-suit node 3.113. Measurements are given in milliseconds.

Averaging the data in Table 1 over off-suit nodes, the inbound latency is 832 ms with a standard deviation of 83.0 ms and the outbound latency is 846 ms with a standard deviation of 9.8 ms. Thus, the latency of inbound and outbound channels appears to be the same from a statistical perspective. A detailed examination of the data in Table 1 and Figure 7 reveals that there is a strong temporal dependence in the latency numbers. For example, examining the latency numbers for the channel between node 3.111 and the PAS inbound node 1.107 there is a significant jump in latency measurements between the first latency measurement and the remaining five measurements (a jump of about 20 ms). If the standard deviation estimate for these data is restricted to the last five latency measurements it drops from 9.8 ms to 1.9 ms. Similar jumps occur in each set of latency measurements. In particular data taken for the link from PAS Audio (outbound) to radio node 3.113 and the link from PAS Audio outbound to radio node 3.114 each have two jumps of approximately 20 ms. As discussed above, the second set of latency measurements for the inbound link from node 3.111 showed a discrepancy in average latency of nearly 200 ms while the second set of measurements had a very low spread. Further note that comparing average latency numbers for node 3.113 to those in Figure 7 (for gJBS =

Table 1: Audio for Off-Suit Nodes

(a) Inbound Audio

Off-Suit Nodes			
Msmt#	3.111	3.112	3.113
1	751	837	853
2	730	822	854
3	725	817	859
4	729	819	853
5	729	819	836
6	729	792	816
Avg	732.2	817.7	845.2
Std	9.4	14.5	16.3

(b) Outbound Audio

Off-Suit Nodes			
Msmt#	3.111	3.112	3.113
1	872	837	845
2	831	837	849
3	830	862	875
4	829	861	873
5	829	864	842
6	808	856	841
Avg	833.2	852.8	854.2
Std	20.4	12.5	15.6

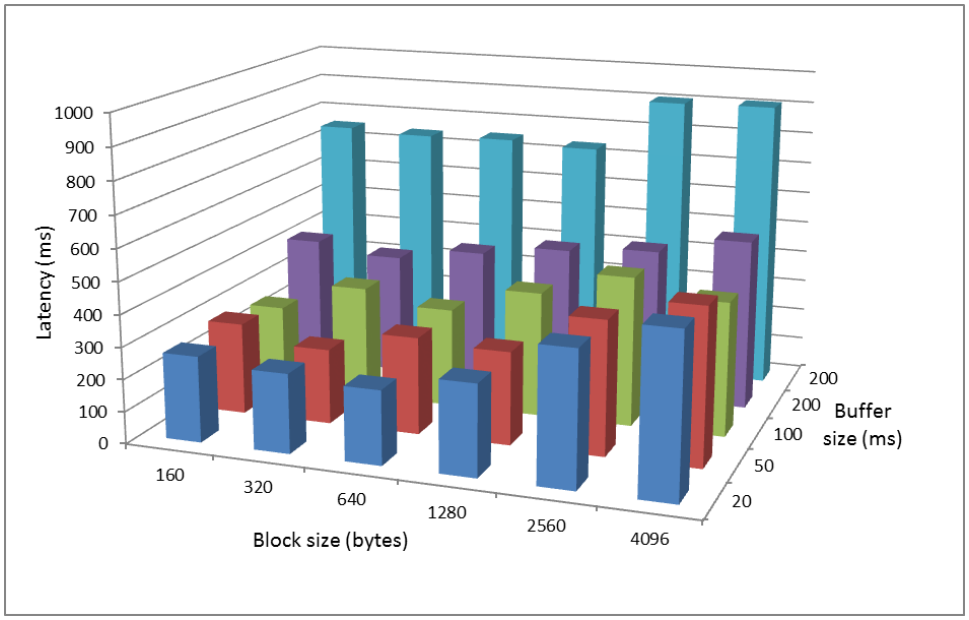
500 ms, gBS = 4096 bytes) shows discrepancies of approximately 60 ms for the inbound channel and approximately 40 ms for the outbound channel. Channel latency appears to be very stable over a period of some tens of seconds with latency measurements varying only a few milliseconds over such a period of time. However, over time, significant jumps in the latency associated with a particular channel are observed. These observed jumps range from 20 ms to nearly 200 ms.

Examining the data in Figure 7 reveals that, with a few notable exceptions, there is a general increase in latency with block size and jitter buffer size. This is true for both inbound and outbound channels. A casual fit of the latency data in Figure 7 to gJBS and gBS results in expressions for the inbound latency (Lin) and outbound latency (Lout) in milliseconds:

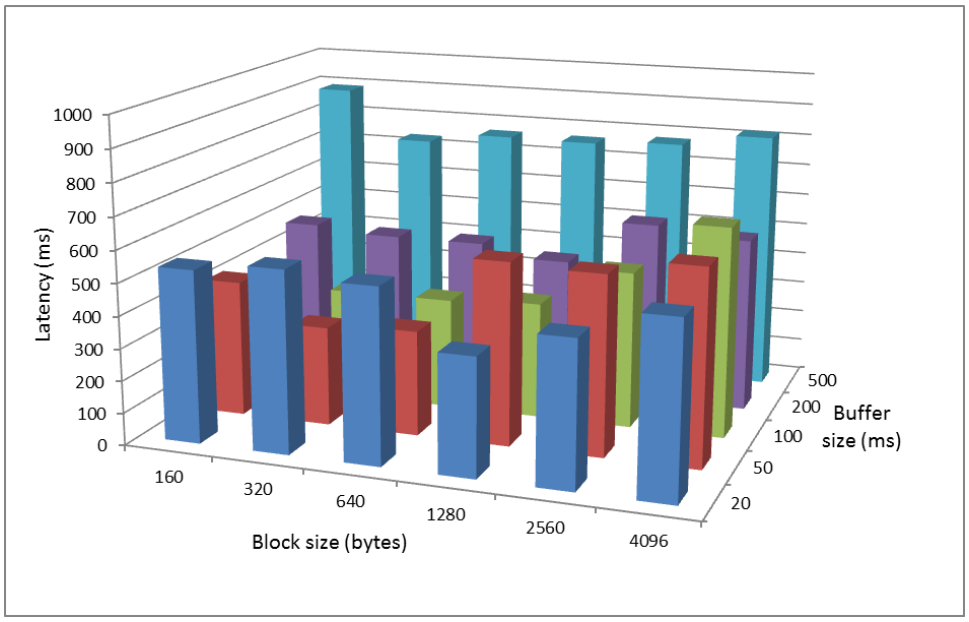
$$Lin \approx gJBS + \left(\frac{gBS}{20}\right) + 200 \qquad (1)$$

$$Lout \approx 0.7gJBS + \left(\frac{gBS}{30}\right) + 360 \qquad (2)$$

These formulae produce an RMS fit error of approximately 41 ms for and 88 ms for which are within the variability in latency measurements observed over time, presumably due to networking and buffering issues. The data, together with the formulae, indicate that there are no possible combinations of gJBS and gBS that will produce an end-to-end



(a) Inbound Audio



(b) Outbound Audio

Figure 7: Audio Latency

latency of less than 200 ms. For the outbound channel of the PAS Audio assembly, the audio signals were intelligible. No discernible artifacts or distortion were observed at the off-suit nodes outputs. However, significant degradations in audio quality occurred for the inbound channel of the PAS Audio assembly. The degradation in the inbound channel is a progressive 'breaking-up' of the audio in which gaps in the audio signal became progressively more frequent until the signal became unintelligible and, eventually, the inbound channel of the PAS Audio assembly became silent. The amount of time for the inbound audio signal to degrade was mainly a function of gJBS and is shown in Table 2.

However, a value of gBS of less than 320 bytes also causes significant degradations in audio quality for the inbound PAS subsystem regardless of gJBS. Note, these results are specifically for the GStreamer architecture implementation shown in Figure 5

Table 2: Degradation time of Inbound audio signal as a function of gJBS

Buffer size (ms)	20	50	100	200	500
Time to Break-up (minutes)	<1	<1	~1	~5	30

As noted above, the overall end-to-end latency requirements were not met. It is not clear which portion(s) of the PAS Subsystem are deficient. Additionally, large jumps in channel latency are observed in the data. Changes in PAS bus and RF network congestion are candidate causes for the variable latency. Internal issues with GStreamer and or the Linux operating system buffers are suspects for both the oversized latency numbers as well as the variability of the latency numbers. It is worth noting that the end-to-end streaming audio data delivery was accomplished using UDP (without retries) yet the RF network uses significant amounts of Automatic Repeat-reQuest (ARQ). Therefore, significant potential exists for inter-network interactions that may compound problems with network congestion. Future testing should include monitoring the RF network. Suggest using AirPcap ® for monitoring the RF network.

Additionally, in order to diagnose the issues with latency, these tests could be repeated without the RF network. In this case the "off-suit nodes" could be connected directly to the PAS bus and latency would be measured. Also, monitoring of the network conditions using Wireshark or other devices could eliminate the PAS bus network and the RF network from consideration as the cause of the large end-to-end system latencies. Such devices may prove useful in determining if the PAS bus and the radio are meeting their allocations of the overall end-to-end latency requirements.

The current version of GStreamer used in the off-suit nodes and on the PAS Audio assemblies is version 0.10. However, version 1.2 of GStreamer software now exists. Reports indicate that the newer version of GStreamer has fixed many bugs in the GStreamer buffering system. The older version should be replaced with the newer version.

Finally, moving to a different set of software or hardware may be needed in order to resolve system latency issues in the current Audio assembly. The GStreamer software is intended for use with 'generic' streaming media and is intended to work well with either video streams, audio streams, or blended streams. Moving to a set of software that is more specific to only streaming audio may produce better results. For example, the BIONET ® software [10] used in the radio network for the Desert Research and Technology Studies (DRATS) assembly was written specifically for streaming audio. System latencies in the DRATS subsystem were not considered to be excessive [11]. Alternate hardware technology includes consideration of an alternate processor, or more likely, an alternate set of peripheral chips (e.g. Ethernet Physical Transceiver (PHY) chips) or an Field Programmable Gate Array (FPGA) implementation of streaming audio for the PAS assemblies. Such software changes may provide a remedy for the gradual degradation that is observed with audio quality as well as the eventual crashing of the audio streams on the inbound PAS audio assembly. As large, yet infrequent, variations in channel latency may not be detected by speech intelligibility testing, further investigation is needed to determine if these variations are tolerable and, if not, changes in channel latency should be addressed by separate requirements.

5.5.4. Inbound Audio Stream Mixing

The purpose of this test was to demonstrate effective movement of compressed streaming audio from the off-suit nodes to the PAS Audio assembly and that these streams are properly mixed and muted according to switch settings sent from the CWCS assembly. Compressed streaming audio data created in each of the off-suit nodes, moves through the wireless network and RF interface and then through the PAS bus to the PAS Audio assembly, where it is decoded, mixed with other streams and played through the speakers of the inbound portion of the PAS Audio assembly. This test demonstrated that the PAS Audio assembly meets both the inbound portion of requirement R2.1.420, but not the outbound portion, and meets R2.1.413 requirement with a modification to four channels instead of five.

- R2.1.420 (PAS2009) Full Duplex Voice Communication The PAS Audio Assembly shall provide full-duplex voice communications to the crewmember.

- R2.1.413 (PAS2002) Audio Mixing. The PAS Audio Assembly shall mix audio signals from up to five separate sources - To Be Resolved (TBR) <TBR-PAS-005>

Subsystem components (including the PAS Audio assembly, COMM assembly, radio nodes, etc.) were powered-on and set-up per standard procedures including using a commanded suit pressure of 10 psia, 500 ms for jitter buffer size and 4096 bytes for GStreamer Block size. Wireshark software was used for monitoring network traffic. Audio streams were created by each off-suit node and the mute flags and volume controls were exercised using the CWCS GUI for each inbound channel.

Audio from each off-suit node was observed. Response to changes in volume controls and muting buttons in each of the channels was appropriate. The audio signals were intelligible. No discernible artifacts or distortion were observed on the PAS audio output under nominal conditions. The test is considered successful. However, during subsystem setup it was observed that the entire inbound PAS Audio assembly would cease to function in cases where a stream from an off-suit node would cease for reasons such as unintentional power-down or disconnection from the network. Such a lack of robustness in the overall audio function is not desired.

In order to remedy the crashing of the inbound audio processing the addition of messaging from the off-suit nodes or COMM assembly to the inbound audio process indicating the absence of transmissions may help. General changes of inbound audio GStreamer configuration may also help here. The crashing of the inbound audio processing was unexpected. Therefore, it is recommended that a requirement be added for robust operation of the audio system in cases where inbound audio streams stop and start spontaneously. Alternately, existing requirements regarding streaming audio could be rephrased to explicitly include cases in which operation must seamlessly resume and recover following network interruption. The mixing tests included no cases in which *combinations* of inbound streams are observed. In the future, the audio mixing tests should not be restricted to observing a single audio stream at a time. For four channels there are 15 different non-trivial combinations of audio streams. Some subset of these combinations should be tested.

5.5.5. Audio Tones

The purpose of this test was to demonstrate effective creation of tones as commanded by the CWCS assembly. Various conditions exist where starting and stopping of the tones is commanded by the CWCS assembly[5]. This test demonstrated that the PAS audio assembly meets requirement R1.110.

- R1.110 Performance Warnings. The AEMU shall detect the following failure conditions and alert the crewmember of the failure by a simultaneous audible tone and display. The tone shall deactivate after 10 seconds and shall be common to all warning conditions: 1) High oxygen flow, 2) Low Suit pressure, 3) Low vent flow, 4) Low water flow, 5) Low feed-water quantity, 6) High CO2 and 7) Battery Volts low.

For the Tone Control tests, all tones should be generated per Table 3. Testing of tones was done with all inbound audio channels muted. The tone priority function was also tested during this test. Tones of a higher priority must preempt tones of a lower priority and tones of a lower priority must not preempt tones of a higher priority. Note, that a redundant interface between the audio sub-assembly and the CWCS sub-assembly was not implemented. As such, the function of a second communications pathway between CWCS and audio was not tested.

Tone volume should be about 13 dB higher than the assumed NC60 noise profile for inside of the suit, yielding a required Sound Pressure Level (SPL) of approximately 73 dBA [12]. This was accomplished using an SPL meter with the volume control set by the CWCS GUI. Data were collected using Wireshark. The volume of the tones was adjusted to ensure that tones were generated with sufficient volume so that they can be heard above suit noise level. A shell script program was used on the CWCS GUI computer and the off-suit nodes to generate the appropriate messages and audio streams for the test. The appropriate responses to tone messages received by the inbound PAS Audio assembly is observed by listening to the inbound PAS audio.

The subsystem setup was performed according to test procedure found in the Appendix A. The required SPL of the tones was achieved. With the exception of the stopping of the warning tone, all tones were observed as expected. For the particular GStreamer pipeline architecture used during this test, the warning tone seemed to "slow down" and stop at about 3.5 minutes into its required duration of 5 minutes. Status tones were able to pre-empt information tones and alert tones were able to pre-empt status tones.

[5]NOTE: The warning and alert tones characteristics are from Table 10.8-3 Shuttle Specifications for Class 1-3 Alerts.

Table 3: Implementation of audio alert tones for PAS audio assembly

Tone (Priority Order is Low to High)	Type	Duration	When Activated
Information	Single single tone, 400Hz, with 50% duty cycle (on 1s, off 1s)	30 seconds, interrupt-able by acknowledge message	Fault message from buddy crew member displayed on Informatics display (30 seconds unless unless acknowledged by crew member)
Status	Continuous single tone of 1100Hz,	5 seconds	• Failed Leak Check • Failed Automated Checkout • Secondary O2 Time Left < 6 Minutes
Alert	Continuous single tone of 550Hz	0.5 seconds	• Start of Leak Check • Start of Automated Checkout • Successful Leak Check • Successful Automated Checkout • Airlock pressure, P = 6.0 during depress • Airlock pressure, P= 4.0 during repress • Airlock pressure, depress & repress start/stop
Warning	Warble, Siren, varying frequency from 666Hz to 1470Hz and return over 5 second period	5 minutes, interrupt-able by acknowledge message	• BITE Circuitry Activated • Fault message displayed on text display (5 min. unless DISPL switch is pressed to PROC position or anomaly disappears)

It is recommended to debug the warning tone duration issue. General changes to inbound audio GStreamer configuration may help here. It is also recommended that requirements for SPL of tones and tone priority be developed. Consideration of a second communications path between the audio sub-assembly and CWCS should be developed and tested. Testing of the redundant interface should include disabling each interface, in turn, and checking for proper operations of the tone function. Procedures for disabling the Audio/CWCS interface should, at a minimum, include a physical disconnection of the each cable that carries data between CWCS and the audio sub-assembly.

5.5.6. Audio Outbound Pressure Compensation

The purpose of this test was to demonstrate effective compensation for variations in suit static pressure. Compensation for the effects of static pressure on crewmember vocalization is required by requirement R2.1.412.

- R2.1.412 (PAS2001) Suit Pressure Audio Compensation: The PAS Audio Assembly shall compensate for variations in crewmember speech level associated with changing suit pressure levels.

Data in Clark et. al.r "The Effects of High Altitude on Speech" [12] indicate that the crewmembers speech in SPL reduces by about 15 dB as the altitude changes from sea level to an altitude of 30000 ft. The corresponding change in static pressure is $14.7 - 4.3 \approx 10$ psia. Data discussed in the Clark paper also indicate that changes in vocalization level vary in direct proportion with changes in static pressure. The pressure compensation function, Gpc, counteract losses in vocalization level for static pressure levels below atmospheric and pass signals unchanged at normal atmospheric conditions. The gain (in dB) needed to counteract the reduced SPL brought on by lower static pressure is, therefore: $Gpc = (14.7 - P) * 15/10$, where P is the in-suit pressure (in psia). The essential feature of this function is that a drop of 1 psi in static pressure produces a 1.5 dB increase in signal gain. Extension of this function beyond hyperbaric conditions provides attenuation, but data is not available for vocalization in hyperbaric environments. As such, providing attenuation is not an unreasonable approach. This pressure compensation gain is implemented in the outbound audio stream of the PAS Audio assembly. While pressure compensation is, currently, only implemented as a broadband gain (not frequency-dependent) future versions will include linear filtering of the outbound signal to more fully compensate for the effects of varying static pressure on crewmember vocalizations.

Testing was accomplished by simulating changes of the in-suit static pressure level. In this test the CWCS assembly sent suit pressure messages to the PAS Audio assembly. The changes in the level of the audio stream coming from the PAS outbound Audio assembly were observed with every CWCS GUI pressure change. During the test, a constant tone at a constant SPL into the PAS outbound audio stream was maintained by placing the microphone of the PAS outbound Audio assembly in a microphone calibrator shown in Figure 8(a). The microphone calibrator produces a 1 kHz tone at 94 dB SPL. Also, the volume control of the off-suit node was kept at a fixed level. The level of the outbound signal was, therefore, only dependent on the pressure compensating gain applied to the outbound channel by the PAS assembly. Again, the gain applied to the microphone varied inversely with suit pressure. If the suit pressure is reduced, the gain of the microphone is increased to compensate. If the suit pressure is increased, the gain of the microphone is reduced. This is intended to provide a constant SPL from the microphone, regardless of suit pressure. Observations showed a good correlation of speaker SPL versus commanded static suit pressure that closely matched the expected results from the Clark paper.

A sound pressure meter shown in Figure 8(b) was used to measure the level of the outbound signal at a distance of 1m from the four off-suit nodes (192.168.3.111 through 192.168.3.114) that produced the aural signal. The measured SPL of output audio produced by the off-suit node is shown by the blue line, with measured data points, in Figure 9. Also included on Figure 9 is an affine (linear) function that is fit to the data and its associated formula.

The observed pressure compensation system response follows the expected affine function and provides a slope of -1.4464 dB/psi which is very close to the slope of the pressure compensation gain function of -1.5 dB/psi. The essential feature of the pressure compensation function is, thereby, demonstrated. The "intercept" of +98.2 dB depends on internal system gains internal to the PAS Audio assembly and the off-suit node and volume control of the audio output of the off-suit node. In spite of the high level of acoustic SPL used for the outbound PAS signal (94 dBA) there is no evidence that the pressure compensation has saturated the dynamic range of the system.

This test successfully demonstrated gain-only pressure compensation for sub-atmospheric static pressure levels. Follow-on development will include frequency-dependent pressure compensation. Testing of the frequency-dependent compensation function will need to be developed and performed to demonstrate compliance. Future testing should include better testing for saturation (e.g. measurement of Total Harmonic Distortion (THD)). However, such testing may be included in speech quality testing. Also, verification of the extension of the pressure compensation law to hyperbaric conditions should be made. Requirements for pressure compensation should be modified to discuss hyperbaric conditions (if appropriate). Detailed requirements on processing could be developed based on the Clark paper[6] and/or additional testing with people in altitude chambers.

[6]The Clark paper is more than 65 years old.

(a) Microphone Calibrator

(b) Pressure Sensor Meter

Figure 8: Audio Test Equipment

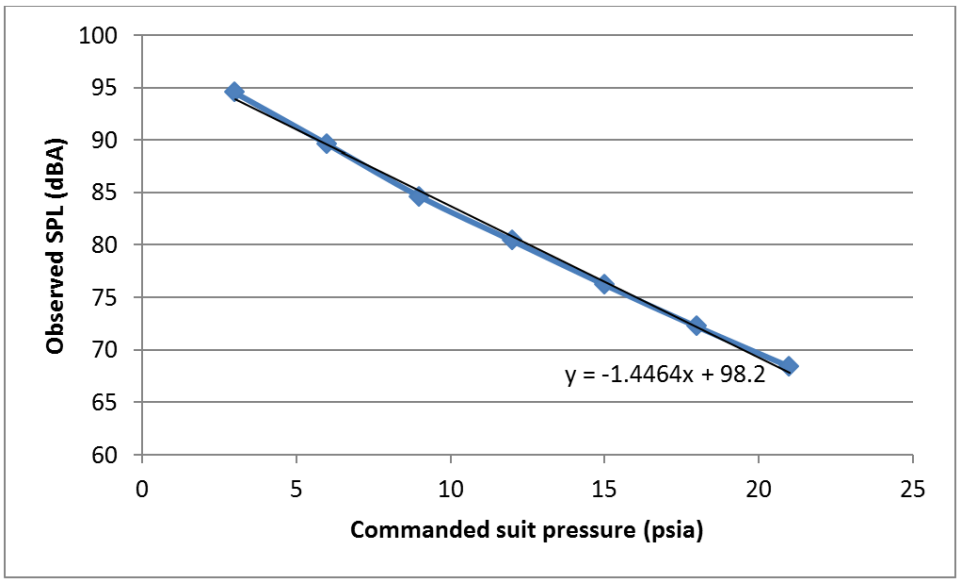


Figure 9: Commanded Audio assembly output level. Acoustic SPL input to PAS Subsystem and output volume control were held constant

5.5.7. Transmit Mode Functions

The purpose of this test was to demonstrate effective voice mode functioning of the outbound PAS Audio assembly per requirement R1.103.1.

- R1.103.1 Voice Modes: The voice transmission system shall have push-to-talk, voice activated, and off (receive only) modes that are controllable by the crewmember.

Three different modes of the audio are required including Voice Operated Switch (VOX), Push-To-Talk (PTT) and Off (receive only) mode. The outbound audio mode is controlled through switches on the Audio assembly. The testing consists of generating audio streams on the outbound PAS Audio assembly and observing the audio stream at an off-suit node while exercising the transmit modes and PTT switches. Implementation of the transmit mode in the outbound audio assembly included a PTT mode as required. However the Off (receive only) mode was not implemented. In place of the Off (receive only) mode, an 'Open Mic' mode was implemented. This was done because

of the fact that an Off mode would be redundant to the PTT mode for the case in which the PTT switch (or microphone switch) was not depressed.

Also, as the PTT switch is considered to be a momentary (push button) type of switch (as would be the case for a mobile radio), the default position is the unpressed (or unkeyed) position. Therefore, it was decided to adopt a keyed-mic mode function switch position to account for situations when one needs to key the microphone for extended periods without holding the momentary push-button down for those extended periods and freeing one's hand for other uses. In the case where the PTT switch is of the push-push or toggle variety (as is normally the case for a speaker phone) the third (keyed-mic) mode is unnecessary.

When the audio mode was set to the open-mic mode the PTT switch had no effect on the observed audio stream. When the transmit mode switch was set to the PTT mode setting the PTT switch blocked the audio stream when it was not engaged and allowed audio to be played on the off-suit nodes when engaged. When the transmit mode was set to VOX, audio flowed regardless of speech level. The testing for the keyed-mic and PTT modes was successful. For these PAS 1.0 subsystem integration tests, VOX mode had not yet been implemented. Thus, testing of the VOX function indicated a failure.

With the advent of a modern Voice-Over-IP (VOIP) style of digital voice transmission implemented on a modern space suit, there are numerous possibilities for transmit mode control. Indeed the use of digital telephony allows for the case that appropriate control of the suit audio outbound stream is accomplished entirely on the receive side of the loop and on a receiver-by-receiver basis albeit at the expense of using additional bandwidth (i.e. more and continuous bits on the RF link). For such a system there would be no need for on-suit controls like the transmit mode switch or the PTT switch. Alternately, a wide array of transmit mode control possibilities could be implemented with mode selecting using a soft-switch (or menu) in combination with a single, hardwired transmit control switch. Therefore, Requirement R1.103.1 should be revisited and considerations taken to modernize the capabilities to reflect those available in newer VOIP technologies.

5.5.8. Audio Post Test Results

Many of the audio tests were unsatisfactory. Many of those results had to do with misunderstanding of the GStreamer framework and the particular implementation used. Following the initial tests, the audio pipeline was redesigned. The final architecture is shown in Figure 10.

For the audio voice streams, instead of having two audio mixers inline as in the original implementation shown in Figure 5, four independent audio pipelines were implemented along with one pipeline for the caution and warning tones. These pipelines consisted of a UDP receiver and filter element followed by a jitter buffer and simple queue. The jitter buffer smooths out any delay variations that may occur in the network. In our case, most of those variations would be a result of packet retries and congestion on the wireless link. The GSteamer depayload element removes Ethernet framing and presents the raw payload to the *u*Law decoder which decodes the G.711 audio stream. The volume is then adjusted and the individual pipeline passed to the Linux kernel.

The caution and warning tone pipeline consists of a Tone Source module, Ts, to generate caution and warning tones and a volume control module. The Tone Source module is a program written in C that generates tones using the GStreamer audiotestsrc plugin. The audiotestsrc plugin can be used to create audio test signals of given frequency and volume.

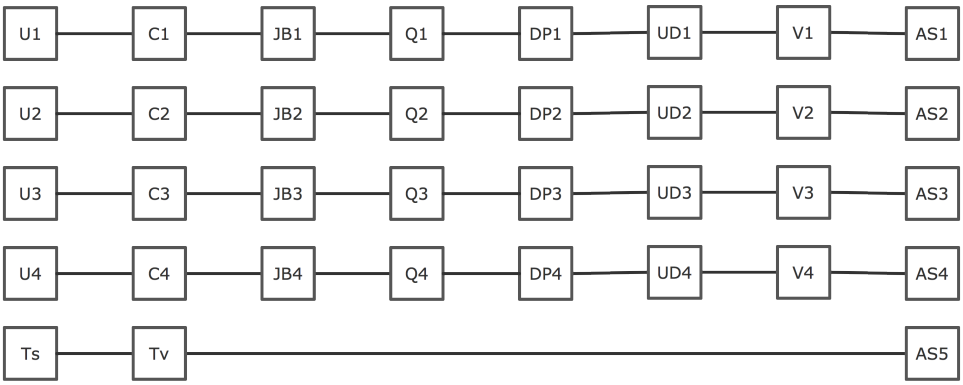
With this implementation, there were no distortions or breakups of voice or audio tones over time. Thus, almost every anomaly for the implementation in Figure 5 was fixed with the exception of delay. The implementation still required a significant jitter buffer resulting in unacceptable delay. Furthermore, there were still intermittent discontinuities in delay where a jump of 20 msec would periodically occur.

A few questions remain:

1. Why did the alternate inbound implementation work better than the original? and
2. Why are there still problems with delay and delay discontinuity?

Regarding the first question, it appears that the GStreamer audio mixer takes each separate pipeline and time aligns the audio streams then recombines them into a single stream. This takes significant processing - particularly when done in software as is the case here. It appears the hardware used did not have sufficient processing power to handle the

audio mixing. The explanation for delays and jumps in delay remains the same as presented in the latency discussion of section 5.5.3 Also note, that a search of user groups and bug reports indicated that there were numerous known bugs in the GStreamer jitter buffer implementation in the software version utilized, GStreamer 0.10.36. Whether or not this contributed to the problems originally observed is purely conjecture.



U = udpsrc— Receive data over the network via UDP
C = capsfilter — Pass data without modification, limiting formats
JB = rtpjitterbuffer — A buffer that deals with network jitter and other transmission faults
Q = queue — Simple data queue
DP = depayload — implements generic functions such as buffering, reordering and dropping packets
uD = uLaw Decoder (G.711)
V = Volume
AS = Advance Linux Sound Architecture (ALSA} Sink
Ts = Tone Source
Tv = Tone Volume

Figure 10: Alternate Inbound Implementation

5.6. Video Stress Test

The intent of the Video Stress Test was to examine network behavior during periods of highest data loading caused by video streaming. Video stream rates are expected to be as high as 15 Mbps for high quality HD at 30 fps. These rates are orders of magnitude higher than all other bus traffic that may exist in the currently envisioned PAS 1.0 subsystem. As such, video has a greater potential to interfere with other system traffic. In order to identify possible interactions, a video stream was sent over the GigE bus while other test are being performed to ensure there is sufficient bandwidth on the bus to allow all necessary communication between subsystems and that the quality of communications remains intact.

The video source was a SANYO VCC-HD4600 high definition IP Camera. The Sanyo camera was chosen as a close functional and performance approximation to the camera currently envisioned for the suit. The purpose of the suit camera is for record keeping and to provide situational awareness to ground support and other mission assets during EVA. It could also be used for public relation events and potentially for procedure close out verification of EVAs. Thus with a wide range of uses, the camera must have broad flexibility in resolution, frame rate, and the ability to generate multiple video streams simultaneously.

The Sanyo camera was attached to the NetGear Gigabit switch, the PAS bus, on one of the available ports. It was given address 192.168.1.106. The video sink (or destination) was displayed on a computer attached to the NetGear Gigabit Ethernet switch on another port. It was given address 192.168.1.116. The SANYO camera provided client software both to configure the camera and to display the video via a Web-based application interface.

Two basic tests were performed. The first put video on the bus over a variety of rates using HyperText Transfer Protocol (HTTP). The second put that same video over the bus using UDP Multicast, but only at the highest rate. Note, the maximum data rate we could get out of this setup was 8 Mbps for a single stream and 8.250 Mbps for two streams. Table 4 shows the various settings for each of 6 different tests performed.

Table 4: Video Stream Data Rates

		Stream 1			Stream 2		
Test	Transport	Coding	Resolution	Bit Rate (kbps)	Coding	Resolution	Bit Rate (kbps)
1	HTTP	H.264	1920x1080	560			
2	HTTP	H.264	1920x1080	1000			
3	HTTP	H.264	1920x1080	4000			
4	HTTP	H.264	1920x1080	8000			
5	HTTP	H.264	1920x1080	8000	H.264	320x180	250
6	Multicast	H.264	1920x1080	8000			

For tests 1 through 5, adding video to the PAS bus did not affect any of the other assemblies even though video dominated the overall transmission through the switch. This was to be expected, as the bus is a switched bus. Thus, the video was never seen by any of the other assemblies, as the transport was HTTP. The only ports that received this video were the client video display PC and the mirrored test and monitoring port. For test 6, an 8 Mbps video stream was sent using UDP multicast. This placed the video stream on all ports. Still this did not affect the performance of any of the other assembly as those assemblies were not listening for the multicast data and thus ignored it. Many of the subsystems were only capable of 100 Mbps Ethernet. Even so, 8 Mbps does not come close to stressing that bandwidth capability. Even with only 8 Mbps of video, video traffic dominated the overall link utilization. In all cases video was observed at the destination monitor without any image artifacts or dropped frames.

The original test plan called for testing at a maximum video data rate of 20 Mbps. However using H.264 compression the maximum single stream data rate was 8 Mbps for the Sanyo camera. The 8 Mbps limit is a function of the H.264 encoder used on the Sanyo camera. Future testing for video data streams could utilize the camera's Motion Joint Photographic Experts Group (JPEG) format, which can produce stream rates in excess of 20 Mbps as Motion JPEG makes no use of any video compression techniques to reduce the data. Rather, Motion JPEG is a series of still,

complete images. The resultant data stream has a relatively high bit rate or low compression ratio for the delivered quality compared with video compression standards such as Motion Picture Experts Group (MPEG)-4 and H.264 [13].

The original test plan called for evaluation using two simultaneous video streams on the network, which would simulate a configuration where the camera is streaming video to more than one node. Attempts to establish two simultaneous video stream links was not successful, due to a procedural misunderstanding and was not tested. If the PAS architecture continues to evolve with configurations where two simultaneous video steam links are likely, then the test could be repeated.

The video source was a relatively inexpensive IP camera used mainly for commercial surveillance. It was convenient to use as it has an Ethernet connection and web-based applications for display and configuration. But, it has limitations in simulating the maximum desired data rates in H.264 format. As an alternative to using an IP Camera, basic bus stress testing could be done using IPerf [14], nuttcp [15] or ttcp [16] sending UDP multicast traffic. This would provide better control regarding data rates and data-rate range. A switched fabric will definitely isolate the video traffic from subsystems that do not need to see it. In future iterations of the PAS data bus, it may be more appropriate to simply put the video traffic on its own link, as high definition video will likely be sent directly to storage for retrieval at the conclusion of the sortie. A dedicated link between video and storage may allow the voice and messaging traffic to be sent between subsystem over a much lower-speed and lower-power bus.

5.7. Network Performance

Network throughput, jitter, and packet/datagram loss are primary metrics for evaluating the performance of a network. Network throughput is a measure of the maximum data rate that may be achieved over the network. Jitter is the time-derivative of the arrival time of the network packets/datagrams, which is important for streaming data. Datagram loss is also measured, which is a percentage of the UDP datagrams [17] that were lost during transmission. The following network performance tests used an open-source tool called IPerf [14] to obtain these values for the PAS system, which flooded the network with various prescribed amounts of UDP/IP traffic in order to compute the metrics. Three machines were used for these tests; the radio testbed, a computer on the wireless network that represents another suit or an access point, and a computer on the PAS bus that represents any PAS assembly. Note, the 802.11 wireless network was in ad hoc mode (a.k.a independent basic service set (IBSS)) [18]. Various configurations of client/server pairs were tested.

The following tables contain the network performance measurements for the various configurations of tests. For each network configuration, several throughput levels were attempted, and these levels were selected to demonstrate the limits of each link.

Table 5 contains the measurements of the Ethernet link from the computer that emulated a PAS assembly to the Radio Testbed. As expected of any Ethernet link, the jitter and loss increase as the network traffic is increased due to the buffering and receive processing of the data (e.g. between 192.168.1.122 and 192.168.1.105 in Figure 1). The relatively low throughput for this test shows that the BeagleBone is not capable of receiving data from the network at the full advertised rate of 100 Mbps. Table 6 shows the same measurements as the first test, but in the reverse direction. The limiting factor in this test was how fast the BeagleBone can generate network traffic. The Personal Computer (PC) on the receiving end had no issues processing the traffic. Table 7 is a test in which the first two tests are performed simultaneously. Since these links are full-duplex, the network traffic flowing in one direction does not affect the network performance in the other direction.

Table 5: Network Performance Results, PAS Assembly to Radio Testbed

Attmp'd Throughput (Mbits/s)	15	30	45	60
Measured Throughput (Mbits/s)	13.1	24.7	34.7	6.0
Jitter (ms)	0.02	0.02	8.30	493.10
Datagram Loss	12%	18%	21%	89%

Tables 8-10 show the results of tests similar to those in the first three tables, but these three tests use a PC connected to the Radio Testbed over the wireless link instead of the PAS bus (e.g. between 192.168.3.111 and 192.168.3.105

Table 6: Network Performance Results, Radio Testbed to PAS Assembly

Attmp'd Throughput (Mbits/s)	15	30	45	60
Actual Throughput (Mbits/s)	14.9	29.2	42.1	50.9
Jitter (ms)	0.04	0.04	0.03	0.07
Datagram Loss	0%	0%	0%	0%

Table 7: Network Performance Results, Bi-directional between Radio Testbed and PAS Assembly

Attmp'd Throughput (Mbits/s)	15	30	45	60
Fwd. Throughput (Mbits/s)	13.4	23.6	35.8	4.31
Return Throughput (Mbits/s)	15.0	30.0	45.0	60.0
Jitter (ms)	0.02	8.98	0.02	555.90
Datagram Loss	11%	19%	21%	93%

(COMM) in Figure 1). Notice that the jitter values are generally higher over wireless links and the throughput is limited by the mode in which wireless network is operating. This was expected due to the nature of wireless networking and 802.11. Jitter is introduced because datagram transmissions are often delayed by random periods of time while the wireless channel is occupied by another wireless device. The throughput is limited by the signal strength of the links. The higher attempted throughput cases in Table 9 indicate that the BeagleBone platform was not as efficient in handling heavy traffic loads as were the more computationally powerful netbooks.

Table 8: Network Performance Results, Wireless Node to Radio Testbed

Attmp'd Throughput (Mbits/s)	5	15	30
Actual Throughput (Mbits/s)	4.2	4.5	4.39
Jitter (ms)	0.90	9.96	14.18
Datagram Loss	2%	30%	6.3%

Table 9: Network Performance Results, Radio Testbed to Wireless Node

Attmp'd Throughput (Mbits/s)	5	15	30
Actual Throughput (Mbits/s)	5.1	0.93	0.71
Jitter (ms)	6.09	50.82	48.50
Datagram Loss	0%	6%	4.8%

Tables 11-13 show the results of testing the system end-to-end with the Radio Testbed as the middle node between the wireless netbooks and the PAS bus PC (e.g. between 192.168.3.111 and 192.168.1.122 in Figure 1). The lower data rates of the wireless interface limit the end-to-end throughput. An interesting case is shown in Table 12. When 12 Mbps of traffic was sent over the bus to the radio for re-transmission over the wireless link, the data was lost due to the mismatch between wired and wireless data rates. Information was being sent to the Radio Testbed faster than it could re-transmit it over the wireless link, therefore the Radio Testbed experienced buffer overflows.

Table 10: Network Performance Results, Bi-Directional between Radio Testbed and Wireless Node

Attmp'd Throughput (Mbits/s)	5	15	30
Fwd. Throughput (Mbits/s)	4.3	3.8	5.1
Return Throughput (Mbits/s)	4.5	4.0	5.3
Jitter (ms)	21.35	14.01	13.03
Datagram Loss	3.4%	4.6%	2.8%

Table 11: Network Performance Results, Wireless Node to PAS Assembly (through Radio Testbed)

Attmp'd Throughput (Mbits/s)	4	8	12
Actual Throughput (Mbits/s)	4.1	5.3	5.1
Jitter (ms)	3.05	0.09	12.01
Datagram Loss	0.76%	0.49%	0.50%

Table 12: Network Performance Results, PAS Assembly to Wireless Node (through Radio Testbed)

Attmp'd Throughput (Mbits/s)	4	8	12
Actual Throughput (Mbits/s)	4.0	7.9	0.5
Jitter (ms)	7.06	6.52	31.71
Datagram Loss	0.24%	0.03%	78%

Table 13: Network Performance Results, Bi-Directional between PAS Assembly and Wireless Node (through Radio Testbed)

Attmp'd Throughput (Mbits/s)	4	8	12
Fwd. Throughput (Mbits/s)	4.0	4.2	4.8
Return Throughput (Mbits/s)	4.0	4.23	4.86
Jitter (ms)	3.92	12.83	0.95
Datagram Loss	0.47%	1.9%	1%

Overall, the Radio Testbed successfully exercised all of the functionality that was required during the integration testing campaign. The PAS bus and the wireless network were successfully bridged, and the Radio Testbed hardware was more than sufficient to handle the network traffic flows that were presented to it during these tests. The Google protocol buffers and ZeroMQ software worked well and were a very efficient means of providing a PAS telemetry capability to the subsystem.

The following general conclusions can be drawn from the tables above:

- The wireless link has a much smaller bandwidth than the PAS switched Gigabit Ethernet bus, but at this point in the project's development, it would be wise to provide for growth capabilities for both the PAS bus and the wireless link. It is not clear if 802.11 wireless will be fast enough, and the PAS bus might require significantly higher data rates than the COMM assembly or the Radio. The current implementation has not demonstrated a need for higher bus rates, but potential unknowns that may require more bandwidth include high and low rate

video streams, streaming audio, and any high speed controls that may need to be addressed.

- The wireless network deployed was configured to automatically adjust link speed. It was operated at a raw bit rate between 5 Mbps and 11 Mbps. Actual wireless network throughput is less than the raw bit rate.

- The implementation details of the Ethernet protocol in hardware and software play a significant role in determining network performance. Special attention needs to be paid in multi-system configurations to ensure that all implementations are performing as expected. It is not enough to just specify that the project is using Ethernet. Variations in buffer sizing, clock speed, and power management, for example, need close attention. Tables 5 and 6 show that the computer used to emulate a PAS assembly was able to successfully receive as much data as the 100MB/s Ethernet link can provide, whereas the Radio testbed, while supporting 100MB/s Ethernet, begins to drop datagrams as traffic increases. It is recommended for future testing that these performance measures be examined, and if possible potential bottlenecks should be eliminated by adjustment of implementation specific parameters.

- The Ethernet link is susceptible to significant data loss during high traffic loading. This is most likely due to the implementation issues described in the previous conclusion. The wireless link did not show this property because the limited data rates of the 802.11 protocol were small enough as to not stress the processor and data buffers on the receiver side. It is important to note, however, that this may not be true for future versions of the radio testbed if the processing capability is reduced.

- The 2-hop test results in Tables 11 - 13 show that the network retains the worst properties of both links; namely, reduced throughput when the traffic originates over the wireless network and high datagram loss when the PAS bus sources much more data than the wireless network can handle.

- A wireless interface should be included to emulate a habitat, spacecraft, or another suit and serve as a data sink for PAS telemetry that is meant to be sent off-suit.

- This round of tests featured a single 802.11 wireless interface. The addition of a second wireless interface will be necessary in order to follow the current AEMU Radio design concept. A lower-rate interface will carry voice and telemetry traffic flows to and from the suit, and the higher-rate 802.11 interface will handle video, data transfers, and act as a contingency interface if the lower rate interface is not operating properly. This would also facilitate the testing of actual suit network traffic, which was not fully tested during these tests. Future tests should include the simultaneous wireless transmission of suit telemetry, multiple voice streams, and video.

5.8. Bus Traffic Analysis

For each major test performed for the PAS Interface Testing, data was capture on the Gigabit Ethernet Bus (GigE) using Wireshark® [8] and saved in a file for analysis and, if so desired, playback. Wireshark is a network protocol analyzer that enables a view into what is happening on a network. For convenience and ease of use, analysis was performed using a third party software package, Cascade Pilot® from Riverbed Technology, Inc.

Only three tests are of interest for this analysis: Network Connectivity Tests, Audio Inbound Tests, and Video Tests. All other tests had similar traffic patterns to at least one of these three tests.

The Network Connectivity Test consisted of each assembly performing Ping tests. The point of this test is to ensure all assemblies can communicate bidirectionally to each other.

Rather than showing Network Connectivity Tests, which would have all traffic between hosts about equal, Figure 11 shows the various IP conversations that occurred during the Unicast Video Test. It shows all IP hosts that were communicating. The size of the host is relative to the amount of data it has transmitted and the size of the connection line is relative to the amount of data transmitted between those hosts. In this example, the HTTP video is being transmitted between the camera, 192.168.1.106 and the display computer, 192.168.1.116.

The next test of interest is the Inbound Audio Stream Mixing Test found in section 2.8. In Figure 12, four G.711 channels of inbound audio and one G.711 channel of outbound audio are shown being transmitted over the bus as well as one LPCM audio stream from the audio processing outbound board to Informatics. No video is running during this test. Note in Figure 12 that the five channels of audio only consume approximately 700 kbps of bandwidth. The

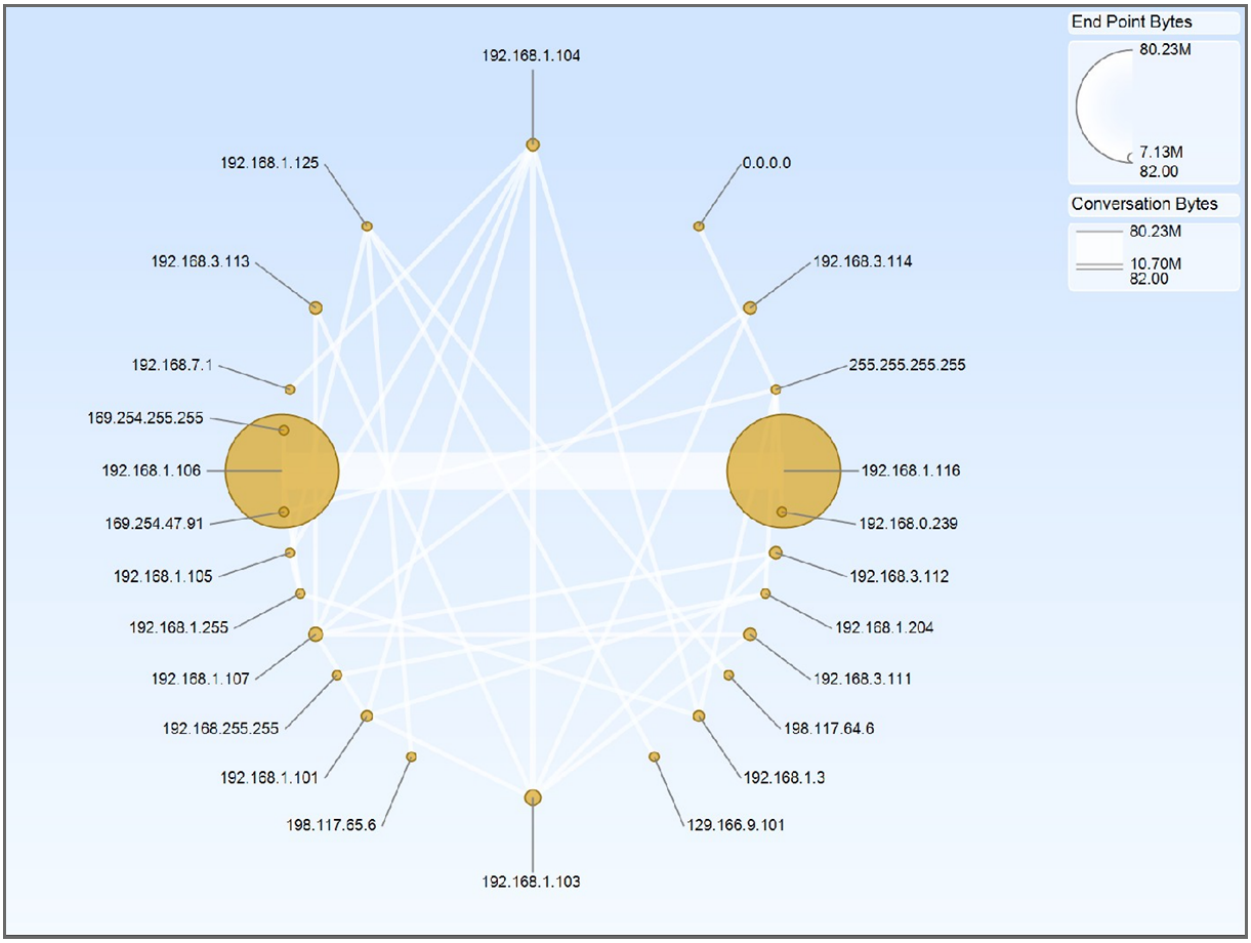


Figure 11: Unicast Video Test

"Unknown" traffic here is the various ZeroMQ Event and Status messaging being transmitted between each of the PAS assemblies. Thus, the overall total bandwidth on the bus, minus video is less than 1 Mbps.

Figure 13 shows all the traffic on the bus over time. Here, two video streams are being sent, one is 8 Mbps while the other is 250 kbps. It is obvious from Figures 13 and Figure 11 that the video data dominates in the PAS bus traffic. This particular video data was sent via multicast whereas the video data from Figure 11 was sent via unicast. The purpose of sending multicast traffic was to ensure it arrived on every switch port and was therefore seen by every assembly. This exercise was performed to determine if the streaming video on the bus would have any adverse effects on the assembly data traffic. It did not. This was because even though the traffic was on each link, except for the video monitoring system, all other assemblies were not subscribed to this multicast traffic and thus ignored it. The audio traffic was running simultaneously during this test and was also unaffected by the video traffic.

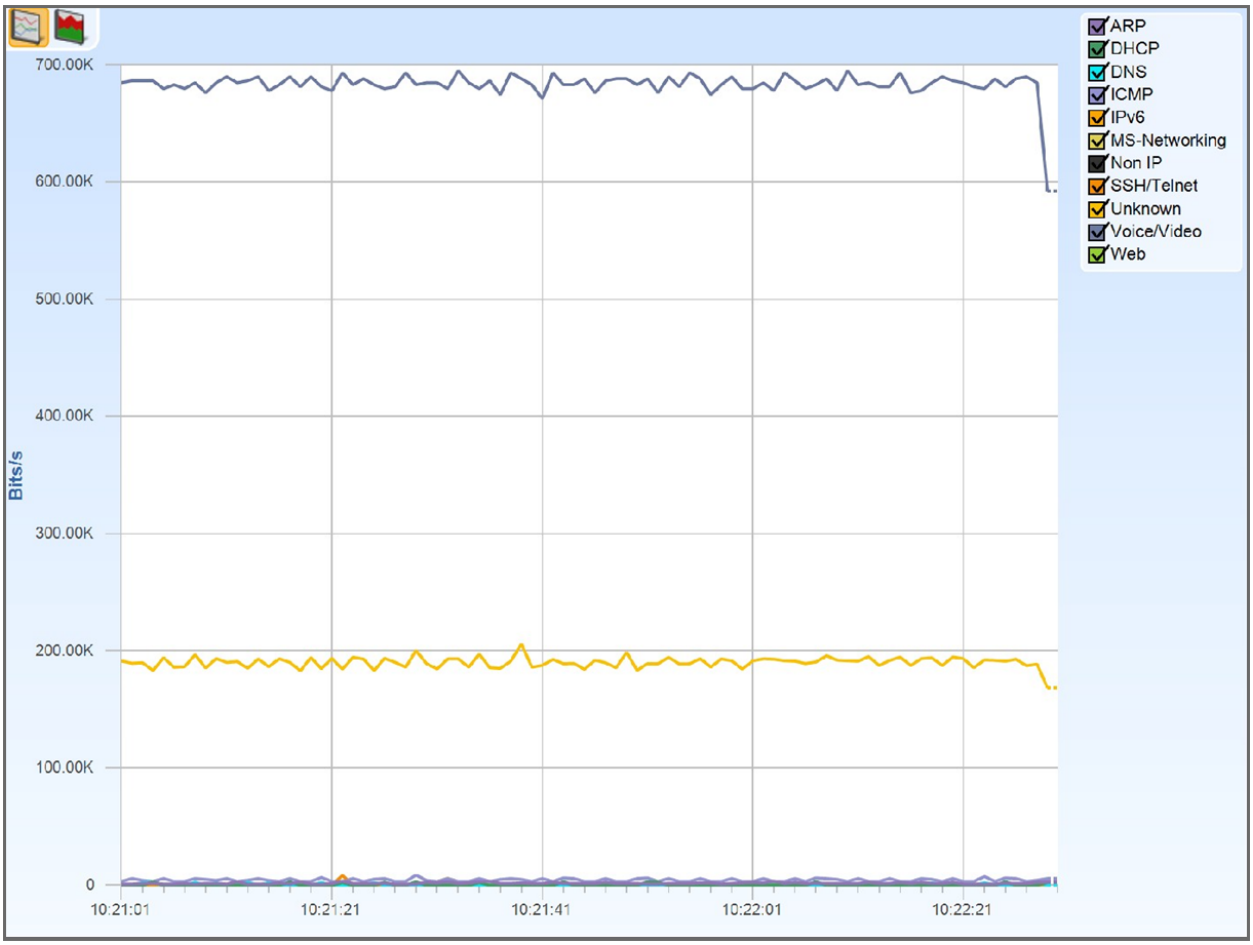


Figure 12: Inbound Audio Stream Mixing Test

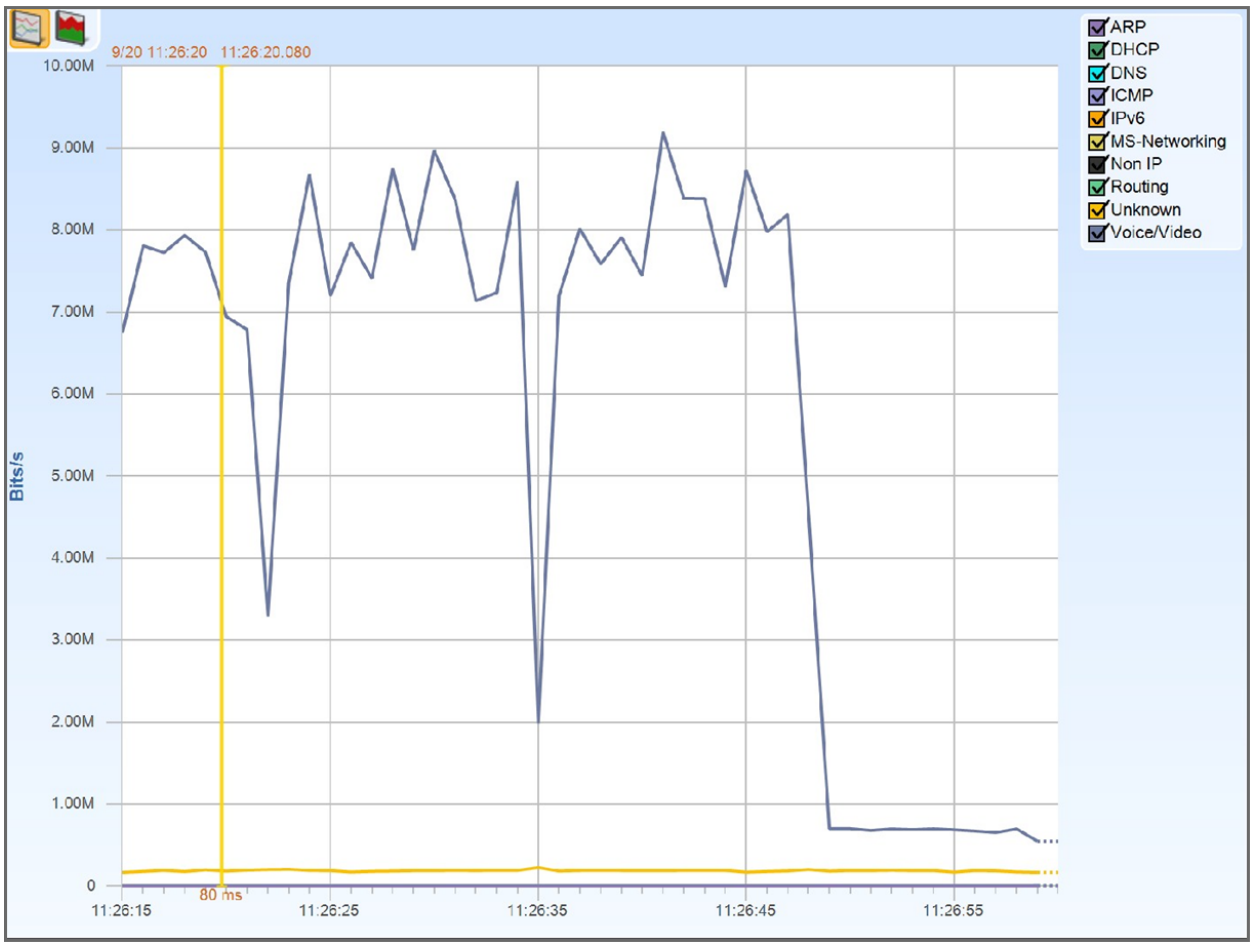


Figure 13: Subsystem traffic and Two Multicast Video streams

6. Summary and Conclusions

These subsystem integration tests provide useful input and insight into the next iteration of the AEMU. Following are some of the major conclusions:

- The types of information flow (i.e. messages and telemetry) that need to be passed between subsystems were identified.
  - Google Protocol Buffers (GPB) worked well for construction messages.
  - ZeroMQ over TCP worked well as our publish/subscribe middleware.
  - Use of GPB and ZeroMQ saved significant programing time as we did not have to reinvent and recode what was already available.
  - GPB and ZeroMQ require significant library support that may not be available for embedded processor or operating systems that are available on radiation hardened processors.
- Use of GStreamer allowed us to implement a Voice-Over-IP (VOIP) operational system with the time and resource allocation provided and then test out much of the audio requirements.
  - GStreamer requires a reasonable learning curve and has many nuances regarding pipeline construction.
  - GStreamer, like many open source software projects, has limited documentation with most support available via user groups and mail lists.
  - GStreamer is quite powerful, but probably but rather code intensive.
- Using a standard bus such as Ethernet, simplifies application layer programming as the drivers and libraries exist that abstract the datalink layer away.
- Using a share Ethernet bus meant each subsystem had to manage only one interface with the exception of COMM which, by design, is multi-homed (i.e. it has interfaces on two distinct subnetworks).
- Using a shared Ethernet bus and Internet Protocols (IPs) allowed use of widely available and proven debugging and analysis tools such as Wireshark.
  - Bus debugging, dissecting and analysis tools will be required for whatever future bus architectures may be deployed. Thus, information must flow in a standardize form in order to develop these tools and have them work across any deployment.
- More often than not, peripheral support is lacking on development boards processors such as those used in many of the PAS 1.0 assemblies, thereby reducing the speed and throughput information flow between subsystems.

There following recommendations are a direct result of this testing:

- Investigate other VOIP methods besides GStreamer such as using a simple G.729 encoder and decoder written in C.
- If GPB and ZeroMQ prove to be too complex for flight implementation, other publish/subscribe methods can be used.
- Network Time Protocol (NTP) proved to be inappropriate for the AEMU. NTP operates under the assumption that higher tiered clock data is available to the lower-tiered clocks for long periods of time [7]. Since the suit electronics are only powered up for several hours at a time, this process is too long and involved for this application. The behavior of NTP was modified to suit the needs of PAS, but most of these modifications simply disabled a lot of the advanced functionality of NTP. Thus, the recommendation is to investigate use of the time-stamp field in the telemetry packet from the COMM subsystem to synchronize the other subsystems as highly precise time-synchronization currently is not a requirement.

- Consider moving video to its own dedicated high-speed bus thereby allowing all other traffic to flow over a much slower, low-power bus.
- If using IP, use multicast (one-to-many) instead of unicast (one-to-one) to dramatically reduce voice traffic over the RF link. In the PAS 1.0 testing, with four end systems, there would be a four times improvement as each end system only has to send one multicast voice packet destined to four systems instead of four separate unicast voice packets.

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List of Acronyms

AEMU Advanced Extravehicular Mobility Unit

AES Advanced Exploration Systems

ALSA Advanced Linux Sound Architecture

ARM Advanced RISC Machines

ARQ Automatic Repeat-reQuest

API Application Program Interface

COMM Communication Subassembly

COTS Commercial-Off-The-Shelf

CRC Cyclical Redundancy Check

CWC Caution, Warning and Control

CWCS Caution, Warning and Control System

DRATS Desert Research and Technology Studies

DTS Decode Time Stamp

EMU Extravehicular Mobility Unit

EVA Extravehicular Activity

FPGA Field Programmable Gate Array

GigE Gigabit Ethernet

GPB Google Protocol Buffers

GPU Graphics Processing Unit

GRC Glenn Research Center

GUI Graphical User Interface

gBS GStreamer Block Size

gJBS GStreamer Jitter Buffer Size

HD High Definition

HDTV High Definition Television

HTTP HyperText Transfer Protocol

IP Internet Protocol

IPv4 Internet Protocol version 4

JPEG Joint Photographic Experts Group

JSC Johnson Space Center

LCD Liquid Crystal Display

LPCM Linear Pulse Code Modulation

MPEG Motion Picture Experts Group

MTV Measurement, Test and Validation

NASA National Aeronautics and Space Administration

NTP Network Time Protocol

PAS Power, Avionics and Software

PC Personal Computer

PHY Physical Transceiver

PLSS Primary Life Support Subsystem

POSIX Portable Operating System Interface for Unix (application programming interface)

PTS Presentation Time Stamp

PTT Push-To-Talk

RF Radio Frequency

RTP Real-time Transport Protocol

SPL Sound Pressure Level

TBR To Be Resolved

TCP Transmission Control Protocol

THD Total Harmonic Distortion

UDP User Datagram Protocol

VOX Voice Operated Switch

VOIP Voice-Over-IP

WiFi Wireless Fidelity (a.k.a. Wireless Internet)

WLAN Wireless Local Area Network

Date of Test: 9/19/2013

Time of Test: 8:06 a.m.

Names of personnel present during testing:

Daniel R. Oldham

Ted Wright

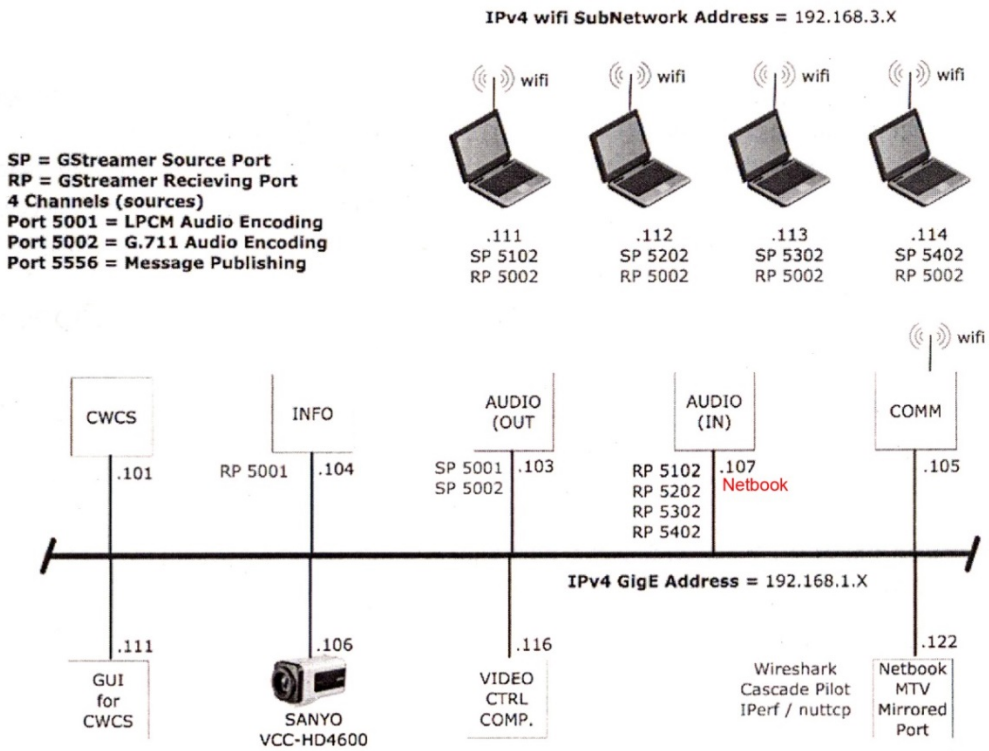
Will Ivancic

Martin Bradish

Joseph M. Klebau

Scott Sands

Casey Bakula



Architecture Lead: Martin Bradish

PAS Subsystem Interface Test Procedure

This test assesses the PAS subsystems ability to communicate between the PAS assemblies using the common bus. For the most part all the test can be run individually, but it is recommended that the "Test setup and Power-on Sequence" and the "Network Connectivity Testing" be run first before any other tests are run to insure proper setup and powering on of the subsystem. Also, it is best to run the "IPerf Stress Testing" last in any test sequence to avoid have to power cycle the assemblies.

Test setup and Power-on Sequence

The main assumptions for this testing are that each of the assemblies have passed there function testing and are ready to perform a subsystem interface test and that the assemblies are powered on and are connected to the PAS bus switch. The following is just guidance to the testers on a preferred connection and power-on sequence.

1. Connect inbound audio assembly to the GigE switch
2. Connect outbound audio assembly to the GigE switch
3. Connect Informatics assembly to the GigE switch
4. Connect CWCS assembly to the GigE switch
5. Connect Monitor, Test and Validation system to the GigE switch
6. Connect Video assembly to the GigE switch
7. Connect Communications assembly to the GigE switch
8. Power on all PAS nodes assemblies (except Video) attached to GigE switch and all external (off-suit) nodes. The order in which systems are powered-on is not important. However, the AUDIO assembly requires a specific startup procedure in part due to how the GStreamer software functions. The required startup sequence for a properly functioning AUDIO assembly using GStreamer as the underlying AUDIO application is given in steps 9-14.
9. Node 3.111 (radio node):
    a. log-on CogNet1
    b. Start terminal window: ./demo2/Linux/demo2node +ro +ti 5102
10. Node 3.112 (radio node):
    a. log-on CogNet2
    b. Start Diskutility and mount 4GB disk.
    c. Start terminal window: /media/4GB/root2/demo2/Linux/demo2node +ro +ti 5202
11. Node 3.113 (radio node):

   a. log-on paspc113
   b. Start terminal window: ./demo2/Linux/demo2node +ro +ti 5302
12. Node 3.114 (radio node):
   a. log-on CogNet4
   b. Start terminal window: ./demo2/Linux/demo2node +ro +ti 5402
13. Node 3.103 (outbound audio):
   a. start Minicom on Node115
   b. turn on board
   c. log-on root
   d. ./demo2/demo2out/demo2out +to
14. Node 3.115 (inbound audio):
   a. log-on paspc102:
   b. Start terminal window: ./demo2/demo2/demo2 +ri
15. Power on Comm
16. Power on Informatics
17. Power on CWCS
18. Power on MTV into Windows OS.

Network Connectivity Testing

1. **Test Description:** This test should be the first test performed. It is a basic connectivity test to ensure all assemblies and other test or monitoring systems are communicating with one another on the bus. As such, this tests the PAS subsystem's ability to properly route messages to check the COMM routing and ARP tables to ensure correctness as this is the gateway for all subsystems to communicate off-suit. (Need to ssh to 192.168.1.105, password?). User: root Password: root

Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence". Check the routing table in COMM to ensure correctness. This is the gateway for all assemblies to communicate off-suit. 1. Give the test script approximately 40sec to run.

Procedure	Pass	Fail
1. Start Wireshark capture. 2. From node 3.111 run test scrip to ping all nodes (./test1.sh>test1.txt) 3. Once the script is finished display results. (cat test1.txt) a. Did all nodes respond correctly?	✓	
4. From node 3.112 run test scrip to ping all nodes (./test1.sh>test1.txt) 5. Once the script is finished display results. (cat test1.txt) a. Did all nodes respond correctly?	✓	
6. From node 3.113 run test scrip to ping all nodes (./test1.sh>test1.txt) 7. Once the script is finished display results. (cat test1.txt) a. Did all nodes respond correctly?	✓	
8. From node 3.114 run test scrip to ping all nodes (./test1.sh>test1.txt) 9. Once the script is finished display results. (cat test1.txt) a. Did all nodes respond correctly?	✓	
10. From node 1.103 run test scrip to ping all nodes (./test1.sh>test1.txt) 11. Once the script is finished display results. (cat test1.txt)	✓	

a. Did all nodes respond correctly?		
12. From node 1.107 run test scrip to ping all nodes (./test1.sh>test1.txt) 13. Once the script is finished display results. (cat test1.txt) a. Did all nodes respond correctly?	✓	
14. Stop Wireshark capture and save the file. a. Log the file name (Test1-09192013)	✓	

All saved under PSP13 folder.

*Note: Perform network checks as needed beyond the required network checks above. This test will be exhaustive in any final version of a PAS test procedure and is recommended to be an automated.

Network Time Protocol Testing

Test Description: This test shows the ability of COMM to distribute time to the PAS subsystems.

Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence". All network time protocol (NTP) clients should be configured to use COMM as the NTP server. This is accomplished by adding the line "server 192.168.1.105 iburst" to the NTP configuration file. All other "server" lines should be commented out.

- COMM will revert back to Jan. 1, 2000 after each reboot
- NTP may not allow for large jump in the system clock. It may be necessary for the NTP client to force a time sync after/during boot before these tests can be run. This can be accomplished by shutting down the NTP service on the client device, running "ntpdate –bv 192.168.1.105", and then restarting the NTP service.

Procedure	Pass	Fail
1. Startup NTP server on COMM (this should start automatically upon bootup) 2. Start a new Wireshark capture on the MTV machine (port mirror).	✓	
3. Change the time and date in upper-level NTP server (be it COMM or the external source) a. Is Date/Time updated?	✓	

1. From node 3.111 run test scrip (./test2.sh) 2. Once the script is finished display results. (cat test1.txt) a. Did node respond correctly?	✓		
3. From node 3.112 run test scrip (./test2.sh) 4. Once the script is finished display results. (cat test1.txt) a. Did node respond correctly?	✓		
5. From node 3.113 run test scrip (./test2.sh) 6. Once the script is finished display results. (cat test1.txt) a. Did node respond correctly?	✓		
7. From node 3.114 run test scrip (./test2.sh) 8. Once the script is finished display results. (cat test1.txt) a. Did node respond correctly?	✓		
9. From node 1.101 manually update the time and date. a. Did node respond correctly?	✓		
10. From node 1.103 run test scrip (./test2.sh) 11. Once the script is finished display results. (cat test1.txt) a. Did node respond correctly?	✓	X	DRO
12. From node 1.104 manually update the time and date. a. Did node respond correctly?	✓		
13. From node 1.107 run test scrip (./test2.sh) 14. Once the script is finished display results. (cat test1.txt) a. Did node respond correctly?	✓		
15. Verify that the NTP clients have synchronized with COMM.	✓		
16. Note in the NTP server Log how long it took assemblies to synchronize and how well they synchronized. a. Take a screen capture of the log files and save. b. Record the file name 8.1 sec to jump to correct time			
17. Stop Wireshark and save the trace files a. Record the file name test2.1-09192013	✓		

Will pinged assembly on PAS bus with round trip of 0.3 ms. And then radio round trip was 1.3 ms.

Assembly Status Testing

Test Description: This test checks that each assembly is updating the Informatics assembly with their health and status telemetry.
Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence". Run script Test 3 on all offsuit nodes & 1.103 & 1.107

Procedure	Pass	Fail
1. From the informatics Comm screen a. Is the Informatics block colored green?	✓	
2. From the informatics Comm screen a. Is the Audio block colored green?	✓	
3. From the informatics Comm screen a. Is the Radio block colored green?		
4. From the informatics Comm screen a. Is the CWCS block colored green?	✓	

Other Telemetry Testing

Test Description: This test checks that the CWCS system is sending consumable telemetry.
Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence".

Consumables

Procedure	Pass	Fail
5. Start Wireshark capture. 6. From the informatics screen is PO2 being decremented?	✓	
7. From the informatics screen is S02 being decremented?	✓	
8. From the informatics screen is Batt being decremented?	✓	
9. From the informatics screen is H2O being decremented?	✓	

Physiological

Procedure	Pass	Fail
1. From the Informatics O2 page is the metabolic rate being updated?	✓	
2. From the Informatics O2 page is the heart rate being updated?	✓	

Basics

Procedure	Pass	Fail
1. Navigate to the Informatics O2 page a. Is the EVA time being updated?	✓	
2. At the CWCS GUI adjust the suit pressure. From 12 to 13 psia a. Is the Suit Pressure being updated?	✓	
3. From the Informatics H2O page, a. Is the Feed Water Pressure at it set pressure?	✓	

b. Is the Water Temperature at it set pressure?		✓
4. From the Informatics H2O page, a. Is the Battery Voltage at it set pressure?		✓
b. Is the Battery Current at it set pressure?		✓

All messages were verified by capture.

CWCS Two Line Display

Procedure	Pass	Fail
1. From the Informatics C&W page view the CWCS two line display mirrored image. a. Is the display being updated?		✓

Wireshark = test 2 telemetry-09192013.pcapng

Streaming Audio to Off-suit Nodes Testing

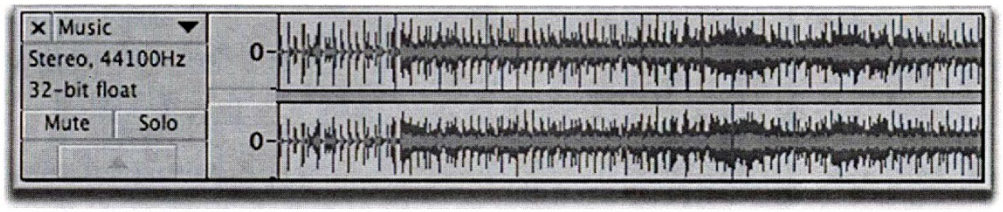
Test Description: This test will demonstrate movement of compressed streaming audio off the suit through the COMM assembly. Use of G.711 for encoding/decoding of voice streams from radio nodes can be verified by code inspection. Wireshark packet captures and audio reception and the output is fed to the network monitoring software package. Note: be sure to boot the MTV notebook into Windows OS.

Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence". Have ready a microphone and speaker set where needed.

Procedure	Pass	Fail
1. From CWCS set the suit pressure to 10 psia. 2. Start a new Wireshark capture on the MTV machine (port mirror). 3. Produce sound from Audio (Out) assembly. a. Was audio heard from each of the off-suit nodes (3.111 – 3.114)?		✓
4. Stop capture and save file a. Record the file name test 3-09192013.pcapng		✓

Streaming Audio Latency Testing

Test Description: This test uses Audacity. Audacity is a free, easy-to-use, multi-track audio editor and recorder for Windows, Mac OS X, GNU/Linux and other operating systems. The test uses two microphones connected to Audacity. One records on track 1 (left) and the other on track 2 (right). The measurement occurs between the two tracks. An example of two tracks is shown in the figure below:



Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence". Connect the two microphones to the Monitor, Test and Validation (MTV) system, start and setup Audacity. Place the two microphones a meter or more apart.

Procedure	Pass	Fail
1. Start the recording of Audio from Audacity 2. Next to one microphone make a click sound loud enough to be heard through both microphones. 3. Stop recording and observe the audio latency from Audacity. (difference should be approximately 2-3ms). a. Record the latency (_____). Is it within limit? 4. Clear recording.		
6. Place one microphone by the Audio (Out) assembly next to its microphone input. 7. Place the second microphone next to the Audio (In) assembly's speakers. 8. Start recording and make a click sound next to the Audio (Out) assembly's microphone.		

In test 5 to setup node sent pressure to 3 psia

9. Stop recording and observe the audio latency. Repeat six times and record the mean and standard deviation. a. Record the mean latency (_____). b. Record the standard deviation (_____).		
5. Save the file to a unique name (ex. Test5.wav) a. Log the file name. b.		
6. Document the Gstreamer buffer settings. a.		
7. Repeat steps 6-10 for each of the off-suit nodes. Calculate the mean and standard deviation of the audio time delays. The Audiacity recording does not need archived. a. Audio (out) to node 3.111 i. Mean: _____ ii. Std Dev: _____ b. Audio (out) to node 3.112 i. Mean: _____ ii. Std Dev: _____ c. Audio (out) to node 3.113 i. Mean: _____ ii. Std Dev: _____ d. Audio (out) to node 3.114 i. Mean: _____ ii. Std Dev: _____		
8. Run test setup script (./test5.sh) on nodes 3.111-3.114, 1.107 & 1.103. 9. Use the following tables to fill in data from the Latency tests.		

Latency Measurement from off-suit node 3.113 to PAS node 1.107 (In bound) Netbook

		Block Size (Bytes)		
Jitter Buffer Size (ms)		160	320	640
	50	287 Started good then stopped	235 Bad break	303 Bad breaking up
	100	262 Stopped	353	311
	200	416	385	425

50/160-Audio started good then brokeup then stopped.

50/320-1.107 needed to be restarted sounds fine but then seems to breakup. ran 1 min to 3 min

Test 5 audio inbound.pcapng

200/160-Audio good for 5 min ended test.

200/320-Audio good for 5 min then stopped.

200/640-Same

Latency Measurement from PAS node 1.103 to off-suit node 3.113 (Out bound)

		Block Size (Bytes)		
Jitter Buffer Size (ms)		160	320	640
	50	425 Breaking up.	309 Good	327
	100	378 Bad, breaking up	345	347
	200	479 Choppy	462	463

100/640-Audio was good for about 5 min then broke up

*Testing Report: 500/4096-for inbound and outbound-Latency (out) = 817 test went over 15 min.
-Latency (in) = 909 ended test.

*New Test: 500/320-Latency (out) = 750
-Latency (in) = 672 } → Failed burn-in test after 1 hr

*New Test: 200/2510

Latency Measurement from PAS node 1.107 to off-suit nodes

200/2560				

Radio → PA2

	Block size (Bytes)			
Jitter Buffer Size (ms)		1280 (160 ms)	2560 (370 ms)	4096 (640 ms)
	20	283 BU	417 BUB	502 Burst and echo
	200	459 Good	480 Good	536 Good
	500	736 Good	/	909

1.103 → 3.113

PAS → Radio

	Block size (Bytes)			
Jitter Buffer Size (ms)		1280 (160 ms)	2560 (370 ms)	4096 (640 ms)
	20	336 Good	450 Good	538 Good
	200	427 Overdrive	571 Good	545 Good
	500	762 Good	/	817

	Jitter Buffer Size (ms)			
	200	100	100	100
	Block size (Bytes)			
	240	240	280	320
PAS → Radio	Not bad	Not bad	Not bad	337 Good
Radio → PAS	Good	Good	Good	347 Good

Radio → PAS

		Block size (Bytes)		
Jitter Buffer Size (ms)		160	320	640
	50	340 BU	274 Good	328 Good
	100	338 BUB	356 Good	/
	200	286 BUB	/	/

Block size, 4094 Bytes—Jitter Buffer size, 500 ms

Radio → PAS

	111	112	113	114
M1	751/917	837	853	~~898~~
M2	730/915	822	854	~~893~~
M3	725/915	817	859	955
M4	729/914	819	853	949
M5	729/911	819	836	952
M6	729/912	792	816	950

PAS → Radio

	111	112	113	114
M1	872	837	845	837
M2	831	837	849	814
M3	830	862	875	848
M4	829	861	873	856
M5	829	864	842	853
M6	808	856	841	851

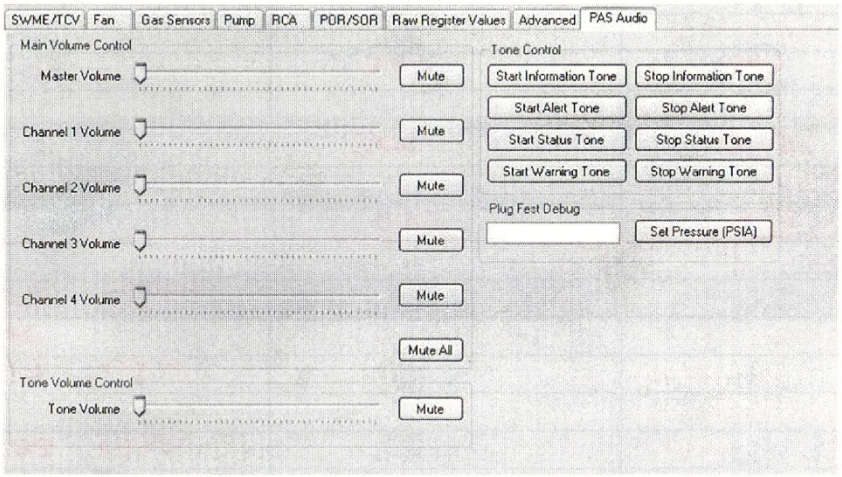
Latency Measurements from the off-suit nodes to the PAS nodes

	Block size	20	50	100	200	500
Radio →PAS	160	268,279	281	262	416	753
	320	247	235	353	385	743
	640	230	303	311	425	748
	1280	283 < 1 min	290	393	459	736
	2560	417 < 1 min	418	470	480	905
	4096	502 < 1 min	488	420	536	909
PAS→ Radio	160	537,570	425	328	479	890
	320	565 ?	309	345	462	731
	640	542 ?	327	347	463	764
	1280	366 < 1 min	568	363	427	762
	2560	450 < 1 min	558	488	571	776
	4096	538 < 1 min	606	653	545	817

Inbound Audio Stream Mixing Test

Test Description: For this test the CWCS-GUI will be used to generate various EVENT_PLAY_AUDIO messages from CWCS to AUDIO. The GUI is show in the figure below. Note, "Master Volume" belongs to AUDIO, not CWCS and thus, will not be exercised here.
Validation is via Audio perception, Wireshark packet capture and telemetry observation on Informatics display.

Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence". Setup and source an audio stream from each of the off-suit nodes (3.111 through 3.114) with a destination to the inbound Audio assembly (node 1.107). Also, ensure the suit pressure is set to 10 psia.



Procedure	Pass	Fail
1. Run test setup script (./test6.sh) on nodes 3.111-3.114 & 1.107. 2. Start Wireshark capture. 3. From the CWCS GUI mute all four inbound channels. 4. Unmute channel 1 a. Did channel 1 unmute?	✓	
5. From the GUI adjust channel 1 volume to max. a. Did channel 1 respond correctly?	✓	

6. From the GUI, mute channel 1.		
7. Repeat steps 3-5 for the remaining audio channels. a. Did channel 2 unmute? b. Did channel 2 respond correctly?	✓	
8. Repeat steps 3-5 for the remaining audio channels. a. Did channel 3 unmute? b. Did channel 3 respond correctly?	✓	
9. Repeat steps 3-5 for the remaining audio channels. a. Did channel 4 unmute? b. Did channel 4 respond correctly?	✓	
10. Save the Wireshark file. a. Log the file name (test6-09202013)	✓	

Audio Tones Testing

Test Description: This test tests the ability to start and stop tone playing and the ability to stop the tone when acknowledged.
Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence". Set volume tone level to midrange and the suit pressure to 10 psia. Also, ensure that all inbound audio streams are muted.

Test Tone Volume and Starting and Stopping of Tones:

Procedure	Pass	Fail
1. Run test setup script (./test7.sh) on nodes 3.111-3.114 & 1.107. *2. Start Wireshark capture. 3. Place a sound pressure meter 1 meter from the Audio (in) speakers. 4. From the CWCS GUI generate a Status tone. 5. From CWCS GUI, adjust the tone volume of 73 dBA (approx. 13 dB above in-suit noise level)		

*Start capture after number 5.

a. Record the speaker volume (73.2)	✓	
4. From CWCS GUI, send an Information tone. a. Did tone sound correctly and last for 30s?	✓	
5. Start Information Tone then depress Stop Information Tone. a. Did Information tone stop?	✓	
6. From CWCS GUI, send a Status tone. a. Did tone sound correctly and last for 5s?	✓	
7. Start Status Tone then depress Stop Status Tone. a. Did Status tone stop?	✓	
8. From CWCS GUI, send an Alert tone. a. Did tone sound correctly and last for 0.5s?	✓	
9. Start Alert Tone then depress Stop Alert Tone. a. Did Alert tone stop? This can be verified by stop message being sent.	✓	
10. From CWCS GUI, send a Warning tone. a. Did tone sound correctly and last for 5min?		✓*
11. Start Warning Tone then depress Stop Warning Tone. a. Did Warning tone stop?	✓	
12. Save the Wireshark file. a. Log the file name test7a-09202013.pcnpng, test7b-09202013.pcnpng test7c-09202013.pcnpng	✓	

*Warning tone seems to starts to slow down 3.5 min and hear varying beeps. At 4 min tone stopped playing Pip tone died.

Test Tone Priority:

Procedure	Pass	Fail
1. Start Wireshark capture. 2. From CWCS GUI, send an Information tone. a. Did Information tone sound?	✓	
3. While Information tone is sounding start a Status tone. a. Did Information tone stop and Status tone sound?	✓	
4. From CWCS GUI, send a Status tone and while Status tone is sounding start an Alert tone.	✓	

a. Did Status tone stop and Alert tone sound?	✓	
5. From CWCS GUI, send an Alert tone and while Alert tone is sounding start a Warning tone. a. Did Alert tone stop and Warning tone sound? This may be verified by monitoring message traffic.	✓	
6. While the Warning tone is sounding try and start an Alert, Status and Information tone. a. Did Warning tone keep sounding?	✓	
7. While the Alert tone is sounding try and start a Status and Information tone. a. Did Alert tone keep sounding?	✓	
8. While the Status tone is sounding try and start an Information tone. 9. Did Status tone keep sounding?	✓	
10. Save the Wireshark file. a. Log the file name (test7priority-09202013.pcapng)		

Test Tone not Adjusted by Master Volume Test:

Procedure	Pass	Fail
1. From CWCS GUI, start a Warning tone. While a Warning tone is sounding adjust the Master volume from the Audio assembly. a. Did the tone volume stay the same volume while the master volume changed?	✓	

test7mastertone-09209013.pcapng

Audio Outbound Pressure Testing

Test Description: This test tests the ability of the PAS subsystem to respond to changes in suit pressure (more specifically to send messages and change audio gains).
Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence". Set volume tone level to midrange and the suit pressure to 10 psia.

Procedure	Pass	Fail
1. Place a Audio (out) microphone in to the microphone calibrator. 2. From the CWCS GUI set the pressure level to 3 psia. 3. Turn on the microphone calibrator. 4. From the Radio or Informatics node, measure the sound pressure level in 3 dB level increments. Log the each measurement below. a. 0 psia = (__96.1__dB) b. 3 psia = (__94.5__dB) 6 psia = 89.6 dB c. 9 psia = (__84.6__dB) d. 12 psia = (__80.4__dB) e. 15 psia = (__76.2__dB) f. 18 psia = (__72.2__dB) g. 21 psia = (__68.4__dB) h. 24 psia = (__62.0__dB) i. 27 psia = (__64.1__dB)	✓	

*Note: Analyze data to demonstrate appropriate pressure compensation function. Signal should be maximum at 0 PSI and approximately 15 dB lower at 15 PSI with a linear progression (in dB) of signal level over full range of suit pressure levels.

*Vol set to max

Transmit Mode Function Testing

Test Description: This test tests the ability of the PAS subsystem to respond to transmit mode changes.
Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence".

Procedure	Pass	Fail
1. Run test setup script (./test9.sh) on nodes 3.111-3.114 & 1.107. 2. Start Wireshark capture. 3. Set mode to Open-Mic. 4. Set up audio from Audio (out) to Off-suit node 3.111 5. Press the PPT button. a. Is audio still sounding at 3.111? PTT should not have an effect in this mode.	✓	
6. Set mode to PTT a. Did the audio stop playing at node 3.111?	✓	
7. While in the PTT mode, press the PTT button. a. Is audio heard at node 3.111?	✓	
8. Set mode to VOX a. Did the audio stop playing at node 3.111?		*✓
9. Speak into the AUDIO (Out) microphone. a. Was your speech heard at node 3.111?	✓	
~~10. Generate a TBD spl tone at the AUDIO (Out) microphone. Increase the tone level until audio starts sounding at the 3.111 node.~~ ~~a. Record the tone level (_____dBA)~~ test9-09202013.pcapng	MRB	

*Not implemented

Video Stress Testing

Test Description: This test is being performed to add a video stream onto the subsystem bus tests the ability of the PAS subsystem to respond to transmit mode changes.
Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence".

Do we start audio streams with this test?

Procedure	Pass	Fail
1. Start Wireshark capture.	✓	
2. If not already running, use Work Station 192.168.1.106 to open a TCP/IP connection with the camera by entering the URL http://192.168.1.116 using the Internet Explorer Web Browser. a. If prompted for login, use "admin" for username and password. 3. Select "Client Settings" and verify the following: a. Live Stream: STREAM 1 b. H.264 Streaming Protocol: HTTP (INTERNET) 4. Navigate Sanyo camera "Menu" system to modify following parameters as needed for the Video Stream: a. Recording: OFF b. Codec: H.264 c. Resolution: 640 x 360 d. Digital PTZ: OFF e. GOP: 30 f. Frame Rate: 30 fps g. Priority: BIT RATE h. Bit Rate: 560 kbps	✓	

*Note: The user must hit the "Set" button under the video stream menu to initiate parameter changes. The camera will require 30 to 60 sec. for reconfiguration and reconnection.	✓		
5. Move camera field of view randomly to simulate image content were mounted on a suit. Observe displayed video for image artifacts. a. Is the video void of motion and image artifacts working correctly ← No artifact noted i. From Wireshark record bit rate (_____) *(Note: Maximum data rates will be require the highest Resolution (1920 x 1080) , and maximum frame rate (30fps)	✓		
6. Increment the bit rate to 1 Mbps. a. Is the video void of motion and image artifacts working correctly ← No artifact noted b. From Wireshark record bit rate (_____)	✓		←Change Res 1920 x 1080
7. Increment the bit rate to 4 Mbps a. Is the video void of motion and image artifacts working correctly ← No artifact noted b. From Wireshark record bit rate (_____)	✓		←Res 1920 x 1080
8. Increment the bit rate to 5 Mbps. a. Is the video void of motion and image artifacts working correctly ← No artifact noted b. From Wireshark record bit rate (_____)	✓		←Res 1920 x 1080
9. Increment the bit rate to 8 Mbps a. Is the video void of motion and image artifacts working correctly ← No artifact noted From Wireshark record bit rate (_____)	✓		
~~10. Increment the bit rate to 20 Mbps.~~ ~~a. Is the video void of motion and image artifacts working correctly~~ ~~11. From Wireshark record bit rate (_____)~~			Could not go to 20 Mbps
12. If test at 20 mbps passes, use camera Menu system to create a 2nd video	✓		

stream with the following parameters: a. Recording: OFF b. Codec: H.264 c. Resolution: 320 x180 d. Digital PTZ: OFF e. GOP: 30 f. Frame Rate: 5 ips g. Priority: BIT RATE h. Bit Rate: 250 Kbps 13. Connect another node of the PAS breadboard to display the second video stream using steps similar to Steps 2 and 3 above, only specify the Live Steam as STREAM 2. 14. Observe both streams for image artifacts while randomly moving the camera FOV.	✓		Could not figure out how to view two videos at one time.
15. If this test passes go to the work station and change the setting of the stream from Unicast to a Multicast stream. a. Log how the subsystem responded.	✓		
*16. Save the Wireshark file. a. Log the file name (_____)	✓		

*Took different Wireshark capturer for each setting.test-video-ucast-566.pcapng
1000
4000
8000
8000-250
mcast-8000-250
Set Client from HTTP to multicast (puts video on all switch points) Audio still working

IPerf Stress Testing

Test Description: This test is being performed to stress the wireless link capacity.
Test Setup: Ensure that all subsystem assemblies and associated equipment is powered and connected to the switch as outlined at the beginning of this test procedure under "Test setup and power-on Sequence".

f = one direction
d = two directions

Procedure	Pass	Fail	
1. Connect a Linux-based PC to the PAS bus as 192.168.1.25 and configure it as an IPerf UDP server. a. Is PC visible on the network?	✓		
17. Using an available notebook connected to the network, perform IPerf test. 1. Enter the command "iperf –u –r –c 192.168.1.25 –t 60 –b 10M" a. Record client response (_____) b. Record server response (9.94 Mbps, 1.27 ms) No packet loss	✓		
18. Enter the command "iperf –u –r –c 192.168.1.25 –t 60 –b 15M" a. Record client response (_____) b. Record server response (14.8 Mbps, 0.414 ms, 0.015%)	✓		Bit rate, jitter, packet loss (%)
19. Enter the command "iperf –u –r –c 192.168.1.25 –t 60 –b 20M" a. Record client response (_____) b. Record server response (19.7 Mbps, 006 ms) No packet loss	✓		
20. Enter the command "iperf –u –r –c 192.168.1.25 –t 60 –b 30M" a. Record client response (_____) b. Record server response (24.0 Mbps, 1.232 ms) No packet loss	✓		
6. Enter the command "iperf –u –d –c 192.168.1.25 –t 60 –b 10M" c. Record client response (4.03 Mbps, 2.291 ms, 58%) d. Record server response (4.19 Mbps, 12.212 ms, 5.4%)	✓		
7. Enter the command "iperf –u –d –c 192.168.1.25 –t 60 –b 15M" a. Record client response (_____) b. Record server response (_____)		✓	Throughput limit was reached.

8. Enter the command "iperf –u –d –c 192.168.1.25 –t 60 –b 20M" a. Record client response (_____) b. Record server response (_____)		✓
9. Enter the command "iperf –u –d –c 192.168.1.25 –t 60 –b 30M" a. Record client response (_____) b. Record server response (_____)		✓

*Note: After some of the higher bandwidth tests, the network interfaces may become locked up. If this occurs, then steps 1-3 will have to be repeated before any more tests can be run.