AUDIOVISUAL
PRESERVATION
SOLUTIONS

350 7th Avenue
Suite 1603
New York, New York
10001
212-564-2140

Digital Audio Interstitial Errors:

Raising Awareness and Developing New Methodologies for Detection

by
Chris Lacinak

AudioVisual Preservation Solutions
http://www.avpreserve.com
info@avpreserve.com

January 6, 2010

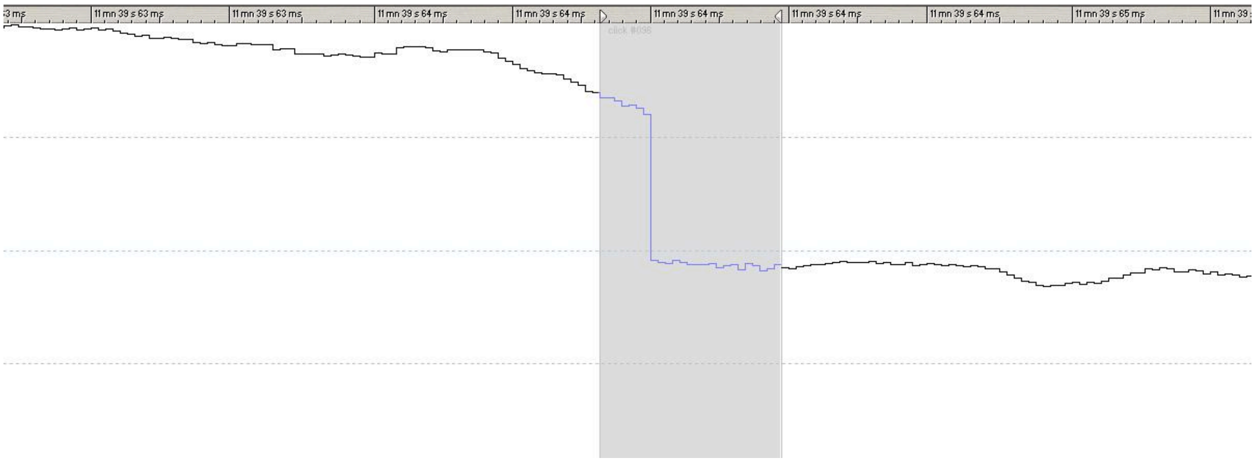

Image 1

Introduction

This is an image of an audio waveform. It was digitized using a professional analog-to-digital (A/D) converter and custom, streamlined and fully tested Digital Audio Workstation (DAW)[1] from Sonica Labs[2]. What you see in the waveform highlighted grey with a straight vertical line is the result of an error during the digitization process. In this case the A/D converter was operating properly, there were no extra applications on the DAW, it was not connected to the network or Internet, and there was no virus protection or similar software installed. This begs the question, "What could have caused this to happen?"

Put simply, while the digital stream was being written to disk something happened to the system resource allocation. In the middle of writing the samples of the digital stream to disk the system got "distracted", skipped some number of samples, "remembered" it was supposed to be recording and then "came back" to continue its original task. In other words, for a short bit of time the samples were not recorded, resulting in a few "missing" milliseconds from the stream of sound. If those samples had been recorded, the line would move in an orderly way down a slope instead of making the abrupt drop that you see in image 1 above. We have defined the type of error seen in image 1 as a *Digital Interstitial Error.*

---

[1] Benchmark ADC Analog-to-Digital converter via AES to an RME Hammerfall 24/192 card seated in a Sonica Labs Turnkey fully tested Digital Audio Workstation.

[2] Sonica Labs builds computers from base level components such as chipsets, busses, and processors which have been tested for compatibility with software and hardware devices used in digital audio studios. They install all the software from scratch so there are only the essential applications and utilities required and no unnecessary software that may cause conflicts or audio issues. Prior to sending their systems out Sonica Labs executes a series of performance tests using the audio application the system will be used with in the studio to ensure that everything is operating properly. In addition to this, we have not installed ANY additional software and have not connected the DAW to the network.

Images 2 and 3 below are waveforms produced by two recording passes made from the same section of audiotape. The audiotape was played two times and recorded two times. If all went well in both recording passes the waveforms should look the same. However you will notice that they do not. This is because image 2 represents the scenario described in the paragraph above resulting in missing samples (In fact this is the same audio as shown in image 1, only zoomed out further to show more of the captured signal). Image 3 represents a second recording pass of the same audio that was error free. These images start and end with the exact same sample of the recording. The first vertical pink line in each image represents where the samples started getting lost in image 2. The second vertical pink line is used as a common point of reference in both images that is after the section missing samples. The difference in length of the waveforms, represented by the space between the two vertical pink lines, is equal to the missing audio in the first image. In this case it happens to be a difference of 513 samples (706 samples in between the two pink lines in image 3 minus 193 samples in between the two pink lines in image 2), or 5.34 milliseconds at a sample rate of 96kHz. This number can vary depending on the severity of the problem.

Image 2

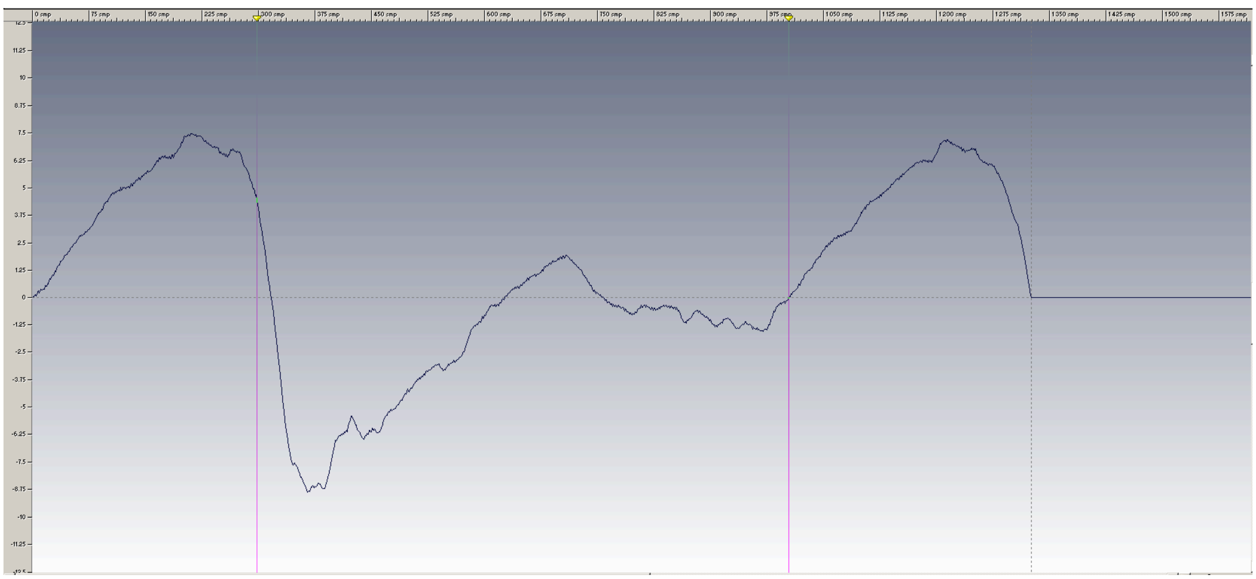

Image 3

With this particular DAW, this digitization error happened only once after roughly one-hundred hours of recording.

Maybe it's your system?

I have been recording using DAWs since 1996 in world-class multi-million dollar facilities and basement studios put together for very little money. These experiences introduced me to numerous types of operating systems, software applications, computers, configurations—you name it. The same issue identified above arose in every studio I ever worked in for any significant amount of time.

After years of recording and editing voiceovers and sample libraries, and mastering and performing audiovisual preservation work, I have become particularly aware of what is going on at the sample level where these issues "hide". I believe many of these errors frequently go unnoticed.

During the years of experience that made the widespread occurrence of these errors apparent to me I also noticed that this is a secret that no one likes to talk about. When I bring this issue up in the close company of other engineers, I am often met with similar war stories. It seems no one wants to admit it because they are afraid it is something wrong with only their system. The nature of the problem and the variety of components involved in the DAW make the culprit a mystery most of the time.

This experience led me to proceed with both eyes open on the lookout for interstitial errors regardless of how good I feel about a system or how much it costs. No system is entirely immune from these errors. I suggest always incorporating a quality control process that identifies errors such as these.

How do you do that?

AVPS uses Wavelab's Error Detection Analysis tool. The problem with this tool is the same as with any automated signal analysis tools: In order to be sure you capture all of the errors you have to set the detection threshold to be very sensitive, often resulting in major over-reporting.

In any given audio file there will be hundreds of points flagged as possible errors. Each one of those must be reviewed manually and hopefully 100% of them are false. In a month of transfers this was the case until we found this one error. The time involved in analyzing and reviewing each file is significant. It adds approximately 5 - 10 minutes per file on average (a cassette would yield 1 file per side). Needless to say, this adds a considerable amount of time and cost to the preservation process.

Isn't there a better way?

Not yet, but one is on the way! Current analysis systems have two primary drawbacks. One is that they use algorithms to guess what might be an issue according to given criteria. For instance, a click detection algorithm may identify a possible click anywhere in the audio signal where there is an amplitude difference greater than "x" over some fixed number of samples. The more sensitive you make the settings the more points are identified as suspects. The less sensitive, the greater the chance of missing a real error, and even then you are still likely to get over-reporting. The implication of this major over and under-reporting is that excessive human input is required to yield meaningful results.

The second issue is that analysis systems which monitor the signal in real-time to identify “digital errors” exactly such as those discussed in this document seem to analyze the signal prior to being written to disk. What is earlier described as the system getting “distracted” is something that usually happens as the data is being written to disk. Monitoring the signal before that point will not catch these issues. The error is occurring in that interstitial moment between analog playback and finalization of writing the digital file to disk. The signal must be analyzed as it is played back or retrieved from disk, not before.

It's only a few samples. Does it really matter?
From the perspective of archiving and preservation, it absolutely matters. If authenticity and integrity are important to you, then it matters.

To draw an analogy it would be the same as cutting a little piece of tape out of the middle of an audiotape or videotape, or cutting out some number of frames from a film. These physical equivalents are avoided at all costs in the preservation of content stored on physical media, why shouldn’t the same principals apply to digital objects?

Many may be willing to accept the odds associated with the example stated above (only one glitch out of a hundred hours of digitized content) in order to save time and cost. But if left unmonitored, these issues can be rampant and may be indicators of larger system issues. This is particularly true of non-streamlined, non-turnkey solutions. I have seen these errors littered throughout even short sections of audio.

In further testing at AVPS we were able to generate more errors by transferring data to a network drive as a background process while recording. Keep in mind that this is only a one-channel recording, which barely taxes system resources at all, so it's not as if we were pushing the limits of our fully tested, custom DAW. The images below show two different sections (images 4 through 7 and images 8 through 11) of a recording where errors occurred during this experiment. It is notable that range of time represented by each set is less than 20 milliseconds, and that multiple errors occur within this time range.

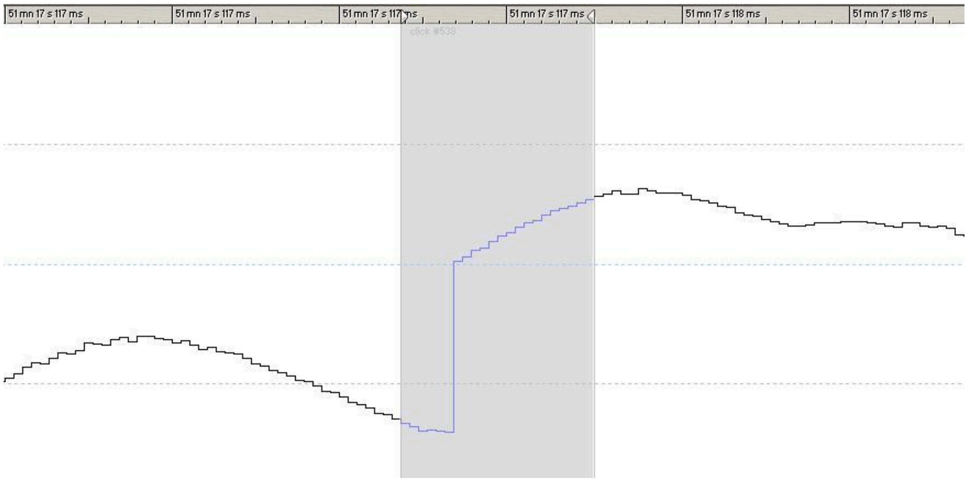

image 4

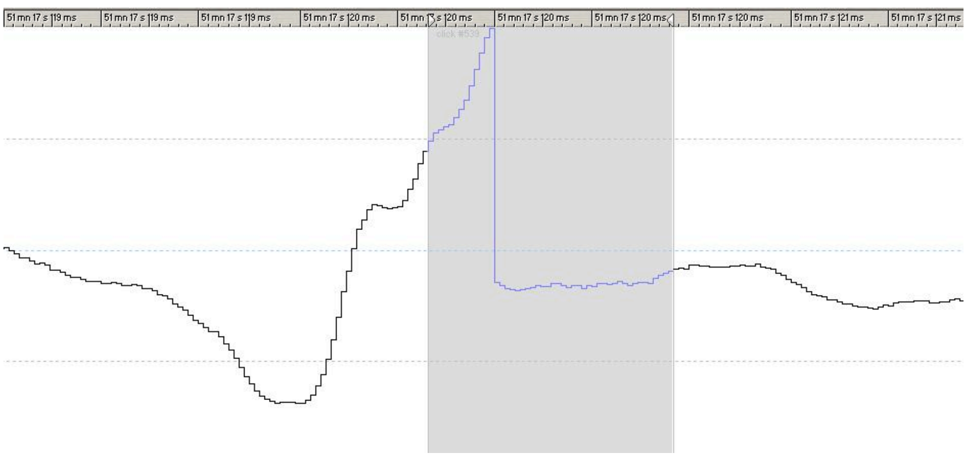

image 5

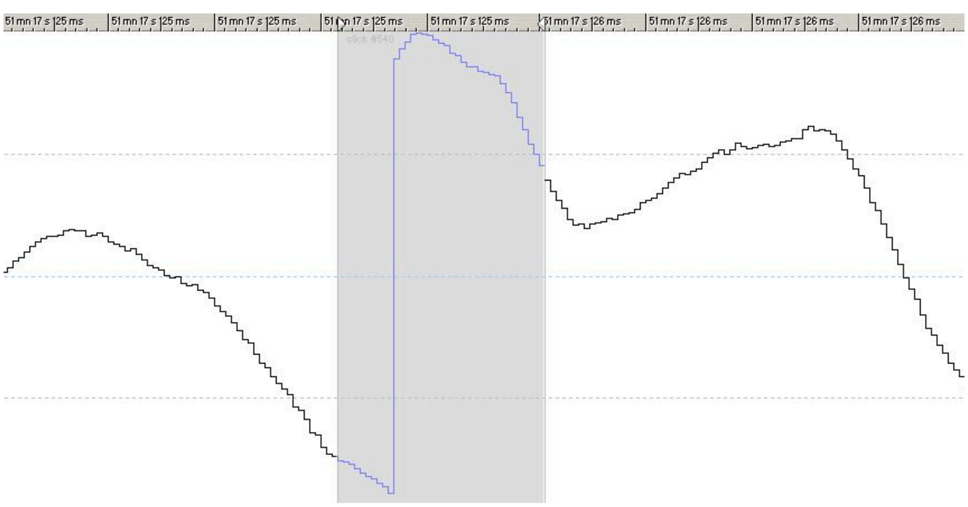

image 6

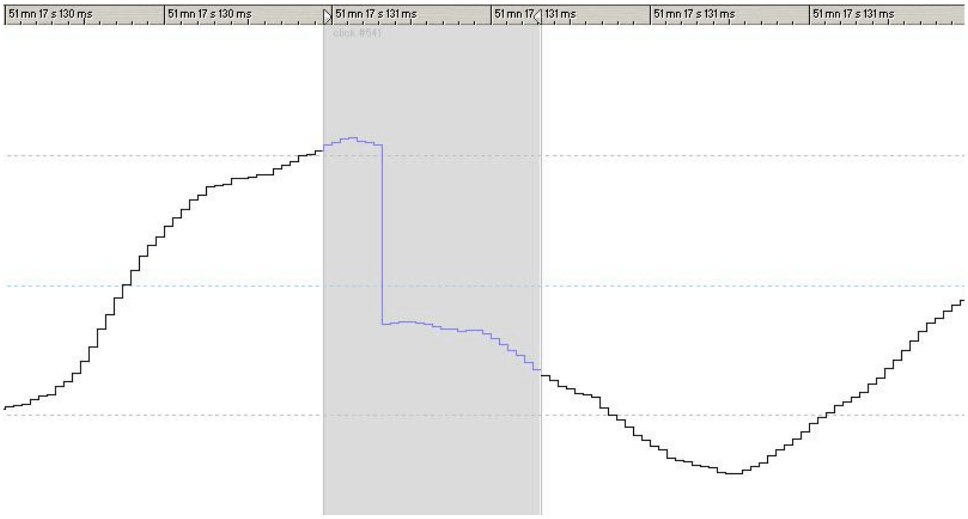

Image 7

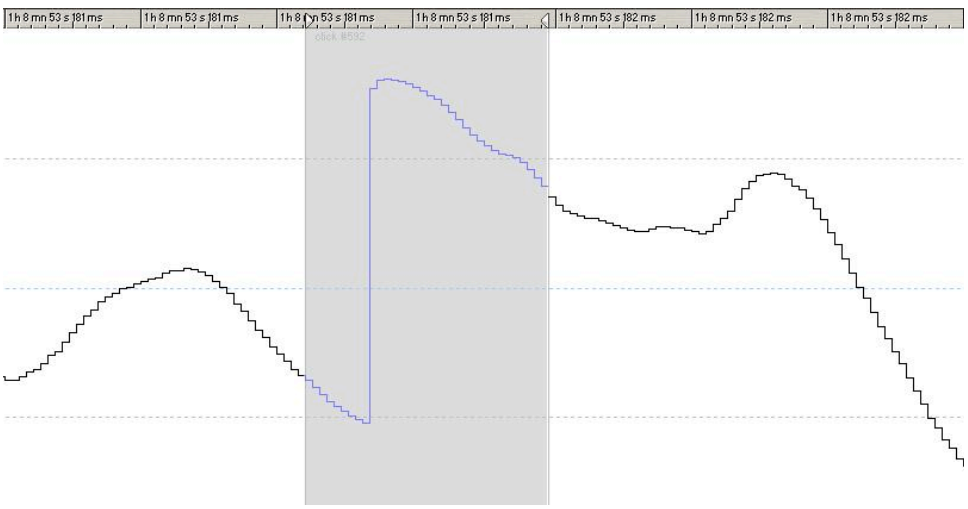

image 8

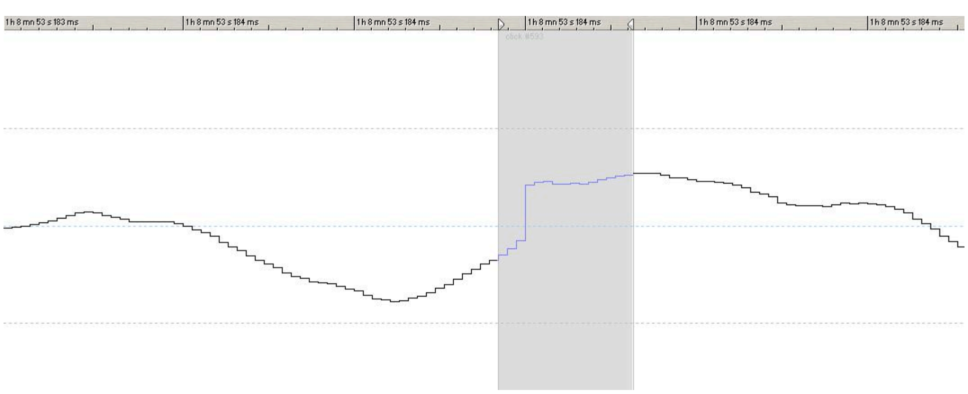

image 9

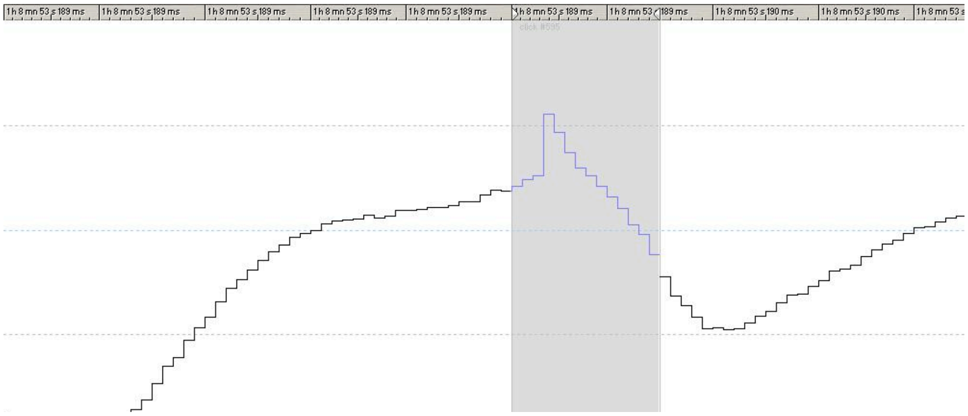

image 10

Image 11

The bottom line is that you can't be complacent regardless of the amount of money you spend. Also, the less "turnkey" and “streamlined” your DAW is, the more diligent you need to be to mitigate the increased risk of errors. There are some simple rules you can follow which will decrease the risk, but there are never any promises.

A solution is on the way

Believe it or not as common as this problem is, there is no standard test method to evaluate this type of digital error. Existing standards for tests and measurements of digital audio address an entirely different and more traditional set of audio performance criteria. New standards and tools are required to properly test and measure *digital audio systems*.

The Federal Agencies Digitization Guidelines Initiative has commissioned AVPS in a research and development project toward these ends. This group of Federal Agencies includes The Library of Congress, The National Archives and Records Administration, The Smithsonian Institution, and others. We are exploring the development of multiple tests, one of which addresses exactly the problem discussed in this document. Simultaneously, through our involvement in the Audio Engineering Society (AES) standards working groups, we are driving the development of new test and measurement standards to address these issues.

Within this body of work AVPS is proposing a different method of detection for digital interstitial errors than is currently used by signal analysis tools. We approach the issue through comparative analysis and believe that the intermittent and unpredictable nature of digital interstitial errors calls for implementing ongoing testing. In addition, our proposed testing criteria and methodology will not produce over and under-reporting, thereby saving a tremendous amount of time and money.

Traditional system performance testing assumes that testing will be undertaken routinely—once per day/week/month. Using this approach, the system is under test for a couple of minutes at most. For monitoring many types of issues this snapshot is sufficient. For digital interstitial errors, the chances of identifying them during a periodic routine test such as this are next to none.

With this understanding, it becomes evident that the best way to catch these intermittent issues is to perform ongoing analysis of the system using the comparative analysis methods proposed by AVPS. In other words, to constantly test the system during digitization of all content.

The basic idea is fairly simple. The approach is to record as usual to the DAW while inserting a digital distribution amplifier (DA) at the digital output of the A/D converter (some A/D converters already provide multiple digital outputs, alleviating any need for a DA). This would yield two identical digital outputs from the A/D converter. One digital output of the DA would go to the DAW digital input. The other digital output would go to a standalone file-based recording device. Comparative analysis of the two resulting files would determine whether there are any significant differences, which in turn would reveal points in the file where digital interstitial errors have occurred.

When will it be Available?

While the work of the Federal Agencies and AES are related, they are not directly linked to each other. They will proceed independently at their own pace. Although these ideas are fairly simple in concept, the standards process is necessarily arduous. Also, the Federal Agencies are proceeding with this project as one of many. The timeline is dependent on many factors that disallow any certain predictions about completion dates. In the meantime there is a role for you to play. We would like to hear from you. The first question that any potential developer wants to know the answer to is "How big is the market?" If we can supplement educated estimates with actual numbers it would help us give potential developers of these tools the information they need to evaluate the business case and come back with numbers on prospective pricing. With this in mind, it would be meaningful to hear from those of you who would be interested in using and purchasing such a tool. Please Email us at info@avpreserve.com so that we can represent your voice in our endeavor to add better test and measurement utilities to the collective tool set.