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Final Project: Generational Loss with Digital Audio Tape

Digital Audio Tape seems to have become a recurring topic of my research in Digital Audio Processes. Last semester, I focused on the sonic benefits that could be afforded by DAT through the introduction of a 24-bit recorder by Tascam. This semester, I decided to investigate the possible sonic degradations that could arise from using DAT as a storage and mixdown medium. I suppose the reason I have focused most of my reasearch in DAP class on DAT and DAT machines is because it is the only digital audio recording device which I currently use in my own studio. Besides the occasional digital delay or digital reverb unit, the recordings I make in my home studio completely exist in the analog domain. From microphone to mixer, to 1/2" reel-to-reel machine, the signal path I use never becomes digitized. Until, of course, the entire mix is fed to the DAT recorder. Originally, I owned a 1/4" 2-track reel-to-reel recorder to use as a mixdown machine. Unfortunately, I was dissatisfied with the sound quality (in retrospect, probably just due to old and worn heads), so I bought a DAT recorder in the hopes of boosting the sound quality of the final mix. The quality difference was far superior with the DAT recorder, but I had always had some trepidation about the mystery and possible problems with transferring sound from one domain (analog) to another (digital). While DAT recorders seem to be the industry standard of mixdown for project studios such as mine, analog machines still seem to dominate in larger, more professional environments. The question arose in my mind, therefore, as to what the disadvantages were to the DAT format. Presumably, if the signal path is going to end up on compact disc anyway (with the same bit level and sampling frequency as DAT), how could a DAT machine be any sort of problem?

The main answer, of course, to professionals' preference of analog tape over DAT is that the conversion process can take place in the mastering house (where ultra-high quality converters are avaiable) instead of the recording studio (where the converters are usually just those internal to the DAT recorder itself). The sonic benefit of using higher-quality converters is obvious but of course subtle. It is a benefit that can be appreciated and afforded by large companies, but really does not seem appropriate for smaller studios such as my own where the sonic quality of the signal path is already limited to an extent by other "semi-pro" equipment. The conversion process, therefore, was not a source of anxiety for me. A bigger factor loomed over my head concerning DAT quality.

Probably everybody who has worked just a little bit (no pun intended) with DAT recorders has at least at one point or another heard a burst error. It is a sound that one does not forget too quickly. The musically is happily playing along and then an odd-harmonic rich, loud burst of noise sounds over the playback. It is a sound that is immediately noticeable and completely unagreeable. With analog tape, sound degradation is very subtle. Perhaps a small portion of the high end response will disappear; perhaps faint ghost images (due to print-through) will appear before the original signal; at worst, a small warm pop will be noticeable, probably due to a speck of dirt in the tape path. For some reason, all of these manifestations of analog tape degradation are only slightly disturbing to the ear. Usually, the average listener does not even pick up on such effects. Conversely with DAT, however, a speck of dirt or an inordinate amount of print-through will cause the tape to be completely misread by the machine; 1's and 0's become a chaotic jumble. The machine itself interprets such random digital information as noise--noise which sounds at a decibel level higher than the recorded progam itself. This noise can never be ignored by the average listener.

The topic in which I became interested was how I could avoid creating burst errors, or even slight sonic imperfections, to DATs. Since the DAT recorder is my main mixdown medium, I have many master recordings in my studio on DAT. These recordings are truthfully "one-of-a-kind", and any sort of degradation is completely unacceptable. So how do DAT tapes become degraded? How do burst errors come into existence? Are there ways to avoid such sonic casualties? I knew that time plays an unfortunate role with magnetic tape (its the same physical tape used in DAT machines as analog cassette decks, just recorded onto in a different). Obviously storing tapes in a cool and dry environment is the safest option to avoid the effects of time. The only other detriment that could plague DAT is sonic degradation due to playback tape loss and generational copy loss.

A lot of voodoo exists in the recording field about what procedures can harm a DAT's recording quality. At Greene Street Recording where I used to work, one of the engineers commented on how he felt that making DAT-to-DAT copies reduced the stereo imaging of the original signal. I cannot imagine how this process could occur since the digital signals in no way could get mixed like analog signals can affect each other. The whole nature of digital recording prevents crosstalk from manifesting itself in the signal (digital is either wrong or right). Another engineer commented that some transfers I had made from DAT to CD sounded like they had less reverb on them. Of course, again, this statement shows the preposterous nature of misinformation that exists among professional studio engineers. But although it is certainly impossible to lose reverb when making a digital copy, could some other sort of artifact be contributing to a sense of lost reverb? Perhaps some high frequency distortion was really to blame for this engineers dissatisfaction with the copies. Only experiments could prove anything.

A mere two years ago, Stephen St. Croix (monthly columnist for Mix Magazine) contributed an article that addressed most people's unworried use of digital audio tape. He reminded us that when digital audio recorders were first released onto the market, they included error indicator lights to report the amount of errors being registered on the tape. A common scenario were three lights for a DAT machine, 1) a green light indicating data correction was occuring but without any pitfalls, 2) a yellow light indicating data correction powers had been surpassed by the error but that interpolation was being used to recreate the samples, and 3) a red light that signaled the breach of both interpolation and error correction powers of the player, thus resulting in a muted signal. The scary thing about these error indicator lights was how often they flashed yellow or red. Apparently, the more tapes were played, the increase in yellow and red lights also became disturbing.

As a result, manufacturers ceased including these blinking light features onto their DAT machines in the effort to conceal from the engineer the imperfect digital audio data that was being read and recorded. Stephen St. Croix is not, however, arguing for people not to use DAT (since they are rather affordable for the sound quality they offer), but just to realize the potential dangers that are involved in storing and replaying information stored digitally on analog tape. He advises making multiple master copies (requiring multiple DAT machines) and also advises treating the masters with care. It really should not be such a surprise that digital audio tape is not the perfect medium for data storage. The head gap, tape track width, and tape speed are all extremely small compared to analog recorders; a speck of dust becomes gigantic in comparison to the digital information and thus poses are larger threat than dust on an analog tape. According to Stephen St. Croix, repeating fifteen times a tape played ten times and then copied should induce a very noticeable loss.

Stephen St. Croix, of course, is not a misinformed studio engineer with minimal training. He is a respected columnist in a national publication. If he is arguing that tape and generational loss with DATs is completely noticeable, I felt is was a subject worth addressing. Not only was it a subject worth addressing, but it was a topic that begged some controlled testing, experimenting, and analysis. So I decided to design a series of tests that would investigate whether or not tape and generational loss was truly something about which the average recording engineer should be concerned. Moreover, if such degradation were truly a problem with DATs, were there any methods by which engineers could avoid the effects? For example, is the digital transfer protocol of AES/EBU superior to S/PDIF in all instances and for all types of sound sources? How much better are these digital tranfers than their analog counterparts? Do instances exist where an analog transfer can recover the effects of tape loss without endowing any of the supposed artifiacts of digital generational loss?

To answer these questions, I needed a variety of sound sources to analyze and a variety of different available transfer methods. I settled on four sound examples to represent the widest possible range of music and controlled sound. Pink noise, with a logarithmically balanced spectrum, seemed like an obvious first choice. If obvious degradations in certain frequency ranges were going to be noticeable, the pink noise should elucidate such imperfections through a Fourier Transform. My second sound source was simple 1 kHz sine tone. I felt that with such a pure signal, any unwanted added harmonics or degradations would be also easily noticeable through Fourier analysis. Not trusting the experiment completely to mathematical methods, I resolved to add two examples which would allow my ears to be the judge of sonic quality. As an example of rock music, I chose fifteen seconds of a rock selection, the opening to Sonic Youth's "Chapel Hill" on their album Dirty. To represent the pure side of classical music, I chose the opening to Chopin's Prelude Opus 28 No. 28 in F-sharp minor performed by Irina Zaritzkaya on the Naxos label. For the sine tone and the pink noise, I recorded ten seconds of sound. In retrospect, these examples should have probably been even shorter since the maximum window length available for an FFT is only 100 milliseconds. For the musical examples, I gave a good fifteen seconds of recording time to capture one complete phrase and allow the ear to really get used to the overall sound quality of each example. Again, in retrospect, I think these examples may have been able to have been cut down to only ten seconds. Since the experiments involved multiple playings and copyings, the length of the total program obviously became a great factor in the time necessary to complete the experiments. If I had been able to cut the program time in half (from 65 seconds to 33 seconds), I could have probably withstood making an even higher number of generations and thus forecasted further down the road of copy loss.

Having thus compiled a master recording of four sound sources, I applied the DAT tape to a variety of rigors to replicate average conditions undergone through normal copying and playback. To make a copy of a DAT, I of course needed two DAT tapes. The master recordings were first transferred to a TDK 16 minute DAT (High Density Recording). To reproduce the errors and generational loss induced with copying, I transferred back and forth between the tapes until I got a 5th generation recording. To also attempt to induce as much tape loss as possible, I also played each transfer once again after it was recorded. By the 5th generation recording, therefore, each musical example had gone over the tape heads 10 times and through the transfer protocol 5 times. The protocols I used were similar to those encountered and used in a variety of environments. The AES/EBU digital transfer method, the coaxial-S/PDIF digital transfer method, and the unbalanced analog outs were all the protocols I used. For the digital transfers, I employed a Panasonic SV-3800 because it is one of the few DAT recorders on the 8th floor that has both AES and S/PDIF digital ins and outs.

Having both digital transfers on the same machine allowed me to compare the transfer protocols with a control of using the same recorder. For the analog transfers, I was interested in recreating the worst possible scenario as a kind of low-end quality reference. The consumer Sony DAT machines on the rollaround carts, having only unbalanced analog ins and outs and presumably average quality A/D/A converters, served this purpose. I conducted the transfers up until the 40th generation, saving each 5th generation for comparative purposes. With 40 generations and an extra playback in between, it meant that each musical example had gone over the playback heads 80 times. In retrospect, the amount of difference between 5 generations was so minimal (if at all) that it would have probably been more efficient and more inductive of tape loss if I had just continued recording the examples over the same sections of tape. However, with my methods, the resulting location of the generations on the two tapes ended up as follows:

Time	Tape I	Tape 2
0:30	Master	AES 5th generation
1:35	AES 10th generation	AES 15th generation
2:40	AES 20th generation	AES 25th generation
3:45	AES 30th generation	AES 35th generation
4:50	AES 40th generation	Coaxial 5th generation
5:55	Coaxial 10th generation	Coaxial 15th generation
7:00	Coaxial 20th generation	Coaxial 25th generation
8:05	Coaxial 30th generation	Coaxial 35th generation
9:10	Coaxial 40th generation	Analog 5th generation
10:15	Analog 10th generation	Analog 15th generation
11:20	Analog 20th generation	Analog 25th generation
12:25	Analog 30th generation	Analog 35th generation
13:30	Analog 40th generation	

After creating all of these dubs, it was necessary for me to both analyze them with a Fourier Transform and with my ears. To facilitate the Fourier analysis, I transferred all of the 40th generation recordings, along with the master, into Sound Designer. Sound Designer allows for a maximum Fourier analysis window of 100 milliseconds. To ensure that this 100 milliseconds was the same moment of analysis for all of the examples, I had to waveform edit each example down to a 100 millisecond sound file. I therefore cut out the first second of each sound file and save 100 milliseconds after the beginning of the 2nd second for each example. By eliminating the first second of time, I felt I was always eliminating any beginning distortions that may have been induced by the dubbing process. Since I was only interested in the overall sound quality change, I thus chose a selection more from the middle of each sound source. When I ran the Fourier anaysis, I was asked to chose from a variety of parameters in Sound Designer. After experimenting with different settings, I found the following set-up to produce the most noticeable differences in the files: **Bands:** 1024, All; **Amplitude:** Logarithmic; **Type:** Frequency. The resultant graphs are included at the end of this paper. Again, each graph shows the Fourier analysis of the 40th generation of each transfer protocol, the master recording, though of course, being original.

The graphs themselves proved to be not as useful as I had expected. The major disappoint was with the pink noise example. I had originally included the pink noise test to be the main analysis example with the Fourier transforms. The harmonic complexity of the pink noise source, however, proved to work against pin-pointing particular frequency response aberrations. One interesting thing to note was that the inexpensive Sony DAT recorders obviously have a slightly lower roll-off on the D/A side than the A/D end. Since the master for the musical examples were recorded on these machines, the existence of frequencies above 20kHz on the master argues for a higher cut-off point. Taking a look at the 40th generation analog example, we see a sharp brick wall cut-off below 20 kHz. Since the frequencies were recorded into the Sony but do not show up on the 40th generation analog, they could only have been lost on the D/A side. Traditionally, this result makes sense, since DAT machines usually have better A/D converters than D/A converters. Why is this so, one may ask? If higher resolutions can be caputered on a recorder, they can be reproduced on a higher fidelity machine; however, if these frequencies are never captured in the first place, moving the DAT to a better machine will afford no advantages.

The sine wave Fourier analysis was a little more useful. Small imperfections around the 1 kHz tone can be noted even in the master recording (around 18 ms and 41 ms). When one views the digital copies, these imperfections double (four in number). With the analog transfer, no more loss than occurred with the digital copies seems apparent; again, four small peaks around 1 kHz show up on the graph. I should mention that the strange mountainous area on all the sine wave graphs after around 80 milliseconds is a mystery to me. Since this imperfection shows up on even the master recording, I can only assume that it is some function of the Fourier analysis (perhaps a multiple of the frequency of analysis) or an imperfection with the original sine wave itself. Also possible is that the original A/D conversion already distorted the waveform enough to be so obviously apparent. However, since the distortion does not seem to get any worse with the 40th generation analog copy, I feel it was more probably an error in analysis or with the source itself.

The third example of Sonic Youth has similar problems with analyzing as the pink noise example. The harmonic information is so rich and dense that small imperfections become lost in the landscape of peaks. However, in general, the analog copy of the Sonic Youth example does seem to have a significantly smaller amount of high frequency information than the master or the digital copies. Up until about 16 kHz, the 40th generation analog copy seems to be as rich as the others, but afterwards only has a few peaks in comparison. Since high impedance unbalanced cables are known for high frequency roll-off with long runs, I would surmise that, even though the cable lengths were average, that they probably are to blame for the noticeable slight roll-off in response.

With the fourth example, I felt I had struck the best example for analysis. The master recording has some small peaks around 2 kHz which become crucial and obvious factors for comparison. The AES and analog copies seemed to have added slight harmonic distortions above 2 kHz (noticeable at 8 and 12 milliseconds respectively). The coaxial S/PDIF copies, however, seemed to lose a lot of the 2 kHz peaks represented in the original without adding any more high frequencies to signal. In summary, therefore, no transfer protocol is exactly perfect with respect to recreating the precise harmonic spectrum of the original recording after 40 generations and 80 plays. However, the true test of quality lies with the ears, not the computer.

To discern any noticeable sonic quality differences, I burned a compact disc (included with this paper) to listen accurately to all the examples. I chose Studio B on the 8th floor because of its controlled listening environment and high quality powered Genelec speaker system. With both the pink noise and sine tone examples, I was unable to discern an appreciable difference between the master and any of the digital copies. The sounds of pink noise and a sine wave are too foreign, presumably, for the average listener to pick up on subtle timbre changes. With the analog recordings of both examples, however, there was a noticeable, although not altogether severe, increase in the noise floor. This decrease the average signal-to-noise ratio was probably a result of a couple factors, including the noise involved in continous A/D/A conversions and the noise induced by multiple unbalanced cable runs.

When it came to the musical examples, I was truthfully again unable to notice any difference between the digital copies and the master with the Sonic Youth rock selection. Again the analog copy had a rise in the noise floor, but once the music started, this background noise was covered and masked by the music itself. For the Chopin piece, I thought I could perceive a small modulation in the tape hiss (recorded on the original compact disc from the original recording session I imagine) with the digital copies. Basically, however, the overall sound quality was identical to that of the master. The analog dub was a little noiser, but was in general completely listenable.

In none of the selections did I experience any obvious dropouts or burst errors. I was careful to clean the DAT recorder heads before each dubbing session, a factor which may have significantly reduced the chance for such problems. Since I could not tell the difference between the digital copies and the master except for a slight possible modulation of the high frequency sounds, I have to admit that digital transfer of DAT information seems rather safe up until the 40th generation (which I cannot envision occuring in a day-to-day scenario). Also, despite the slight increase in the noise floor, I was impressed with the overall quality that the analog copies retained. Presumably, with just early generation copies, analog transfers are sonically unrecognizable from the master or digital copies. In summary, I feel that DATs are rather safe recording medium. Perhaps most of the fear about DATs stems from unclean tape heads or mistreatment of the tapes themselves (exposure to extreme temperatures or moisture). In general, it is of course safest to store master recordings on a multitude of mediums and in a variety of places. DAT, in conclusion, is a sturdy format for such storage purposes.

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