INTRODUCTION TO PROBLEMS OF *IN SITU* PIPE ORGAN SAMPLING

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Abstract: Contemporary possibilities of digital recording and processing of audio signal together with computer technology capabilities enable high fidelity simulation of acoustical instruments. Motivations of pipe organ sampling are for example building of sound archives of historical and valuable instruments, study of interpretation and registration and for practicing purposes. Classical pipe organ placed in real space, most frequently sacral room, represents quite complicated problem for methodology of sampling with respect to acoustical qualities, organ stop list, instrument condition, etc. This contribution describes main problems, which must be solved when developing a method of pipe organ sampling in such rooms. We are speaking mainly about microphone technique, microphone placement and audio signal recording. The summary of main steps of digital sample processing regarding to true sounding simulations of classical pipe organ and computer system optimization for their unproblematic playback are discussed separately.

1. Introduction

Sampling methods for wide range of customers were firstly introduced in early 80ies of 20th century. There were very limited possibilities of samplers in those days and they have rapidly improved until today, thanks to technology development. Instead of special sampling instruments (hardware), we use mostly special computer programs (software samplers) in combination with powerful computer workstations. There are many of advantages of using such a software instruments: there could be one sample for each independently operable pipe on the organ, that is, for each manual, with each stop engaged one at a time, each key could be played separately. It is possible thanks to large amount of RAM space and also direct-from-disk sample streaming. The other advantage is large amount of playable samples in time. Their number is limited only by CPU power. Last but not least, sample set of virtual organ could be changed very fast, only by loading them in RAM. In this way, it is possible to test many kinds of instruments, with different stop-sets etc. There are many of specialized software programs for playing back organ samples in real time. One of them is called „Hauptwerk“, developed by Martin Dyde. Hauptwerk is most often used for study and practice at home by organists, organ enthusiasts and music students, for historical organ and music study and research, for making playable documentary recordings of endangered or valuable pipe organs and in commercial and home recording studios to provide the ultimate pipe organ sound.

2. Organ sample Recording

The samples should be saved in one of many hi-resolution formats. The absolute minimum is 16-bit, 44100 Hz stereo, uncompressed. Recordings can (and should) be made in the highest resolution available and converted to this format later when processing the samples. When recording direct to computer with hi-resolution analog to digital converters, we can use for example WAV or AIFF sample formats.

There are two main approaches to sampling pipe organs:

- ***Ambient recording***: use stereo microphones (or some of well known stereo system, such a M-S, AB, XY, ORTF etc.) kept in a constant position whilst recording each note on each stop on the organ (or at least for whole divisions). With this method, the real reverberation (*reverb*) of the building is recorded into each pipe sample, and relative volumes of pipes as they sound in the building are preserved. Furthermore, other spatial characteristics such as phase difference between the two stereo channels are preserved accurately.
- ***Close-up recording***: use a mono microphone that is positioned close to the pipe being recorded, moving the microphone for each pipe. The aim is to record the sounds with as little reverb and other spatial acoustic

properties as possible, so that the samples are as 'dry' as possible and reverb effect could be added during playing the samples back.

When a sound is produced in a reverberant space, it radiates in many directions. Some of the sound travels directly from its source to the listener or microphone. This is the *direct sound*.

Shortly afterwards, reflections of the sound will be heard from the walls, roof and other surfaces of the space. These reflections will be rather quieter and with a slightly different frequency spectrum, since the surfaces from which they reflected will have absorbed some of their energy; more so for some frequencies than others. These are termed the *early reflections*, being those versions of the original sound that have been reflected from only one surface. After the early reflections, many other smaller and less distinct reflections are heard, having bounced off more than one surface. These are termed the *late reflections*, and there are so many of them that they cannot easily be distinguished apart.

First of all, given the property of linearity for an acoustic space, it is clear that recording reverberation in a set of pipe samples does allow for perfect reproduction of the reverb when they are played back. Because of linearity, it is true that the result of recording two pipes sounding together in a reverberant space is identical to the result of playing back a recording of each, recorded separately in the space, at the same time.

Thus, we can justifiably assert that ambient recording produces accurate reproduction of reverberation (for mono samples, at least), since the samples may safely be 'added', including the reverberation that they contain.

2.1. Close-up Recording

A single, high-quality mono microphone is placed a few inches in front of the mouth of each pipe, one at a time, whilst the pipe is recorded. Thick curtains or some other contrivance are used to prevent any reverb or room acoustics from affecting the recording. Alternatively, it may be possible to record the pipes in an anechoic chamber if they have not yet been assembled into an organ.

Assuming that it is possible to record with little or no reverberation, the benefits of this method are:

- Since there is no reverb in the sample, it is possible to play back the samples in a reverberant space (such as a church) without reverb being heard twice. This is not possible with ambient recording, since the original recorded reverb would be subject to further live reverb when played back in the space. This is the key advantage of close-up recording, that samples may be used to augment or replace a real organ in a reverberant space.
- In principal, it is possible to simulate different acoustic environments when the samples are played back. Hence, artificial reverb may be added upon playback, and its behavior adjusted as required.
- Similarly, each sample may be placed anywhere in the virtual 3D acoustic space by using modeling techniques. This is only really possible with close-up mono recording, since placing samples recorded with the ambient method also makes the reverb appear to come from the same point as the pipe sound, which is unnatural.
- The recording level of each sample can be much higher, since the microphone is much closer to the sound source, hence there is much less recorded noise and the recordings will be better quality, with less noise-reduction and other post-recording processing necessary.
- Because the samples have none of the properties of the acoustic space in which they were recorded, samples recorded from different organs can be combined, and the results should be more convincing. Playing two stops together, ambiently recorded in different churches with different amounts of reverb, would sound unnatural.
- Effects such as tremulants and enclosures (Swell boxes) can be modeled more accurately upon playback than is possible with ambient recording, since the samples can be processed with these effects before artificial reverb and spatial processing are added. Processing power is the real problem, in that per-pipe convolution is not possible, and so this is not currently feasible.
- The release phase of a pipe can be simulated a little more accurately, since reverberation need only be applied to the portion of the sample that has actually played. For example, if a very short note is played and released before the sample has reached full volume, the reverberation need only be applied to the portion of the sample that has actually sounded

(again, assuming that real-time convolution were possible).

- Mono recordings themselves occupy less memory space than stereo (or even 3D) recordings.

The main disadvantages are:

- Vastly more signal processing is required upon playback to simulate reverb and spatial acoustic information accurately, since the processing required will be different for each pipe sample.
- If the reverb and spatial processing is not performed individually for each pipe, but overall for the whole organ or divisions, the spatial information will be very noticeably inaccurate.
- If, on the other hand, such processing is performed individually for each pipe, a separate 'impulse response' sample must be stored for every pipe. Such files store spatial and acoustic information. Recording these will take as long, if not longer than recording the pipes, and the data volumes will be massively larger than the data for the samples themselves.
- There are no domestic computers powerful enough to carry out the amount of processing required if it were performed for each pipe sample in real time.
- It takes much longer to make the recordings, since the microphone must be positioned inside the organ, individually for each pipe, together with damping (such as curtains).

It is still important to record the decay of the sound (even though it should contain no reverberation), since the harmonic content of the sound will change very considerably as it ceases to speak. Simply fading out the sustaining portion of the sound upon playback is a poor approximation.

2.2. Ambient Recording

A pair of stereo microphones is placed in a fixed position in front of the organ, where they remain for the whole recording process. The natural reverb and room acoustics are thus recorded into each sample and do not need to be simulated upon playback. For other recording formats (mono, surround etc.), an appropriate number of microphones is used.

The main advantages of this method are:

- Spatial and reverb information is stored in the sample and is recorded perfectly, rather than being simulated. Hence, provided that the recording is good, all directional and acoustic information about the environment is preserved and reproduced with absolute accuracy.
- The recording process is simple and comparatively much quicker, without the need for moving the microphones, access to the organ chamber, damping, or recording impulse response files.
- The processing overheads required to play back the samples are low, and hence real-time playback is possible, without losing the accuracy of room acoustics and reverb.

The main disadvantages (being principally the advantages listed for close-up recording) are:

- The results will only sound realistic when played back in a reasonably non-reverberant space.
- Acoustic properties and placement cannot be adjusted after the recording, or to suit the environment in which the samples are to be played back.
- The sound quality (signal to noise ratio) will be lower than for close-up recording.
- Tremulants and enclosures cannot be modeled as accurately, since they must be applied across the pipe sound and its reverb, which will sound less realistic.
- Modeling of the release portion of the sample will be a little less accurate.

3. Signal Processing (post-recording)

After successful recording of sound material we must make many steps of post-recording processing. The main tasks are:

- Remove noises before/after sounding the sample by editing and de-noising algorithms during their sound.
- Calibrate relative amplitude between samples.
- Tuning the samples (if necessary – e. g. if original instrument is not well tuned).
- Looping and adding release markers. There many ways how to loop the samples, for example using Sony Sound Forge audio editor. There must be every clicks and other disturbing noises during loops removed.
- Creating organ sample script document (in therminology of Hauptwerk program “organ definition file”. This is the list of all used

samples in virtual simulation, with paths to directories, stop names etc.

4. Computer systems for organ sample playback

The computing speed of contemporary processors systems increases rapidly, but there are also other important components of computer systems, e. g. the amount of free memory required for a given sample set, usually stated as a prerequisite by the creator of the sample set. Roughly speaking, it depends upon:

- The number of samples.
- The average length of the samples. Dry samples usually require less memory because the release samples are shorter.
- The channel format of the samples - stereo or mono. Stereo requires twice as much memory as mono.
- The sample rate - 44.1, 48 or 96 kHz. Higher rates require more memory.
- The sample resolution - 16, 24 or 32-bit. Samples of either 24 or 32-bit require twice as much memory as 16-bit samples, since they are stored in memory as 32-bit for speed.

The samples are usually played-back direct from RAM, without reloading them from HD. This is the most effective method today.

Also the architecture, resolution and speed of main processor (CPU) are very important. It is strongly recommended to use modern dual-core processor with 64-bit architecture.

Last but not least, we need also high-resolution D/A converters (sound cards) with well written device drivers and low latency. The ASIO standard for sound card is recommended.

Acknowledgements

The research was supported by the Ministry of Education and Youth, Czech Republic (Project No. 1M6138498401).

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