Experimentation: Determining audio buffer size from LAN-WLAN vertical handover delay

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Introduction

The experimentation investigates the delay experienced by the application streaming audio over TCP/IP when an uninformed LAN to WLAN vertical handover occurs.

Measurements in the experimentation focus on recording the distinct cache sizes in the application receiving the media stream, and their behavior in the event of the handover. Additionally, the mobility management events and the network traffic during the handover are recorded.

The experimentation platform consists of Ericsson's hip4bsd for FreeBSD 6.1 and the PM&RG Ämppäri mobile music player, both versions developed in the MERCoNe project.

Hypothesis

The hypothesis is, that it is possible to determine the required cache size for the playing application from the cache size recordings done during the vertical handover.

Experimentation setting

The required cache size is specified as the minimum amount of cache as seconds of uncompressed audio that will not cause a depletion in the hardware or operating system audio buffer. The cache size is considered insufficent if the handover causes depletion in the OSS/hardware buffer, the size of which the software does not explicitly state.

The experimentation setting consists of a stream player and a stream sender, connected through a router. Both the stream player and stream sender use HIP, and the stream sender is connected to the router via both LAN and WLAN and uses LAN by default.

Experimentation procedure

The experimentation consists of the following procedure repeated 20 times for each cache size setting from 8192 bytes to 81920 bytes and for two distinct media bitrates of 192 and 320kbps. Constant bitrate MP3 audio is used for streaming, since every packet of audio data uncompresses to the same amount of audio.

1. Stream sender opens a connection to the stream player and starts playing an audio stream.
2. Playback continues for 15 seconds, after which the LAN cable is disconnected, forcing a switch to WLAN.
3. Playback continues in WLAN for 15 seconds.
4. Recordings are saved to the disk on program exit.

Recording file details

The audio processing path follows the conventions of typical audio players, there are three distinct audio buffers, the compressed audio packet cache, uncompressed ring buffer to which the audio packets are uncompressed and the OSS/ hardware audio buffer to which the uncompressed audio data is fed.

All data, incording the ESP packet indexes and the mobility management events are transformed into value-timestamp pairs, that are analyzed in MATLAB.

Phase	Methods	Result
Data collection	Recording of cache sizes, traffic	Packet cache size Ring buffer size OSS audio buffer size ESP packet capture LINK_DOWN timestamps HANDOVER timestamps
Data processing	Determining delays from recordings	Delay in ESP packets (ms) Delay in audio packets (ms) LINK_DOWN reaction time (ms) HANDOVER reaction time (ms)
Data analysis	Determining maximum handover time and typical delays	Maximum handover delay Audio packet delay distribution Reaction time distribution

Table 1: Summary of methods

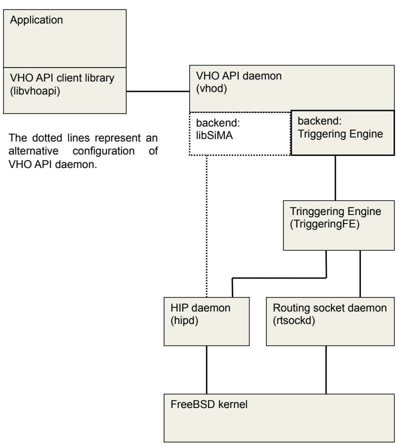

Figure 1: Mobility management architecture

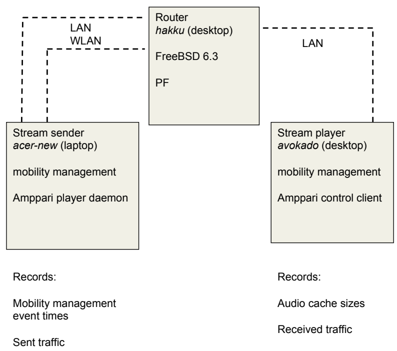

Figure 2: Test network configuration

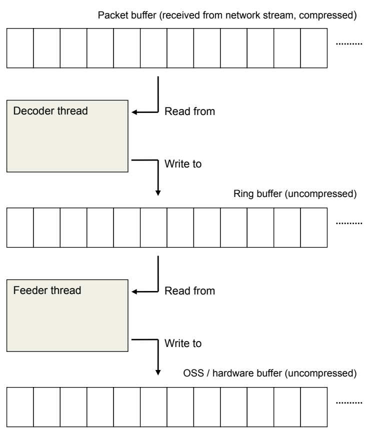

Figure 3: Audio processing path

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