SIMULATION HEART BEAT MONITORING SYSTEM

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A thesis submitted in fulfillment of the

requirements for the award of the degree of

Bachelor of Computer Science and Software Engineering

Faculty of Systems Computer & Software Engineering
University Malaysia Pahang

APRIL, 2010

ABSTRACT

Simulation Heart Beat Monitoring System (SHBMS) is a system which applies technique of signal processing. First of all, the input shall be digitalized into a form which understands by the machine before start analyzing it. Then, the tool used to develop SHBMS is Visual Studio 6 (VB6). There is a library available and able to embed into VB6 in handling the input and generate output in waveform signal. In SHBMS, a simulation input such as audio file of heart beat or beats on microphone shall be used to replace a real heart beat from a human being. The idea of using simulation input is due to the difficulty of capturing real heart beat using a sensor or device. Yet, the SHBMS still able to detect input and react according to the input. The output is generated according to the size of byte (beat per minute) and shall give comment upon the size of byte. Furthermore, waveform signal shall be displayed as well to show the changes of the signal which high or low. This indicates the signal of the heartbeat. Moreover, SHBMS is developing according to Software Development Life Cycle process and able to finish within the proposed period.

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Chapter 1

Introduction

1.1 Introduction

Heart disease most often occurs when cholesterol accumulates and forms "plague" in a coronary artery. With blood flow impeded, the heart becomes starved for oxygen, causing chest pain (angina). Coronary artery disease, the restriction of blood flow to the heart, is the leading cause of death in Malaysia. If a blood clot forms and completely obstructs the artery, a heart attack (myocardial infarction) can occur. The statistical death caused by heart attack in Malaysia has been increasing from one year to another. This shows that heart disease is one of the horror killers living in every human being. During a heart attack, heart muscle is depraved of oxygen and will literally die if the artery remains blocked. The first few hours are critical in saving much of dying muscles and preventing permanent heart damage. Unfortunately, the symptoms vary and the most common reason for critical delays in medical treatment is lack of early warning and patient unawareness.

1.2 Problem Statement

Based on my case studies, I found several problems. There are:

i. Heart attack patient dies in silent condition without anyone noticing such as sleeping.

ii. Does not have constant monitoring of the heart condition.

1.3 Objective

The objectives to create Simulation Heart Beat Monitoring System are:

i. Digitalized the wave file into signal wave form.

ii. Simulation input will be used upon the recognition of heart beat.

iii. Presenting the output in waveform signal form according to the triggered input.

1.4 Scope

The scopes of my project in this SHBMS are;

i. Pattern recognition and frequency of heart beat in signal processing.

ii. Calculation of peak and wave length of the signal obtained from simulation input.

iii. Prompting command according to the beats per minute.

Chapter 2

Literature Review

2.1. Digital Signal Processing (DSP) – Real Time Digital Signal Processing by Thad B. Welch, Cameron H.G. Wright and Michael G.Morrow; Taylor & Francais Group

Whenever the approach to obtain a real world signal in order to process it digitally, firstly, must first convert it from its natural analog form to the more easily manipulated digital form. This involves grabbing, or "sampling" the signal at certain instants in time. Assuming that the sampling instants are equally spaced in time (T_s), so that the sampling frequency (F_s) is equal to $1/T_s$. Each individual sample represents the amplitude of the signal at the instant in time, and the number of bits per sample that being used to store this amplitude to determines how accurately we can represent it. More bits means fidelity, but it also means greater storage and processing requirement.

One potential problem that can occur during sampling is called aliasing, which results in samples do not properly represent the original signal. Once aliasing has crept into the data, no processing in the world can fix the samples so that the original signal can be recovered. To prevent aliasing, the sample frequency, F_s, of the ADC (analog to digital converter) must be greater than twice the maximum frequency f_h contained in the analog input signal. Often the sample frequency is considerably higher $2f_h$. typically, some form of input signal conditioning (such as an analog lowpass filter) ensure that the maximum frequency contained in the analog input signal is less than $F_s/2$.

A generic DSP system

Filtering is one of the most common DSP operations. Filtering can be used for noise suppression, signal enhancement, removal or attenuation of a specific frequency, or to perform a special operation such as differentiation and integration. Filters can be thought of, designed and implemented in either the sample domain or the frequency domain. The notation used in many continuous-time signals and system texts is to label the input signal as x(t), the output signal as y(t), and the impulse response of the system as h(t). These time domain descriptions have frequency domain equivalents; they are obtained using the Fourier transform, which is shown F{}. The Fourier transform x(t) is F{ x(t) } = X(jw); similarly the Fourier transform of y(t) is Y(jw) and that of h(t) is H(jw). H(jw) is also called the frequency response of the system. These Fourier transform pair are summarized as below ;

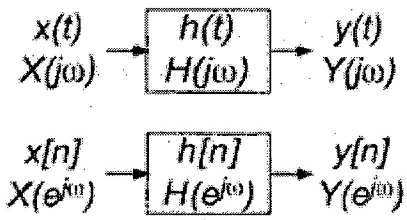
$$x(t) \xleftrightarrow{\mathcal{F}} X(j\omega)$$
$$y(t) \xleftrightarrow{\mathcal{F}} Y(j\omega)$$
$$h(t) \xleftrightarrow{\mathcal{F}} H(j\omega)$$

The most common notation used in discrete-time signals and systems texts is to label the input signal samples as x[n], the output signal samples as y[n], and the impulse response as h[n]. Note the discrete-time impulse response h[n] is called the unit sample response in some texts. As in information, parentheses "()" will be used to denote continuous-time, while square bracket "[]" will be used to denote discrete-time. Discrete-time descriptions (such as x[n], y[n] and h[n]) have frequency domain equivalents that are obtained using the discrete-time Fourier transform (DTFT), which abbreviated as F{}. The DTFT of x[n] is F{ x[n] } = X(e^{jw}), of y[n] is Y(e^{jw}), and of h[n] is H(e^{jw}). H(e^{jw}) is also called the frequency response of the system.

$$x[n] \xleftrightarrow{\mathcal{F}} X\left(e^{j\omega}\right)$$
$$y[n] \xleftrightarrow{\mathcal{F}} Y\left(e^{j\omega}\right)$$
$$h[n] \xleftrightarrow{\mathcal{F}} H\left(e^{j\omega}\right)$$

Notice that the abbreviation used for the continuous-discrete Fourier transform and the DTFT are the same because the context should make it clear which transform is used. For example, if the signal or system being transformed is a discrete-time signal or system, it will be implied by the square bracket notation and thus the DTFT should be inferred. Also of interest is the fact that the DTFT of a discrete-time signal.

To calculate the output of a continuous-time system that has been given a continuous-time input signal we need to convolve the input signal with the system's impulse response. Since this involve continuous signals, integration will be used (discrete signals use summation instead of integration). Thus to calculate the output we need to evaluate the convolution integral. This an operation that many beginning students find to be mysterious and intimidating.



2.2. Dynamic Calibration of Oscilloscope and Waveform recorder using pulse standards – William L.Gans Electromagnetic Field Division, National Institute of Standards and Technology

Oscilloscopes allow us to measure and "see" (through a CRT display and/or hardcopy plot) a functional representation of voltage versus time. With the proper sensors or transducers connected to the oscilloscope, the user not only can measure voltage versus time but parameter of interest versus time. As we would expect with this versatility and utility, there are practically an infinite number of applications for these instruments in science and technology. In addition, there are practically an infinite number of different models available for all of these applications, with innumerable knobs, buttons, and other specialized features.

With all of the available (or future) variations, however, there is really only one basic characteristic of any oscilloscope that ultimately matters to most users. That is, how well does the displayed or recorded voltage-versus-time graph represent the true shape of the signal being measured? Disregarding all of the knobs, buttons, and fancy features, how do we verify that the observed waveform accurately represents the true signal? For decades, as well as at present (with few exceptions), this verification, or "calibration," question has been addressed by performing three basic types of tests.

The first type of test is concerned with the gain and linearity of the voltage (usually y-axis) channel. These tests are designed to verify an accurate mapping between the voltage at the oscilloscope input and the displacement of the electron beam (or other indicator). Usually they consist of applying known dc voltages to the oscilloscope input terminals and visually verifying the accuracy of this mapping over the entire range of permissible voltages.

The second type of test is concerned with the gain and linearity of the time (usually x-axis) channel. Analogous to the voltage channel case, these tests verify the accuracy of the mapping between the passage of time and the displacement of the electron beam. Usually these time channel gain and linearity tests consist of applying a set of time-varying voltages of fixed and known period to the oscilloscope input terminals and, again, verifying the accuracy of this mapping over the entire range of permissible sweep speeds.

The third basic type of test is concerned with the speed of response (displacement) of the voltage channel. The laws of physics dictate that no realizable electronic circuit can produce an instantaneous change at its output in response to an (almost) instantaneous change at its input. As a result, every oscilloscope has some upper limit on speed of response beyond which it will no longer accurately follow the rapid voltage variations of the input signal. The most common test for verifying this speed of response is a two-step process. First a "low" frequency sine wave of known amplitude is applied to the oscilloscope and the displayed amplitude is noted. Then, keeping the amplitude constant, the frequency of this sine wave is increased until the displayed amplitude falls to some prescribed value, usually to 70.7% of its original displayed amplitude. This test, then, determines the frequency at which the voltage channel gain falls **3** dB from its "low" frequency gain. It is often called an oscilloscope "bandwidth test."

2.2.1. The Problem

If the gain and linearity tests of the voltage and time channels are performed properly, they are both correct and sufficient as outlined above. The problem is with the bandwidth or risetime test; it is correct but not sufficient. If we wish to measure and view the shape of voltage-versus-time signal except a sine wave, then the speed-of-response test, outlined above, is inadequate. Simply knowing the bandwidth or transition duration of the oscilloscope voltage channel is not sufficient to guarantee the accuracy of the shape of the displayed or recorded waveform.

To understand why this is true, we need only to consider the Fourier series expansion of any given voltage-versus-time signal. A nonsinusoidal, periodic signal consists of the sum of an (often large) number of harmonically related sinusoids, each with its own amplitude and phase. In the time domain, the shape of this signal is affected by the amplitude and phase of each ofthose sinusoids. Therefore, if the oscilloscope voltage channel does not accurately maintain these amplitude and phase relationships completely, the shape of the displayed waveform can be drastically different from that of the input signal. Simply knowing the frequency at which the voltage channel gain is reduced by some number of decibels tells us very little about the voltage channel's ability to maintain these amplitude and phase relationships. In fact, the *a* way to verify that these relationships are being maintained over some desired frequency range is to measure them. In other words, we have to measure the entire comvlex transfer function of the voltlage channel over the frequency range of interest.

As a simple illustration of this problem, Figure 2.1 is the full-scale display of a 500 mV step-like pulse (first transition duration of about *500* ps) as recorded on a very fast sampling oscilloscope. The bandwidth of this oscilloscope is 20 GHz, which far exceeds the discernable spectral content of this pulse. Therefore, we can assume that the displayed shape of this pulse in Figure 2.1 is accurate.

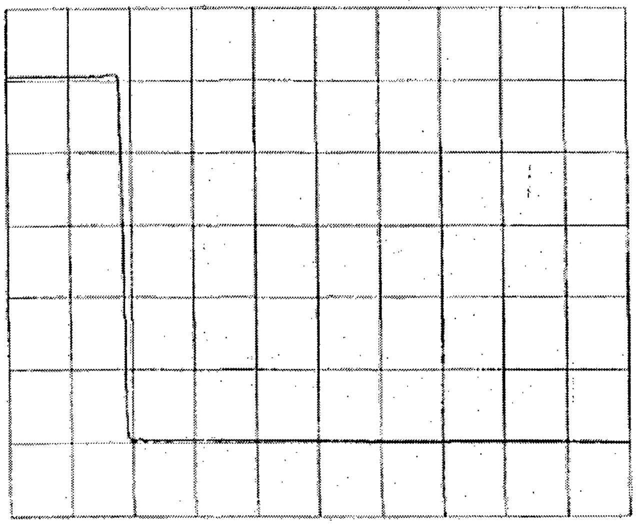


Figure 2.1 : 500mV step-like pulse recorded on a 20Gz sampling oscilloscope. Voltage scale is 100mV/division and time scale is 5n/s division.

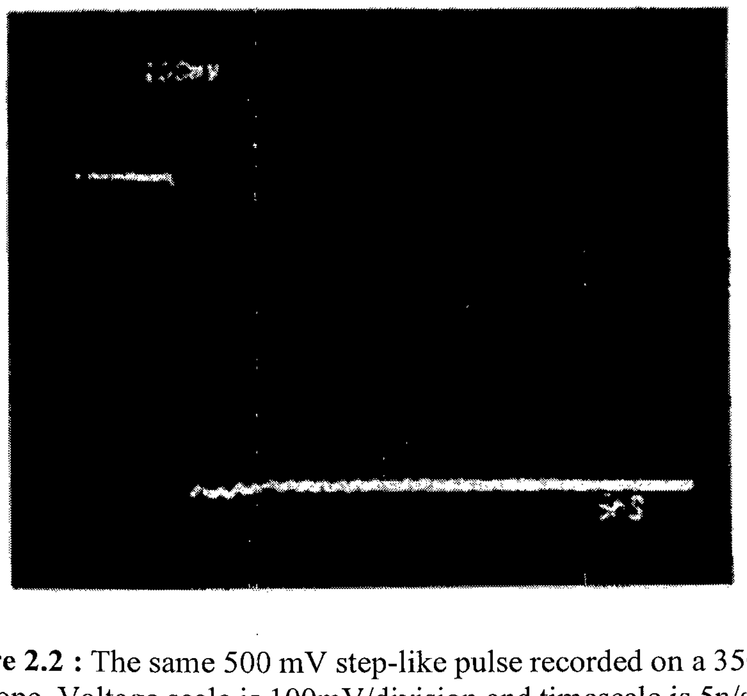


Figure 2.2 : The same 500 mV step-like pulse recorded on a 350Mhz oscilloscope. Voltage scale is 100mV/division and timescale is 5n/s division

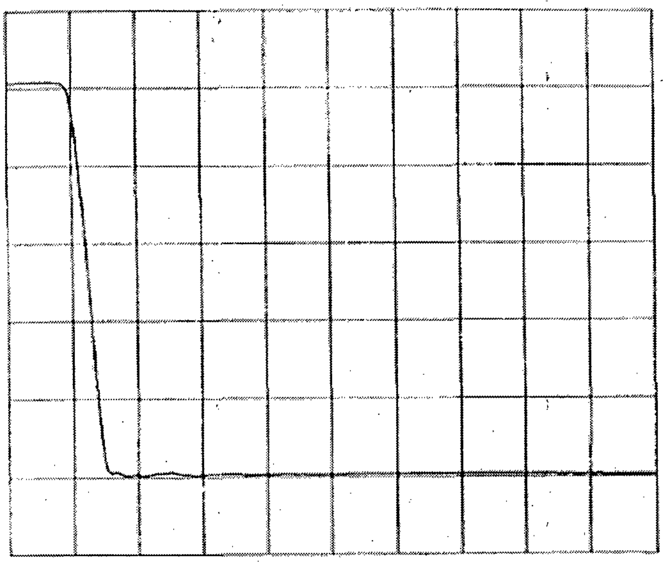


Figure 2.3: Same as figure 2.1 except sweep speed has been increased to 1ns/divison

Figure 2.2 is the display of the same pulse as recorded on an oscilloscope with a specified bandwidth of *350* MHz. The waveform shapes in Figures 2.1 and 2.2 appear to be closely related but they visibly differ somewhat. The oscilloscopes' sweep speeds were increased by a factor of 5 for the recordings in Figures 2.3 and 2.*4*. Now the displayed transition duration of the 20 Ghz scope waveform (about *500* ps) is very different from that displayed on the *350* MHz scope (about 1000 ps). In addition, and almost as important, the aberrations (or "wiggles") on the displayed waveform baselines are noticeably different, a sure sign that the required amplitude and a phase relationships of the 350mhz scope are being distorted

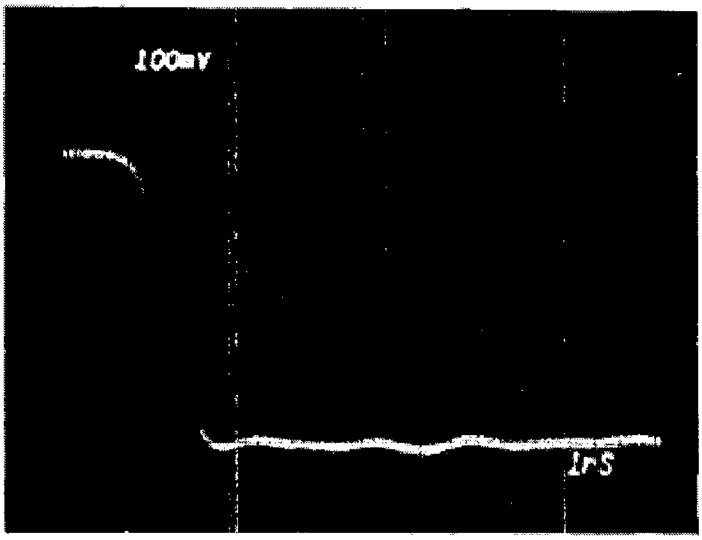


Figure 2.4: Same as figure 2.2 except sweep and speed has been increased to 1n/s division

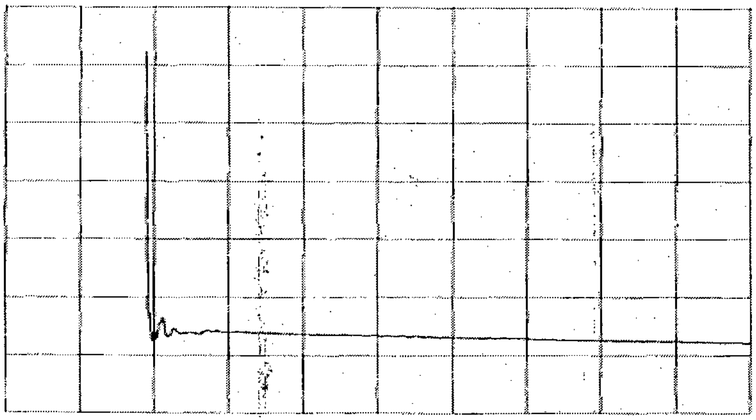


Figure 2.5: Same as Figure 2.1 except voltage scale has been expanded to 10 mV/division

2.3. An Open-Source Algorithm to Detect Onset Arterial Blood Pressure Pulses by Harvard University – MIT Division of Health Sciences and Technology

The ABP waveform contains rich information about the cardiovascular system, such as heart rate, systolic, mean, and diastolic arterial pressures, and it can be used to assess properties of the arterial vessel wall [l, 21. Reliable and accurate ABP pulse detection is crucial for beat-by-beat extraction and analysis of the information mentioned above. This task is rendered difficult, however, since the ABP measurement is prone to noise and artifacts (see Figure 3.1 (b) and (d)). Furthermore, the waveform morphology can change dramatically, even over short periods of time, in response to altered pathologic or hysiologic stresses (Figure 3.1 (a) and (c)). Although it is the ABP pulse onset that denotes the arrival of the arterial pressure pulse at the recording site, most ABP pulse and pulse-component detection algorithms identify the peak of the ABP waveform as the fiducial mark of the ABP pulse [3, 4, 51. The strategy of peak detection rather than pulse onset detection is inappropriate for studying pulse wave velocity [61 and ECG-ABP delay time [7] characteristics, as the duration of the upslope depends, among other things, on ventricular and valvular properties. This study presents an algorithm that determines the onset of arterial pressure pulses by first converting the ABP waveform into a slope sum function (SSF) signal. Subsequent adaptive thresholding and local search strategies allow for ABP onset annotations to be placed in close proximity of the actual pulse onset. We used two different databases for performance evaluation. Our results show that this algorithm is effective in detecting and annotating ABP onsets.

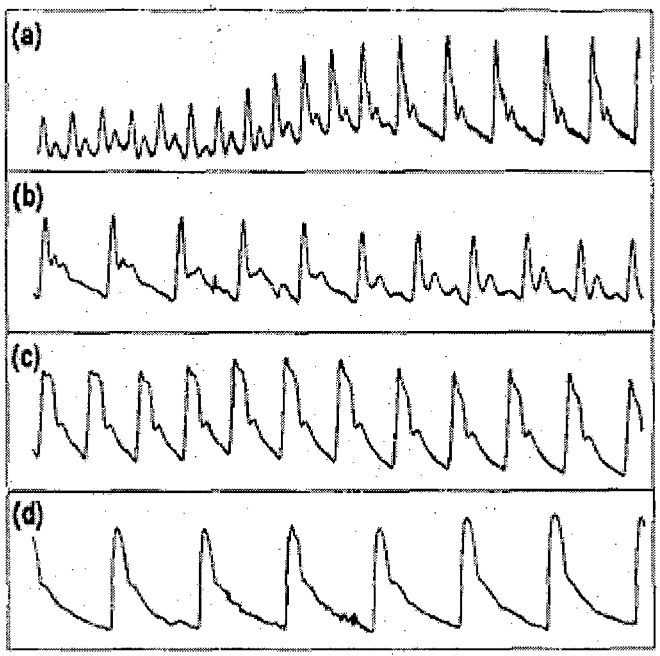


Figure 2.6: Example of ABP waveform signals. (a) and (b) are non-invasive (FINAPRES) recordings; (c) and (d) are invasive ABP recording from the radial artery; 10 seconds per trace.

2.3.1. The algorithm

As shown in Figure 3.2, the algorithm consists of three component: a low pass filter, a windowed and weighted slope sum function and a decision rule. The ABP signal, x_{nr} is the input of the low-pass filter, and y, is the filtered ABP. The slope sum function converts y_n *to* a slope sum signal *t,*.. A decision rule is applied to .z_n *to* determine the AEJP pulse onsets denoted by t_l, t_z, ___.

Figure 2.7: Algorithm Flow Diagram

Low-pass filter: The purpose of the low pass filter is to suppress high frequency noise that might affect the ABP onset detection. We use a second order recursive filter [8]

whose transfer function, frequency response, and difference equation are given below for a sampling frequency of 250 Hz (Le.. a sampling interval T = 4 ms).

$$H(z) = \frac{(1-z^{-5})^2}{(1-z^{-1})^2}, \qquad |H(\omega T)| = \frac{\sin^2(3\omega T)}{\sin^2(\omega T/2)}$$

$$y_n = 2y_{n-1} - y_{n-2} + x_n - 2x_{n-5} + x_{n-10}$$

The 3 dB cut-off frequency is about 16 Hz and the gain is 25 at 0 *Hz.* The phase shift is 20 ms (5 samples at 250 Hz).

Slope **sum** function: The purpose of the slope sum function is to enhance the upslope of the ABP pulse and to suppress the remainder of the pressure waveform. The windowed and weighted slope sum function at time ***i, t i***, is defined **as** follows:

$$z_i = \sum_{k=i-w}^{i} \Delta u_k, \quad \Delta u_k = \begin{cases} \Delta y_k & : \ \Delta y_k > 0 \\ 0 & : \ \Delta y_k \leq 0 \end{cases}$$

where w is the length of the analyzing window; 1 + w 5 i 5 N, N is the total number of ABP samples in the record, Ays = yk - yk-1. and yk is the low-pass filtered AEJP signal as defined above. To maximize the SSF, w is chosen approximately equal to the typical duration of the upslope of the ABP pulse. In the present algorithm, w = 128 ms or 32 samples for the sampling frequency of 250 Hz. The relationship between the AEJP and the SSF signals is shown in Figure 3. The onset of the SSF pulse generally coincides with the onset of the ABP pulse as the SSF signal can only rise when the ABP signal (or noise not removed by filtering) rises.

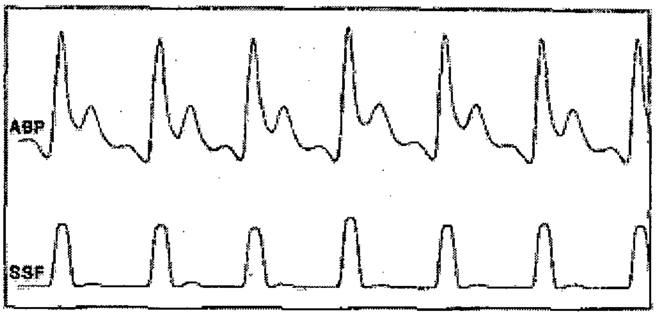


Figure 2.8: Relationship between ABP and SSF signals.

2.3.2. Evaluation Procedure

To evaluate the performance of the algorithm, we first assessed the accuracy of pulse detection and subsequently evaluated the accuracy of pulse onset detection.

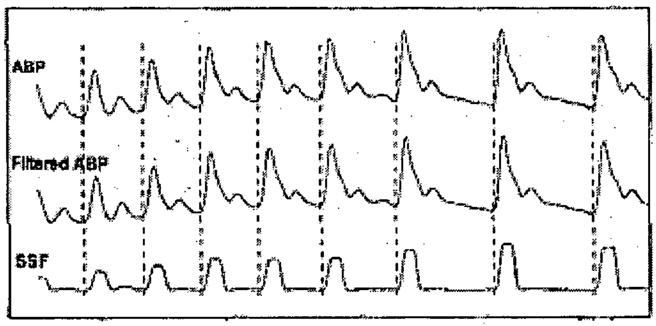


Figure 2.9: Example of ABP pulse onset detection process. Top trace; raw ABP signal; middle trace; filtered ABP signal; bottom trace; SSF signal; vertical dashed lines; ABP onset detection.