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What is inside the electrocardiograph?

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Abstract The details of digital recording and computer processing of a 12-lead electrocardiogram (ECG) remain a source of confusion for many health care professionals. A better understanding of the design and performance tradeoffs inherent in the electrocardiograph design might lead to better quality in ECG recording and better interpretation in ECG reading. This paper serves as a tutorial from an engineering point of view to those who are new to the field of ECG and to those clinicians who want to gain a better understanding of the engineering tradeoffs involved. The problem arises when the benefit of various electrocardiograph features is widely understood while the cost or the tradeoffs are not equally well understood.

An electrocardiograph is divided into 2 main components, the patient module for ECG signal acquisition and the remainder for ECG processing which holds the main processor, fast printer, and display. The low-level ECG signal from the body is amplified and converted to a digital signal for further computer processing. The Electrocardiogram is processed for display by user selectable filters to reduce various artifacts. A high-pass filter is used to attenuate the very low frequency baseline sway or wander. A low-pass filter attenuates the high-frequency muscle artifact and a notch filter attenuates interference from alternating current power. Although the target artifact is reduced in each case, the ECG signal is also distorted slightly by the applied filter. The low-pass filter attenuates high-frequency components of the ECG such as sharp R waves and a high-pass filter can cause ST segment distortion for instance. Good skin preparation and electrode placement reduce artifacts to eliminate the need for common usage of these filters.

Keywords: Electrocardiograph; ECG acquisition module

Introduction

This paper will discuss signal acquisition and signal processing of the 12-lead electrocardiogram (ECG) signal from an engineering point of view to explain the engineering tradeoffs to clinicians. Although the 12-lead ECG is used commonly by many providers throughout the healthcare system, specific knowledge regarding signal acquisition and processing is often lacking. The ECG provides a representation of the electrical activity of the heart and is extremely important as a tool for the diagnosis of various cardiac dysfunctions. Thus, the most accurate representation with the least amount of error is imperative to provide the most benefit to patients.

Functionally, the electrocardiograph can be divided into 6 blocks which will be discussed throughout the paper. These include ECG acquisition, ECG signal processing, real-time ECG display, automated ECG interpretation, storage, and transmission of 12-lead ECG reports. This paper is only concerned with the parts of the system that have a direct impact on the quality of the ECG signal. The purpose of this paper was to assist both biomedical engineers and clinicians

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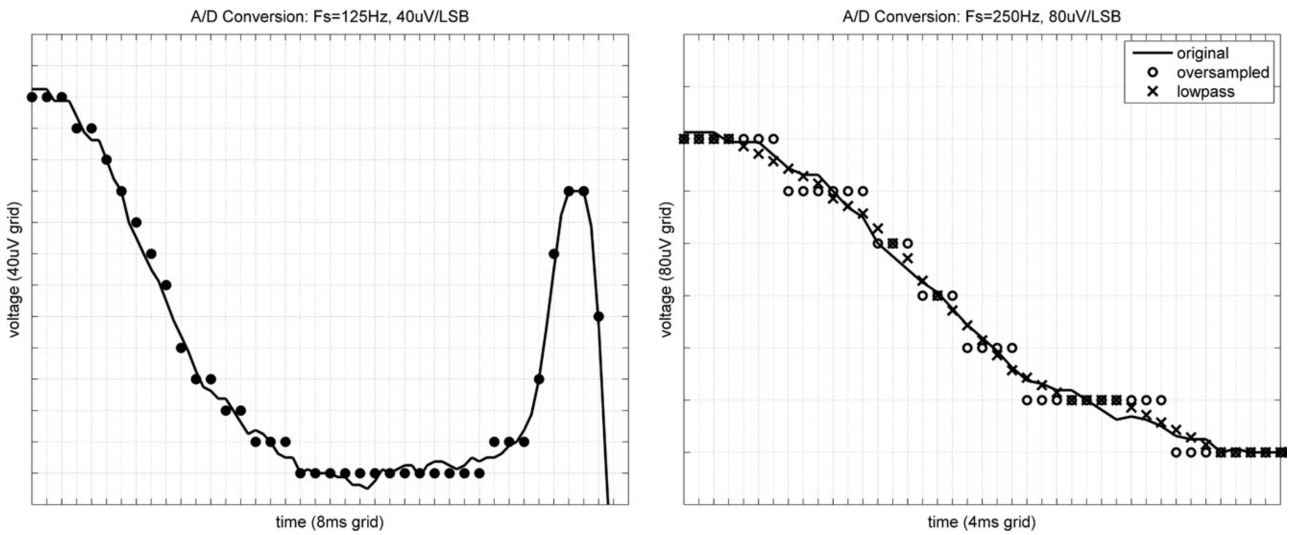

Fig. 1. Left panel: Example of analog to digital conversion as fitting a grid. The original signal, end of T wave and small R wave, is represented by dots lying on the grid. The difference in *y* value between the sample values and the true signal is known as the quantization noise. The amplitude resolution is exaggerated to illustrate the difference between the true signal and sample points. Right panel: The end of T wave is sampled at a worse amplitude resolution (*y* dimension) with a higher sampling rate (*x* dimension) as compared to the left panel. Note the coarse resolution of the original samples (○) and the much reduced differences between the true signal and the low-pass filtered samples (×).

in gaining a better understanding of factors that impact ECG data quality for clinical use.

Electrocardiogram acquisition

Patient module

The ECG signal is acquired through the patient module. In many electrocardiographs, the patient module is made of a microprocessor, an analog application-specific integrated circuit, 10-lead wires connected to the patient using adhesive electrodes, and a cable back to the main part of the cardiograph.[1] The surface ECG is a low amplitude signal recorded in the presence of significant interference. Because of the low signal amplitude and relatively poor signal-to-noise ratio, great care is necessary to obtain a quality result. The interference is actively controlled by right leg drive, discussed in detail in later sections, and then the analog signal is filtered, amplified by a gain of 1000, and converted to a digital signal to allow computer processing to give the result shown in the standard 12-lead ECG report.[2] Note that careful attention to skin preparation and electrode adhesion has a large impact in reducing artifact and interference.[3]

Analog to digital conversion

Conversion of an analog or continuous time, continuous voltage signal to a digital signal involves the process of sampling at discrete points in time. A digital signal is convenient because it can be easily processed by computer. The key to equivalence between the analog and digital signals is the fact that the analog signal can be faithfully reproduced from the digital signal. The term sampling means recording the analog signal's voltage at a regular point in time.[4] The concept of analog to digital conversion as mapping to a grid can be seen in the left panel of Fig. 1. When converted from an analog signal (solid line) which can fall anywhere on the graph to digital values (dots), as seen in Fig. 1, the digital signal can only take discrete values of voltage and time represented by the intersection of the horizontal lines of constant voltage and vertical lines of constant time. The spacing between the horizontal lines of the grid equals the voltage resolution. A typical voltage resolution, also the value recommended by the International Electrotechnical Commission performance standards for electrocardiographs,[1] is 5 μV. The spacing between vertical lines is the sampling period. The number of samples per second is also called the sample rate. At each sample point or vertical line of Fig. 1, the difference between the continuous time signal (solid line) and the samples (dots) is called the quantization noise for that sample. It is an error due to the quantizing or sampling process. An alternate method for reducing quantization noise will be discussed later in the Oversampling section of the paper.

The analog voltage of each sample is represented by an integer value (…−2, −1, 0, 1, 2, …). To convert the integer value into the analog voltage, simply multiply by the voltage resolution. A typical numerical representation of the ECG signal uses 12-bit samples. Analog signals in the range of ±10 mV can be represented by 12-bit samples with a voltage resolution of 5 μV. To represent a signal with a wider range of voltage, more bits are needed and more bits translates into more memory and therefore a higher cost for the electrocardiograph. Resolution values are sometimes quoted as voltage per least-significant-bit. If you increment the integer representation of voltage by the least-significant-bit, the change in voltage is equal to the voltage resolution. For example, an integer value of 200 corresponds to a voltage of 1000 μV when the voltage resolution is 5 μV. Increment the integer representation of the ECG signal from 200 to 201 and the voltage is 1005 μV.

Oversampling

Oversampling is used in the acquisition of the ECG signal to more easily generate a high resolution signal, both high resolution in time and amplitude. Oversampling makes use of the Nyquist theorem, which states that a signal must be sampled at a rate of at least twice its bandwidth to be able to convert back to an analog signal without error.[4] Sampling a signal at a rate much faster than the Nyquist theoretical limit is the essence of oversampling. Sampling at a faster rate means there are many more points or samples to represent the signal. Stating the Nyquist theorem another way, the signal bandwidth must be limited to half the sample rate for faithful representation. A signal with high frequency components, and therefore a wide bandwidth, must be sampled at a high rate to represent those high frequency components. The sample rate is chosen to adequately represent the desired parts of the signal. To be sure there is no high frequency content above the Nyquist limit, half the sample rate, a filter is used before sampling. The factor of 2 between the highest frequency signal component and the sampling rate in the Nyquist theorem is only valid in the ideal case. In a practical system, factors of 3 or more are used because the ideal case requires an impossible to implement ideal filter preceding the analog to digital conversion. Filters used in a sampling system are much easier to design and build for a high sample rate whether they are analog or digital filters. The main problem is designing a filter that attenuates enough in a short transition band. The transition band is the band of frequencies between the passband (what you want to keep) and the stop band (what you want to remove). It is difficult and expensive to implement a filter with a short transition band. For analog filters, expensive precision components must be used. For digital filters, expensive high precision computation is required.[5] By oversampling, the transition band can be much longer. The start of the transition band remains the same while the higher sampling rate allows the edge of the stopband to move out to half the sample rate. In a typical electrocardiograph, the sample rate is 500 Hz, and therefore the Nyquist frequency is 250 Hz. A minimum signal bandwidth is 100 Hz so the transition band in this example is from 100 Hz, the end of the filter passband, to, at worst, the Nyquist frequency of 250 Hz. A long transition band leads to simpler and less expensive filters, a factor that is important for widespread clinical use.

Another advantage of oversampling is the ability to achieve a higher voltage resolution through signal processing. High resolution ECG can be achieved using low cost lower resolution analog to digital conversion. By low-pass filtering (taking the average of a set of points is a simple low-pass filter) many points, a much higher resolution is obtained even if the original voltage resolution is low.[6] This increase in amplitude resolution, that is, resolution in the y direction, can be seen in the right panel of Fig. 1. In this example, the original signal is sampled at a coarse amplitude resolution, but a faster sample rate than the same end of T waveform in the left panel. With low-pass filtering, the difference between the true signal (solid line) and the new samples (× markers) is dramatically reduced compared to the original low resolution samples (○ markers). The original samples represent the signal with large steps but many points. Low resolution in this case means large steps in the y direction. A simple 7-point running average was used for a low-pass filter. The difference between the true signal and the samples, the quantization noise, has been reduced by the low-pass filter resulting in a lower error. This lower error is equivalent to higher amplitude resolution.

Right leg drive

Right leg drive refers to the circuit used by electrocardiographs to reduce common-mode interference. Common-mode interference is an unwanted signal that is common to all electrodes compared to earth ground (the ground wire of the alternating current [AC] wall plug). The dominant form of common-mode interference comes from the electrical power system. The frequency of the interference can be easily verified to be 50 or 60 Hz depending on the country.

The common-mode signal found when measuring ECG is many times larger than the electrical signal of interest. Electrocardiogram signals are measured as a difference between lead wires to remove the signal components common to both wires. Taking the difference would be enough to remove the interference if all the channels of the ECG input were perfectly matched, but often the skin-electrode impedance differs between electrodes resulting in a small fraction of the common-mode signal corrupting the desired difference signal (skin-electrode impedance).[3]

In addition to reducing common mode interference by taking the difference between electrodes, right leg drive attempts to further reduce common-mode interference. The right leg drive system cancels common-mode interference by a negative feedback loop. The feedback system continues to drive a small current into the right leg opposite to the common-mode signal[2] until the measured common-mode signal has been cancelled to zero or as close to zero as can be achieved. Common-mode interference is dominated by interference from the AC power system that uses a frequency of 50 or 60 Hz. If the common-mode interference were a pure sine wave (the form of the signal in AC power), the right leg drive system would output the opposite signal to cancel the original common-mode sinusoid.

Leads-off warning

The leads-off indication warns the clinician obtaining the ECG that a lead wire or lead wires are either not connected or poorly connected to the patient resulting in a poor quality signal. The quality of the connection to the patient can be expressed as electrical impedance or, to simplify, a resistance to the flow of electrical current. A high resistance, to take the analogy of water flowing in a pipe, is like a narrow pipe only allowing a small quantity of water through, whereas a good patient connection is indicated by a low resistance, analogous to a large diameter pipe allowing a low resistance to the flow of water. Electrical current can easily flow with low resistance/impedance. When a lead is off, the resistance is infinite so no current is able to flow.

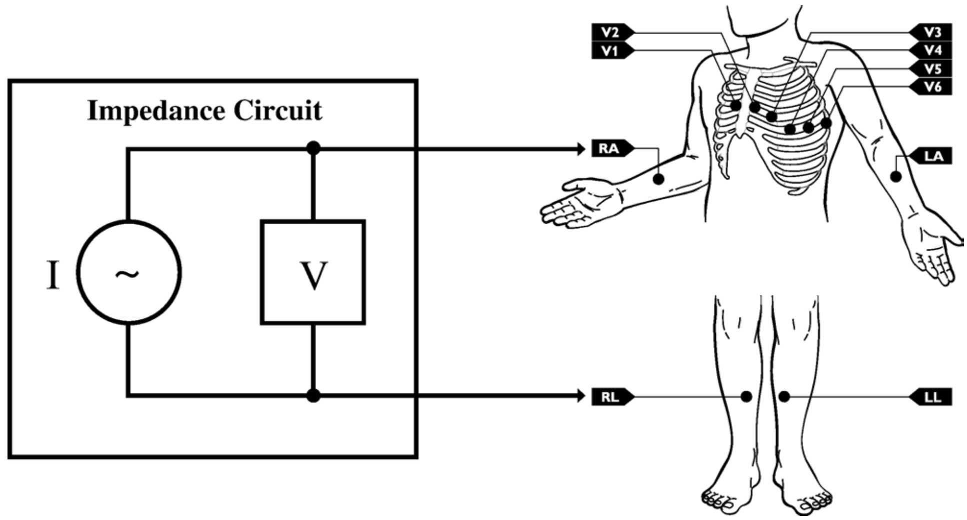

Fig. 2. Schematic diagram of the impedance measurement used for leads-off indication. A known amplitude 30-kHz AC is applied, the voltage is measured without disturbing the current, and the impedance is determined from the voltage and current relationship.

Special circuitry in the cardiograph seen in Fig. 2 is devoted to measuring the resistance between the lead wires to gauge the quality of the patient connection. A known very low amplitude current is applied to the patient, usually a direct current in most cardiographs or alternately an AC 30-kHz sinusoid. The voltage of this signal is measured and then the resistance is calculated from the well-known Kirchhoff's relationship voltage V = (current I) × (resistance R). Standards such as the International Electrotechnical Commission safety standard for electronic medical equipment[7] give a maximum safety limit on the amount of both AC and direct current that can be applied to the patient both intentionally and unintentionally. Intentional applied current for leads-off indication and unintended leakage current must be less than the amount specified in the safety standard. Patient safety against electrical shock is ensured by the very low amplitude allowed by the safety standards of the applied current for leads-off detection.

Electrocardiogram processing

Electrocardiograph

A traditional hospital electrocardiograph consists of a main processor, real-time printer, real-time display, and a means to transmit the ECG for long-term storage. Typical real-time ECG processing includes filtering, buffering, and heart rate detection. Once the operator presses the "go" button, the selected 10-second ECG record is filtered, measured, and interpreted. Lack of real-time processing constraints means that the ECG selected for the 12-lead report can receive enhanced processing. Many ECG applications use CPU time and computer resources in real-time so that the ECG signal processing can only use a fraction of what is available. That constraint does not exist for the traditional diagnostic ECG report because real-time processing is not required.

Filtering

The main purpose of ECG processing is filtering. Filters are used to attenuate unwanted components of the signal commonly called interference, noise, or artifact. The common artifacts of concern are higher frequency noise such as muscle artifact, and power line interference in addition to low frequency baseline wander or baseline. All of these common artifacts need to be minimized to get the highest quality ECG for clinical interpretation.

Power-line filtering

The fundamental frequency of the AC power available at the wall socket is 50 or 60 Hz depending on the locale. For example, 50-Hz AC power is used in Europe and Japan whereas 60-Hz power is used in North America. Although the cardiograph right leg drive removes power line interference, a small residual may remain on the ECG signal

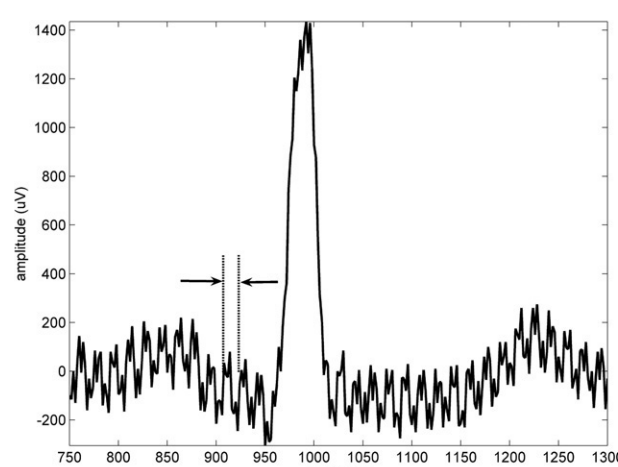

Fig. 3. Example of a PQRST complex corrupted by power-line interference. The cycle length of the interference is 17 milliseconds which corresponds to a frequency of 60 Hz.

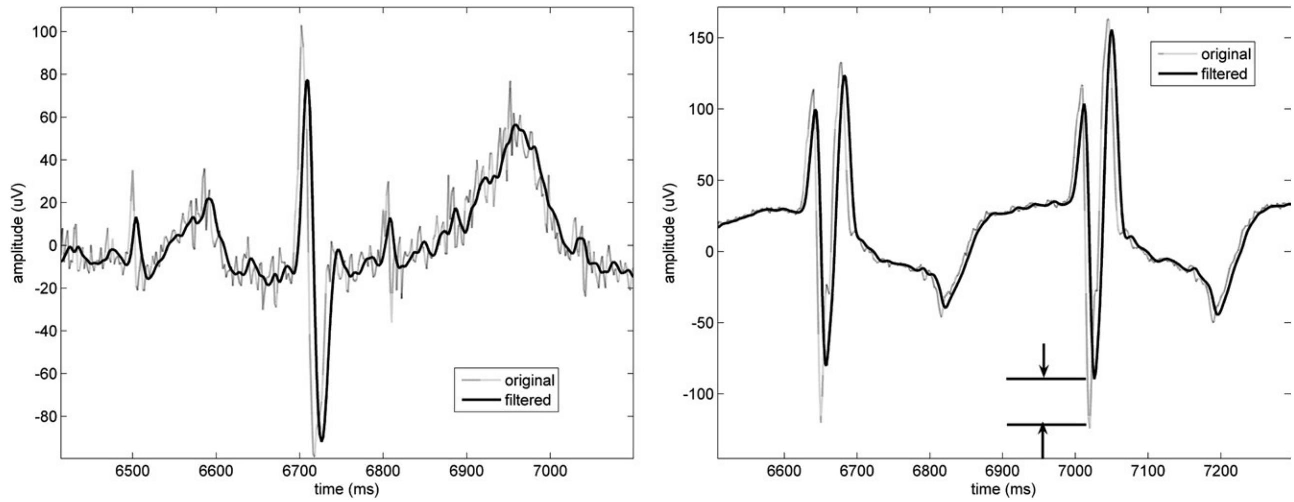

Fig. 4. Left panel: Low-pass filter example. The benefit of the low-pass filter is attenuation of high frequency noise. The filtered signal (solid line) is smoothed compared to the original signal (dotted line). Right panel: The cost of low-pass filtering can be seen in the reduction of the high frequency component of the signal—significant reduction in S wave amplitude in this case.

which can interfere with obtaining the highest quality signal. An example of power line interference can be seen in Fig. 3. The cycle length of the interference is measured to be 17 milliseconds which is equivalent to a frequency of 60 Hz as expected. A common approach to removing power-line interference is a notch filter to remove just the frequency of the interference. In this case, the filter stopband (the frequencies we want to remove) is a narrow range of frequencies about the power line frequency.[2]

Low-pass filtering

Low-pass filters are used to retain the low frequencies that are needed and attenuate the high frequencies which contribute to signal noise and a lower quality ECG. High-frequency muscle noise can be effectively reduced with a low-pass filter which is the main benefit. Typical cutoff frequencies used by 12-lead ECG machines are 40, 100, or 150 Hz. It is important for clinical application to know that the minimum recommended cutoff frequency is 100Hz for adults and 150 Hz for pediatric patients[8] and that the 40-Hz low-pass filter should only be used if absolutely necessary. This is because the 40-Hz filter will remove parts of the signal that may be of diagnostic importance thus creating a situation where important clinical information can be missed.

A noisy PQRST complex can be seen with and without the use of a 40-Hz low-pass filter in the left panel of Fig. 4. The filtered version of the signal in the foreground shows a smooth waveform, dramatically different from the original. However, there is a cost to filtering and a good reason for the Association for the Advancement of Medical Instrumentation recommendation for diagnostic bandwidth. In this case, the cost is a reduction in the amplitude of high frequency components in the signal of interest. QRS complexes have the highest frequency content of all the parts of the PQRST complex.[8] As an example, a severe 20% reduction in the S-wave amplitude can be seen in the right

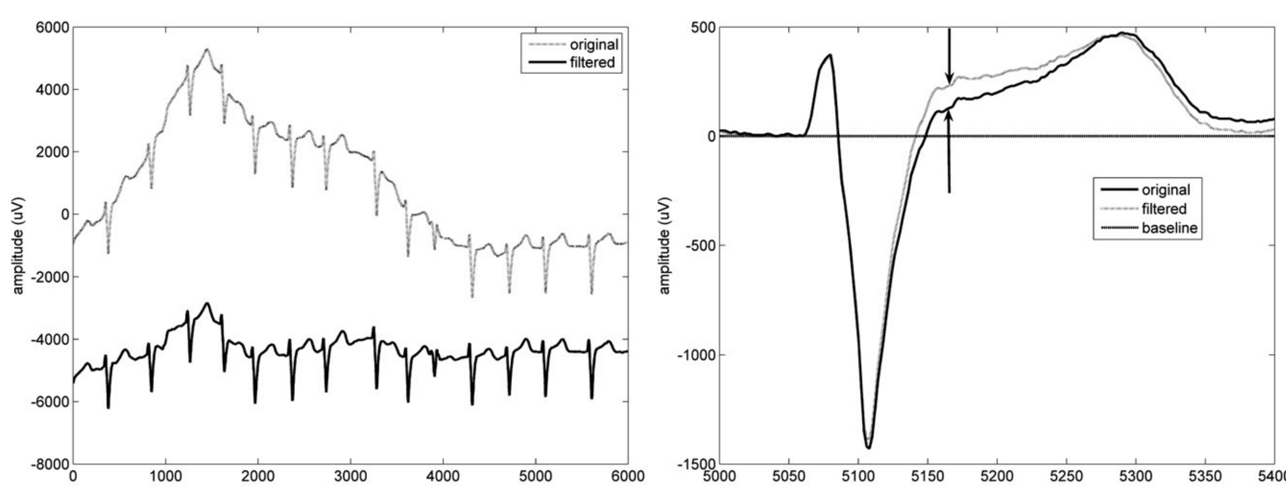

Fig. 5. Left panel: Input and output of a 0.5-Hz high-pass filter. The low frequency noise, baseline wander in this case, is significantly attenuated from input (dotted line) to output (solid line). The benefit is clear—attenuation of low frequency noise. Right panel: The cost of low frequency attenuation is ST-segment distortion. The second to last complex from both input and output are overlayed to show the ST distortion resulting from this type of simple high-pass filter.

panel of Fig. 4 as a result of lowpass filtering. The clinical impact of this reduction to sharp deflections is missing an ECG interpretation based on amplitude criteria such as left ventricular hypertrophy.

High-pass filtering

High-pass filters are used to keep high frequencies and attenuate low frequencies. Low frequency interference comes in the form of baseline sway in the ECG. Common cutoff frequencies for high-pass filters are in the 0.05- to 0.67-Hz range. Note that a frequency of 0.05 Hz corresponds to a cycle length of 20 seconds. Near the upper part of the range, 0.5 Hz corresponds to a cycle length of 2 seconds or, put another way, 5 cycles in a typical 10-second ECG. As with any filter, there is a cost and a benefit. The benefit can be seen in the left panel of Fig. 5. The upper dotted line is the original signal and the solid line is the output of the 0.5-Hz high-pass filter. The amplitude of the baseline sway has been significantly reduced. Another important feature of this example is the baseline sway that still remains after filtering. Remember that the high-pass filter attenuates low frequencies. The initial part of the baseline sway is not low in frequency compared to the filter cutoff frequency of 0.5 Hz so it is not attenuated.

Bidirectional high-pass filter

The cost of high-pass filtering can be seen in the right panel of Fig. 5. An overlay of the same QRST complex from the input and output of the 0.5-Hz high-pass filter is shown. A section of the baseline is included in the example for baseline reference in the PR segment and end of T wave. ST-segment distortion due to the filter appears between the arrows. This ST distortion arises from the filter "overshoot" which occurs because of phase distortion or a nonlinear phase response of the filter. This type of ST distortion could create a significant problem with ECG interpretation, as accurate values of ST segment elevation and depression are needed to assess many cardiac abnormalities including acute myocardial infarction.

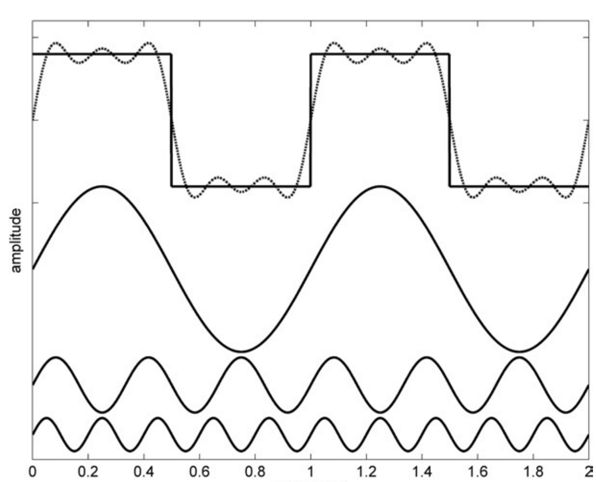

Fig. 6. Fourier approximation (dotted line) of a 1-Hz square wave (solid line) with the sum of 3 sine waves of 1, 3, and 5 Hz found at the bottom of the panel.

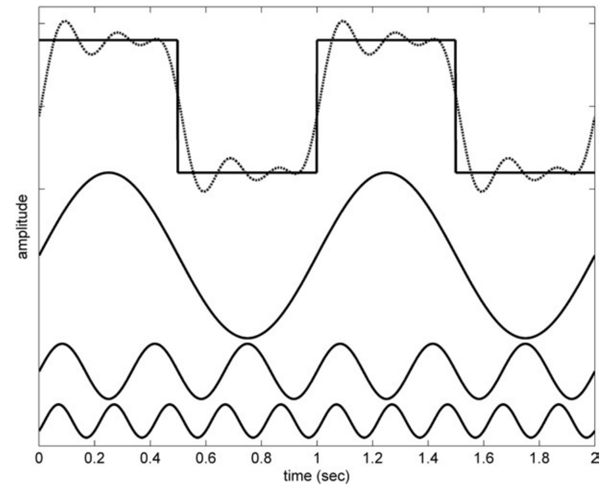

Fig. 7. The effect of delay to the highest frequency component of the Fourier approximation of the 1-Hz square wave can be seen when this figure is compared to the previous. Just a slight delay causes significant overshoot in the rising part of the square wave approximation.

The ECG signal can be considered a sum of components of different frequency according to Fourier transform theory.[9] Any signal can be broken down into component waves of different frequencies. Fig. 6 shows a 1-Hz square wave (solid line) at the top and the component waves below to approximate that square wave as an example of the different frequency components in a repeating waveform. The dotted line approximation of the square wave is the sum of the 3 waves below. The 3 component waves are the first 3 components making up the Fourier transform of a square wave. If more of the component waves of higher frequency were added, the dotted line approximation of the square wave would approach the true square wave. Using the idea of a sum of component waves making up any waveform, we can consider the effect of a filter on each component separately and then sum the components to get the output of the filter.

Phase distortion describes the fact that a filter delays components of different frequencies by different amounts so that a distortion results when the signal components are summed together again. The approximation of a square wave from Fig. 6 is repeated in Fig. 7 but with a slight delay to the highest frequency component. Comparing the square wave approximation of Figs. 6 and 7, it is clear that the slight delay to the higher frequency component has a large distorting effect on the square wave approximation.

To avoid the harmful effect of phase distortion, a linear phase filter is required instead. The linear phase filter keeps the correct delay relationship between the ECG signal components at different frequencies.[10] A simple and common way to implement a linear phase filter is with a bidirectional filter. The bidirectional filter applies the same high-pass filter in the forward direction and then again in the reverse-time direction. Processing in a reverse-time direction means that the signal in memory is processed backward from end to start rather than start to end. Any incorrect delay relationship in the forward time direction is a canceling time-

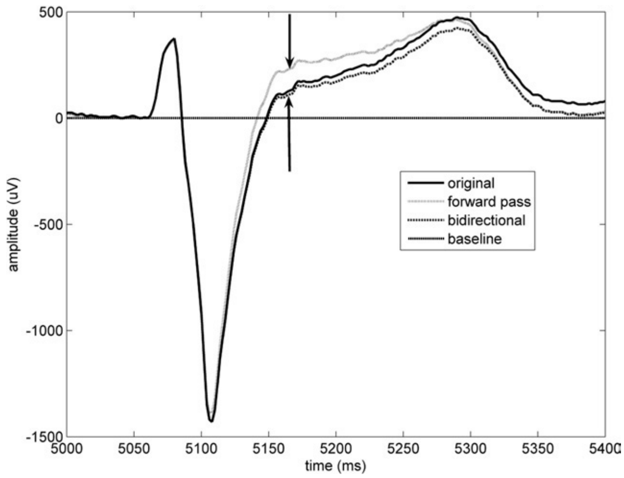

Fig. 8. The same QRST complex chosen from the original signal (solid line), the forward pass of the high-pass filter (dotted line), and the output of the complete bidirectional linear phase filter (dash-dot line). The problem with the standard forward-only high-pass filter can be seen in the ST-segment distortion between the arrows. The bidirectional filter corrects that ST segment distortion as compared to the original signal. Note that the original signal has minor ST segment distortion due to the low level of baseline drift which can be seen at the end of T wave which does not return to baseline.

advance relationship in the reverse-time direction effectively canceling the phase distortion (misalignment in time). The cost of this simple implementation is that time-reversing a signal cannot happen in real time.

The overlay from Fig. 5 is shown again in Fig. 8 including the output by the bidirectional high-pass filter (dash-dot line). The output of the first stage, the forward time filter, shows ST-segment distortion, but the ST distortion has been corrected in the final output of the reverse-time high-pass filter. The figure illustrates the improvement of a linear phase filter. The bidirectional filter described is only one of many ways to implement such a filter.

Now that the benefit of a linear phase high-pass filter is evident, it must be emphasized that the typical 0.5-Hz high-pass filtering used for monitoring is not linear phase filter and can cause ST segment distortion. To prevent ST distortion, a linear phase filter is required or a high-pass filter with a cutoff frequency of 0.05 Hz to prevent distortion.[8]

Conclusion

A better understanding of the ECG processing tradeoffs by all persons involved with diagnostic ECG should result in better quality ECGs and improved ECG interpretation. The most salient impact occurs with the use and misuse of filters as the ECG signal can be distorted by application of a filter without the knowledge of the ECG overreader. That is not to say that filters should never be used. They should only be used when necessary and when the effect of the filter is understood. Most of the artifacts that filters are designed to compensate for can be avoided by careful electrode application (good mechanical and electrical connection to the skin).

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