**Photoplethysmography (PPG)**

Photoplethysmography (PPG) is a simple optical technique used to detect volumetric changes in blood in peripheral circulation. It is a low cost and non-invasive method that makes measurements at the surface of the skin.

The technique provides valuable information related to our cardiovascular system. Recent advances in technology has revived interest in this technique, which is widely used in clinical physiological measurement and monitoring.

**Principle of PPG**

PPG makes uses of low-intensity infrared (IR) light. When light travels through biological tissues it is absorbed by bones, skin pigments and both venous and arterial blood.

Since light is more strongly absorbed by blood than the surrounding tissues, the changes in blood flow can be detected by PPG sensors as changes in the intensity of light.

The voltage signal from PPG is proportional to the quantity of blood flowing through the blood vessels. Even small changes in blood volume can be detected using this method, though it cannot be used to quantify the amount of blood.

A PPG signal has several components including volumetric changes in arterial blood which is associated with cardiac activity, variations in venous blood volume which modulates the PPG signal, a DC component showing the tissues’ optical property and subtle energy changes in the body.

Some major factors affecting the recordings from the PPG are site of measurement and the contact force between the site and the sensor.

Blood flow variations mostly occur in the arteries and not in the veins.

**The PPG Waveform**

PPG shows the blood flow changes as a waveform with the help of a bar or a graph. The waveform has an alternating current (AC) component and a direct current (DC) component.

The AC component corresponds to variations in blood volume in syncronization with the heart beat. The DC component arises from the optical signals reflected or transmitted by the tissues and is determined by the tissue structure as well as venous and arterial blood volumes.

The DC component shows minor changes with respiration. The basic frequency of the AC component varies with the heart rate and is superimposed on the DC baseline.

**Uses of PPG**

Medical devices based on PPG technology are widely used in various applications in the clinical set up.

Specific applications include the following:

* Clinical physiological monitoring
* Blood [oxygen saturation](https://www.news-medical.net/health/What-is-Oxygen-Saturation.aspx)
* Blood pressure
* Cardiac output
* Heart rate
* Respiration
* Vascular assessment
* Arterial disease
* Arterial compliance and ageing
* Venous assessment
* Endothelial function
* Microvascular blood flow
* Vasospastic conditions
* Autonomic function monitoring
* Vasomotor function and thermoregulation
* Blood pressure and heart rate variability
* Orthostasis
* Other cardiovascular variability assessments

**Wearable Devices**

Using this technology, wearable pulse rate monitors have been developed. These low-cost and small devices have high-intensity green light-emitting diodes (LEDs) and photodetectors that help reliable monitoring of the pulse rate in a non-invasive manner.

Important design requirements for these systems include miniaturization, robustness and user-friendliness.

These devices have a sensor that monitors minor variations in the intensity of light transmitted through or reflected from the tissue. These intensity changes are associated with changes in blood flow through the tissue and provide vital cardiovascular information such as the pulse rate.

**Other Systems**

PPG has been used in other technologies such as [telemedicine](https://www.news-medical.net/health/What-is-Telemedicine.aspx), PPG imaging technology and remote monitoring.

Researchers have studied skin blood flow and the related rhythms using a near infrared CCD PPG imaging system. This study aimed at obtaining fresh insights into biological tissue perfusion and studying changes associated with wound healing and formation of ulcers.

Other studies have applied PPG in remote imaging of the distribution of arterial oxygen saturation (SpO2) within a tissue. Such an image might be valuable in medical diagnostics studies such as quantification of tissue viability. In telemedicine, PPG is promising in remote monitoring of patients’ health.

# Trends in EEG signal feature extraction applications

# This paper will focus on electroencephalogram (EEG) signal analysis with an emphasis on common feature extraction techniques mentioned in the research literature, as well as a variety of applications that this can be applied to. In this review, we cover single and multi-dimensional EEG signal processing and feature extraction techniques in the time domain, frequency domain, decomposition domain, time-frequency domain, and spatial domain. We also provide pseudocode for the methods discussed so that they can be replicated by practitioners and researchers in their specific areas of biomedical work. Furthermore, we discuss artificial intelligence applications such as assistive technology, neurological disease classification, brain-computer interface systems, as well as their machine learning integration counterparts, to complete the overall pipeline design for EEG signal analysis. Finally, we discuss future work that can be innovated in the feature extraction domain for EEG signal analysis.

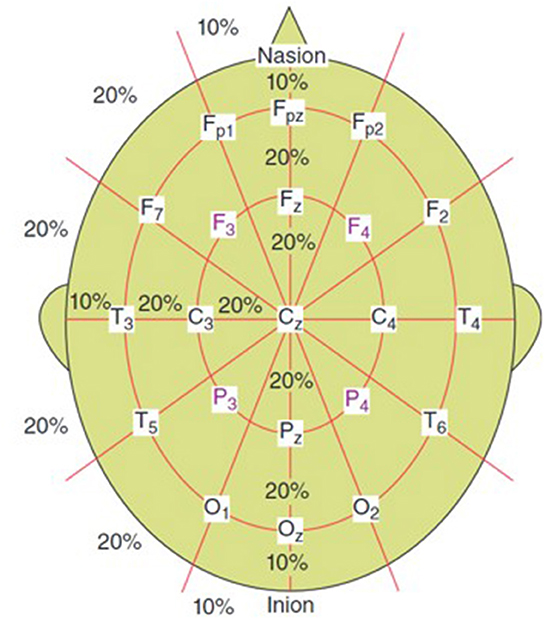
## 1. Introduction

Electroencephalogram (EEG) signals play an important role in understanding the electrical activity associated with brain functioning and brain-related disorders. A typical EEG signal analysis pipeline is as follows: (1) data acquisition, (2) data pre-processing, (3) feature extraction, (4) feature selection, (5) model training and classification, and (6) performance evaluation. Signal analysis, when applied to the EEG, is of particular interest as the entire body's condition, as well as brain status can often be recognized when digital signal processing (DSP) and machine learning (ML) methods are applied ([Sanei and Chambers, 2021](https://www.frontiersin.org/articles/10.3389/frai.2022.1072801/full" \l "B28)).

Carlo Matteucci and Emil Du Bois-Reymond were the first individuals to establish neurophysiology, and were the first to record and display brain activity. Later, Hans Berger discovered alpha wave activity in the brain, and he was the first to use scalp electrodes to record brain activity in the form of electrical signals in the 1870s. Berger was ultimately credited with inventing and measuring the EEG signal. Kornmüller, through his research, focused on multichannel recordings, their importance, and did so by widening the brain region covered by using a higher degree of electrodes. Since its discovery, EEG analysis has brought about significant advancements in studies of diagnosis and treatment of various neurological brain conditions and the overall health of the central nervous system (CNS). It can also be used to drive home-based technologies (telehealth), prosthetics and even in the world of virtual reality and gaming ([Sanei and Chambers, 2021](https://www.frontiersin.org/articles/10.3389/frai.2022.1072801/full" \l "B28)).

EEG systems used for signal acquisition consist of electrodes, differential amplifiers, filters and pen-type registers. A 10–20 EEG electrode placement method is commonly used (refer to [Figure 1](https://www.frontiersin.org/articles/10.3389/frai.2022.1072801/full#F1)). EEG signals are also sampled, quantized and encoded to convert them to digital form. Since the effective bandwidth of EEG signals is ~100 Hz, a minimum frequency of 200 Hz (to satisfy Nyquist criterion) is typically enough to sample the EEG for most applications ([Sanei and Chambers, 2021](https://www.frontiersin.org/articles/10.3389/frai.2022.1072801/full" \l "B28)).

Figure 1

[](https://www.frontiersin.org/files/Articles/1072801/frai-05-1072801-HTML/image_m/frai-05-1072801-g001.jpg)

**Figure 1**. 10–20 electrode setup for EEG ([Sanei and Chambers, 2021](https://www.frontiersin.org/articles/10.3389/frai.2022.1072801/full" \l "B28)).

### 1.1. Challenges in EEG analysis and applications

There are many applications that EEG signal analysis allows for; anywhere from disease diagnosis to even brain-computer interfaces (BCIs). A popular disorder studied heavily through EEG signal analysis is epilepsy.

Epilepsy is characterized by frequent seizures and is classified as a chronic neurological disorder. The EEG is used to identify the onset of seizures as well as for the diagnosis of epilepsy, however, this process is long and manual. Due to the manual nature, it is also subjective and thus can lead to very different diagnoses from various epileptologists. This has led to innovations in the technological realm to develop automated methods of seizure detection ([Bourien et al., 2021](https://www.frontiersin.org/articles/10.3389/frai.2022.1072801/full" \l "B5)).

EEG signal analysis is also being applied to the BCI domain, which is a rapidly growing field of research; it is an interesting field because it allows for a communication bridge between the external world and the human brain. It has been applied to assistive devices which have been used to restore movement to patients, as well as retraining patients to regain motor functionality. BCI systems function by analyzing the incoming brain waves from the EEG and converting the signal into appropriate action. There are, however, many challenges in this domain in terms of usability, training, information transfer rate, as well as technical challenges ([Abdulkader et al., 2015](https://www.frontiersin.org/articles/10.3389/frai.2022.1072801/full" \l "B1)).

Other applications of EEG include, but are not limited to, motor imagery classification, emotional classification, drug effects diagnosis, mental task diagnosis, and sleep state classification. Since large numbers of EEG channels are collected during data acquisition for these applications, there is a need for channel redundancy. There are algorithms that have been developed to assist with the channel selection of EEG signals. Channel selection assists with the reduction of computational complexity, reduce overfitting from redundant channels to improve performance, and reduce setup time in some applications. Some channel selection techniques are as follows: (1) filtering methods in which evaluation criteria are used to “filter” channels, (2) wrapping methods in which a classification algorithm is used, (3) embedded methods that select channels based on criteria generated during the learning process of classifiers, and (4) hybrid methods which combine filtering and wrapper techniques ([Alotaiby et al., 2015](https://www.frontiersin.org/articles/10.3389/frai.2022.1072801/full" \l "B2)).

# Speech signal processing

### 1.) INTRODUCTION

Speech signal processingis the study ofspeechsignalsand the processing methods of these signals. The can be audio, image, control, electrocardiogram signals, etc.The signals are usually processed in adigitalrepresentation, so that the speech processing can be regarded as a special case ofdigital signal processing, applied to speech signal. They are also very close tothe natural language processing(NLP), as its input can come from / output can go to NLP applications. There is an example like text to speech signal which use an information extraction techniques.It refers to the acquisition, manipulation, storage, transfer and output of vocal utterances by a computer. The main applications of speech signal processing are:

### 1.1) Speech recognition

### 1.2) Speech synthesis

### 1.3) Speech compression

The speech signal processing is the combination of the speech processing and the signal processing.

Speech processing is just the study of the signals like audio, image, etc. and then these signals are being processed in the form of digital representation. It is divided into the following five categories: speech coding, speech recognition, voice analysis, speech synthesis and speech enhacement.

Signal processing is an area ofelectrical engineeringandapplied mathematicsthat deals with operations on or analysis of signals, in either discrete or continuous time to perform useful operations on those signals. Depending upon the application, a useful operation could be filtering, spectral analysis,data compression,data transmission, denoising, prediction, smoothing, deblurring, tomographic reconstruction, identification, classification, or a variety of other operations. Signals of interest can includesound,images, time-varying measurement values andsensordata, for example the biological data such as the electrocardiogram signals,the control systemsignals, telecommunicationtransmissionsignals such as radio signals, and many others.Signalsare analog or digital electrical representations of time-varying or spatial-varying physical quantities. In the context of signal processing, arbitrary binary data streams and on-off signals are not considered as signals, but only analog and digital signals that are representations of analog physical quantities. The signal processing is categorised into three types which are: audio signal processing, discrete time processing and the digital signal processing.

### 1.1) SPEECH RECOGNITION

It is also called voice recognition which focuses on capturing the human voice as a digital sond wave and converting it into the format which can be read by computer. It is also known as automatic speech recognition or the computer speech rscognition. It converts the voice of human being into the machine readable input like computers. The term "voice recognition" is sometimes used to refer to speech recognition where the recognition system is trained to a particular speaker - as is the case for most desktop recognition software, hence there is an aspect ofspeaker recognition, which attempts to identify the person speaking, to better recognise what is being said. Speech recognition is a broad term which means it can recognise almost anybodys speech - such as a callcentre system designed to recognise many voices. Voice recognition is a system trained to a particular user, where it recognises their speech based on their unique vocal sound.

### The applications of speech recognition are as follows:

### a.) Health care:

In the health care, voice recognition technologies are sidely used. Speech recognition can be implemented in front-end or back-end of the medical documentation process. Front-End SR is where the provider dictates into a speech-recognition engine, the recognized words are displayed right after they are spoken, and the dictator is responsible for editing and signing off on the document. It never goes through an MT/editor. Back-End SR or Deferred SR is where the provider dictates into a digital dictation system, and the voice is routed through a speech-recognition machine and the recognized draft document is routed along with the original voice file to the MT/editor, who edits the draft and finalizes the report. Deferred SR is being widely used in the industry currently. ManyElectronic Medical Records(EMR) applications can be more effective and may be performed more easily when deployed in conjunction with a speech-recognition engine. Searches, queries, and form filling may all be faster to perform by voice than by using a keyboard.

### b.) Military:

Substantial efforts have been devoted in the last decade to the test and evaluation of speech recognition in fighter aircraft. Of particular note are the U.S. program in speech recognition for the Advanced Fighter Technology Integration (AFTI)/F-16aircraft (F-16 VISTA), the program in France on installing speech recognition systems onMirageaircraft, and programs in the UK dealing with a variety of aircraft platforms. In these programs, speech recognizers have been operated successfully in fighter aircraft with applications including: setting radio frequencies, commanding an autopilot system, setting steer-point coordinates and weapons release parameters, and controlling flight displays. Generally, only very limited, constrained vocabularies have been used successfully, and a major effort has been devoted to integration of the speech recognizer with the avionics system.

### Some important conclusions from the work were as follows:

\* Speech recognition has definite potential for reducing pilot workload, but this potential was not realized consistently.

\* Achievement of very high recognition accuracy (95% or more) was the most critical factor for making the speech recognition system useful— with lower recognition rates, pilots would not use the system.

\* More natural vocabulary and grammar, and shorter training times would be useful, but only if very high recognition rates could be maintained.

Laboratory research in robust speech recognition for military environments has produced promising results which, if extendable to the cockpit, should improve the utility of speech recognition in high-performance aircraft.

Working with Swedish pilots flying in theJAS-39Gripen cockpit, Englund (2004) found recognition deteriorated with increasing G-loads. It was also concluded that adaptation greatly improved the results in all cases and introducing models for breathing was shown to improve recognition scores significantly. Contrary to what might be expected, no effects of the broken English of the speakers were found. It was evident that spontaneous speech caused problems for the recognizer, as could be expected. A restricted vocabulary, and above all, a proper syntax, could thus be expected to improve recognition accuracy substantially.

TheEurofighter Typhooncurrently in service with the UKRAFemploys a speaker-dependent system, i.e. it requires each pilot to create a template. The system is not used for any safety critical or weapon critical tasks, such as weapon release or lowering of the undercarriage, but is used for a wide range of othercockpitfunctions. Voice commands are confirmed by visual and/or aural feedback. The system is seen as a major design feature in the reduction of pilotworkload, and even allows the pilot to assign targets to himself with two simple voice commands or to any of his wingmen with only five commands.

### 1.2) SPEECH SYNTHESIS

The artificial production of human speech is called the speech synthesis. A computer system used for this purpose is called aspeech synthesizer, and can be implemented insoftwareorhardware. It is the reverse process of speech recognition and advances in the area to improve the computer's usability for the visually impaired. Atext-to-speech (TTS)system converts normal language text into speech, other systems rendersymbolic linguistic representationslikephonetic transcriptionsinto speech.

Synthesized speech can be created by concatenating pieces of recorded speech that are stored in adatabase. Systems differ in the size of the stored speech units; a system that storesphonesordiphonesprovides the largest output range, but may lack clarity. For specific usage domains, the storage of entire words or sentences allows for high-quality output. Alternatively, a synthesizer can incorporate a model of thevocal tractand other human voice characteristics to create a completely "synthetic" voice output.

The quality of a speech synthesizer is judged by its similarity to the human voice and by its ability to be understood. An intelligible text-to-speech program allows people withvisual impairmentsorreading disabilitiesto listen to written works on a home computer. Many computer operating systems have included speech synthesizers since the early 1980s.

A text to speech system (TTS) is explained below:

### Overview of a typical TTS system

A text-to-speech system is composed of two parts: afront-endand aback-end. The front-end has two major tasks. First, it converts raw text containing symbols like numbers and abbreviations into the equivalent of written-out words. This process is often calledtext normalization. The front-end then assignsphonetic transcriptionsto each word, and divides and marks the text intoprosodic units, likephrases,clauses, andsentences. The process of assigning phonetic transcriptions to words is calledtext-to-speech. Phonetic transcriptions and prosody information together make up the symbolic linguistic representation that is output by the front-end. The back-end often referred to as thesynthesizer then converts the symbolic linguistic representation into sound.

### The application of speech synthesis are:

### a.) Accessibility:

Speech synthesis have a technology tool and its application is widely spread in some areas. It allows environmental barriers to be removed for people with a wide range of disabilities. The longest application has been in the use ofscreenreadersfor people withvisual impairment, but text-to-speech systems are now commonly used by people withdyslexiaand other reading difficulties as well as by pre-literate youngsters. They are also frequently employed to aid those with severespeech impairmentusually through a dedicatedvoice output communication aid.

### b.) Entertainment:

Speech synthesis techniques are also widely used as entertainment such as games. In 2007, Animo Limited announced the development of a software application package based on its speech synthesis software FineSpeech, explicitly geared towards customers in the entertainment industries, able to generate narration and lines of dialogue according to user specifications. The application reached maturity in 2008, when NECBiglobeannounced a web service that allows users to create phrases from the voices ofCode Geass: Lelouch of the Rebellion R2characters. Software such asVocaloidcan generate singing voices via lyrics and melody. This is also the aim of the Singing Computer project (which uses theGPLsoftwareLilypondandFestival) to help blind people check their lyric input.

### 1.3) SPEECH COMPRESSION

It is very important in the telecommunications area for increasing the amount of information which can be transferred, stored, or heard, for a given set of time and space constraints. The compression of speech signals has many practical applications. One example is in digital cellular technology where many users share the same frequency bandwidth. Compression allows more users to share the system than otherwise possible. Another example is in digital voice storage (e.g. answering machines). For a given memory size, compression allows longer messages to be stored than otherwise.

In the history, the digital speech signals are sampled at a rate of 8000 samples/sec. Each of the sample is represented by 8 bits (using mu-law). This corresponds to an uncompressed rate of 64 kbps (kbits/sec). With current compression techniques (all of which are lossy), it is possible to reduce the rate to 8 kbps with almost no perceptible loss in quality. Further compression is possible at a cost of lower quality. All of the current low-rate speech coders are based on the principle oflinear predictive coding (LPC)which is presented in the following sections.

Speech compression may also means the two different things i.e. speech coding and time compressed speech.

Now, there is an example that how we speak and how speech comes out from our mouth.

Physical Model of Speech Production

When we speak:

Air is pushed from your lung through your vocal tract and out of your mouth comes speech.

a. For certainvoicedsound, your vocal cords vibrate (open and close). The rate at which the vocal cords vibrate determines thepitchof your voice. Women and young children tend to have high pitch (fast vibration) while adult males tend to have low pitch (slow vibration).

b. For certainfricatives and plosive (or unvoiced)sound, your vocal cords do not vibrate but remain constantly opened.

c. The shape of your vocal tract determines the sound that you make.

d. As you speak, your vocal tract changes its shape producing different sound.

e. The shape of the vocal tract changes relatively slowly (on the scale of 10 msec to 100 msec).

f. The amount of air coming from your lung determines the loudness of your voice.

### CONCLUSION

Speech signal processingis the study ofspeechsignalsand the processing methods of these signals. The can be audio, image, control, electrocardiogram signals, etc.The signals are usually processed in adigitalrepresentation. The speech signal processing is the combination of the speech processing and the signal processing. The main applications of speech signal processing are: Speech recognition, Speech synthesis and Speech compression. The artificial production of human speech is called the speech synthesis. Speech processing is just the study of the signals like audio, image, etc. and then these signals are being processed in the form of digital representation.

**Speech Processing:** Speech signals are processed in various ways, such as speech recognition, speaker identification, emotion analysis, and more. Signal processing techniques like filtering, spectrogram analysis, and machine learning are used to extract meaningful information from speech signals.

Speech signal processing has a wide range of applications across various fields due to its importance in communication, human-computer interaction, and information extraction. Here are some key areas where speech signal processing is applied:

1. **Automatic Speech Recognition (ASR):** ASR technology converts spoken language into written text. It's used in voice assistants (e.g., Siri, Alexa), transcription services, language translation, and more.
2. **Speaker Identification and Verification:** This involves recognizing and verifying individuals based on their unique speech patterns. It's used in security systems, access control, and authentication processes.
3. **Emotion Recognition:** Speech signals carry emotional cues like tone, pitch, and prosody. Emotion recognition systems can detect emotions from speech, which finds applications in customer service, mental health monitoring, and entertainment.
4. **Speech Synthesis:** Also known as text-to-speech (TTS), this technology converts text into natural-sounding speech. It's used in navigation systems, audiobooks, assistive devices for the visually impaired, and more.
5. **Voice Conversion:** This technology can change a speaker's voice characteristics while preserving the linguistic content. It's used in entertainment (altering actor voices), personalization, and voice impersonation.
6. **Speech Enhancement:** Techniques for noise reduction and enhancement improve speech quality in noisy environments, such as in telephony, hearing aids, and voice recordings.
7. **Language and Dialect Identification:** Speech signals can help identify languages and even specific dialects, which is useful in language recognition systems and linguistic studies.
8. **Forensic Analysis:** Speech signal analysis can be used in forensic investigations to determine the source of anonymous threats, verify audio evidence, and analyze voice disguises.
9. **Healthcare Applications:** Speech signals can be used for early detection of neurological disorders and diseases like Parkinson's, Alzheimer's, and depression by analyzing changes in speech patterns.
10. **Educational Technology:** Speech processing can aid in language learning by providing feedback on pronunciation and intonation. It's also used in systems that assist individuals with speech and language disorders.
11. **Market Research and Sentiment Analysis:** Speech data can be analyzed to gauge public opinion, emotions, and trends in real-time, benefiting businesses and policymakers.
12. **Human-Machine Interaction:** Speech-enabled interfaces provide natural and intuitive interaction with devices. This is evident in virtual assistants, voice-controlled appliances, and interactive kiosks.
13. **Accessibility:** Speech processing helps make technology more accessible for individuals with disabilities, such as those who have difficulty typing or reading.
14. **Entertainment and Gaming:** Speech recognition and synthesis enhance gaming experiences by allowing players to control characters and interact with virtual worlds using voice commands.
15. **Security and Surveillance:** Speech analysis can aid in monitoring public spaces by detecting abnormal or suspicious behavior through voice patterns and content analysis.
16. **Customer Service:** Many call centers and customer service centers employ speech recognition to route calls, provide automated assistance, and improve customer experience.