

Digital Signal Processors

Assignment

Module Code =

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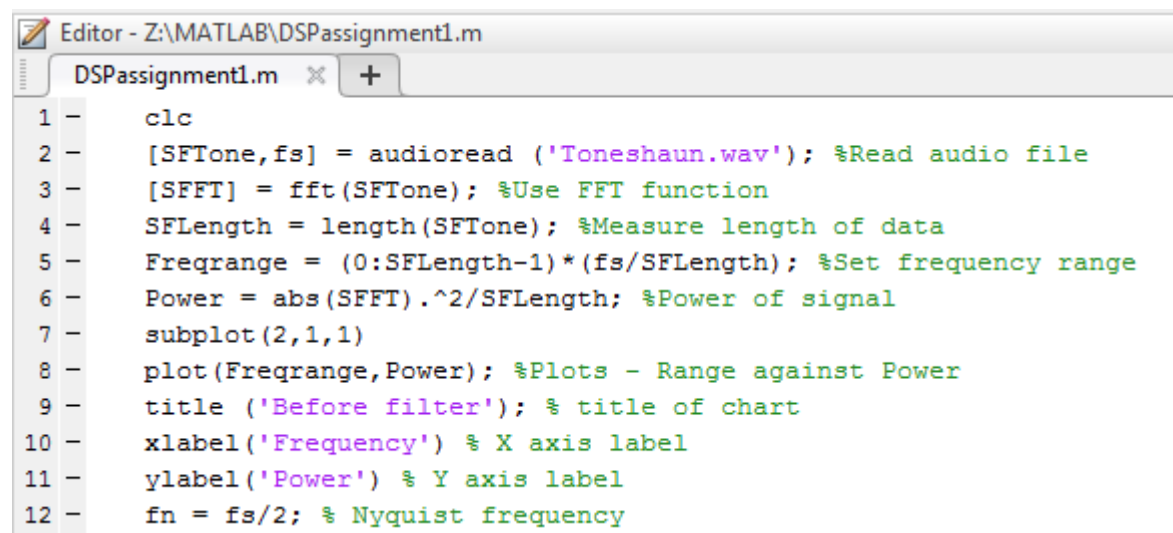
Assignment Task 1

For this task, I was given an audio file called "Tonshaun.wav". The passage of speech read, "I'll be back" and it also contained a corrupted audio tone.

Determining the corrupting frequency

The first part of the task was to determine at what frequency the corrupt tone was.

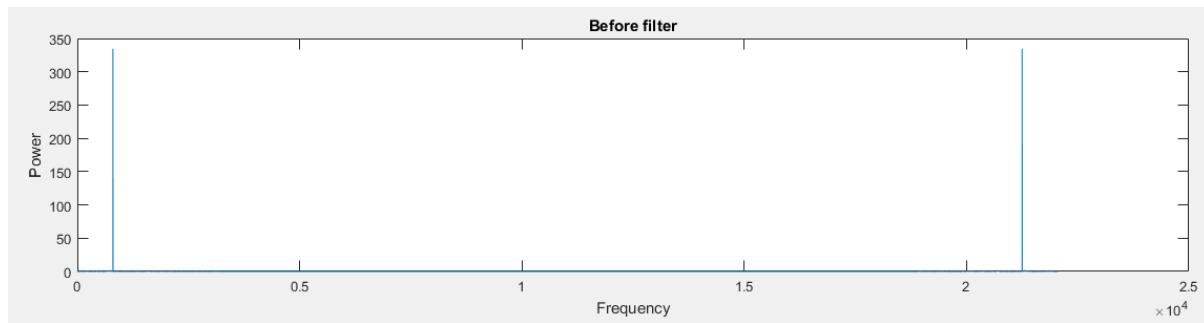
I first read the audio file in the matlab programme. This is shown in line 2 of the image below, this command reads data from the file named in the description and returns the sampled data.



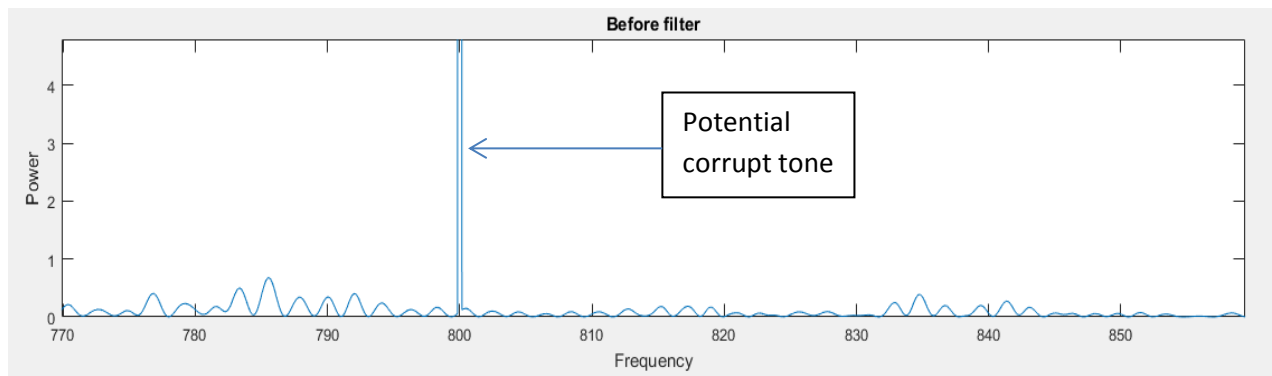
```
Editor - Z:\MATLAB\DSPassignment1.m
DSPassignment1.m x +
1 -   clc
2 -   [SFTone,fs] = audioread ('Tonshaun.wav'); %Read audio file
3 -   [SFFT] = fft(SFTone); %Use FFT function
4 -   SFLength = length(SFTone); %Measure length of data
5 -   Fregrange = (0:SFLength-1)*(fs/SFLength); %Set frequency range
6 -   Power = abs(SFFT).^2/SFLength; %Power of signal
7 -   subplot(2,1,1)
8 -   plot(Fregrange,Power); %Plots - Range against Power
9 -   title ('Before filter'); % title of chart
10 -  xlabel('Frequency') % X axis label
11 -  ylabel('Power') % Y axis label
12 -  fn = fs/2; % Nyquist frequency
```

The next part of the code in line 3 computes the discrete Fourier transform using the fast Fourier transform (FFT) algorithm, this allows me to view the audio in the frequency domain. I have then measured the length of the data in line 4, this is important to ensure that each bin for power/frequency of sampled data is captured. In line 5, I have recorded the range of frequencies in the sampled data. Line 6 records the power of each of the frequencies sampled. Lines 7, 8, 9, 10 and 11 contribute to plotting a chart showing the frequency on the X axis against power on the Y axis. Line 12 shows the sample frequency, this is twice the highest frequency, known as the Nyquist frequency. This should be twice the highest frequency so that no aliasing occurs, potentially introducing errors. (The Math works)

The results I received from reading the audio file are plotted below.



I have then focussed in on the file, this is shown below.



The image above shows where I think the wave form of the corrupt audio tone is; at 800Hz.

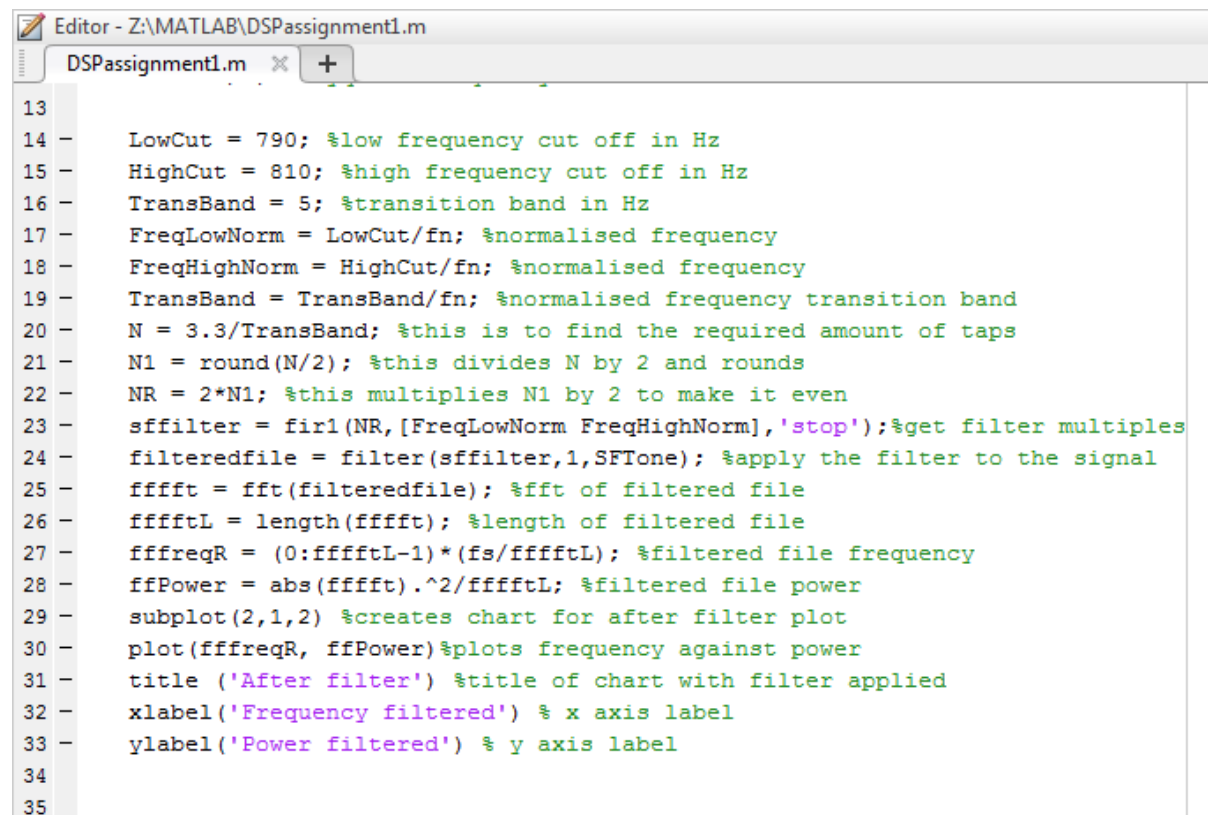
To determine that this is the corrupt tone, I needed to design a filter.

Filter

There are two primary types of filters used in DSP's, these are 'FIR' filters, Finite impulse response filters and 'IIR' filters, Infinite impulse response filters. Following some research, I decided to design a FIR filter. The FIR filter can easily be designed to be linear phase, which slightly delays the input signal but doesn't distort its phase. They are suited to multi rate applications, decimation" (reducing the sampling rate), "interpolation" (increasing the sampling rate), or both. Whether decimating or interpolating, the use of FIR filters allows some of the calculations to be omitted, thus providing an important computational efficiency. (Iowegian International, n.d.)

Matlab filter

The image below shows the matlab code for the filter design and plotting of the chart after the filter has been implemented.



```

Editor - Z:\MATLAB\DSPassignment1.m
DSPassignment1.m x +
13
14 - LowCut = 790; %low frequency cut off in Hz
15 - HighCut = 810; %high frequency cut off in Hz
16 - TransBand = 5; %transition band in Hz
17 - FreqLowNorm = LowCut/fn; %normalised frequency
18 - FreqHighNorm = HighCut/fn; %normalised frequency
19 - TransBand = TransBand/fn; %normalised frequency transition band
20 - N = 3.3/TransBand; %this is to find the required amount of taps
21 - N1 = round(N/2); %this divides N by 2 and rounds
22 - NR = 2*N1; %this multiplies N1 by 2 to make it even
23 - sffilter = fir1(NR,[FreqLowNorm FreqHighNorm],'stop'); %get filter multiples
24 - filteredfile = filter(sffilter,1,SFTone); %apply the filter to the signal
25 - fffft = fft(filteredfile); %fft of filtered file
26 - fffftL = length(fffft); %length of filtered file
27 - fffreqR = (0:fffftL-1)*(fs/fffftL); %filtered file frequency
28 - ffPower = abs(fffft).^2/fffftL; %filtered file power
29 - subplot(2,1,2) %creates chart for after filter plot
30 - plot(fffreqR, ffPower) %plots frequency against power
31 - title ('After filter') %title of chart with filter applied
32 - xlabel('Frequency filtered') % x axis label
33 - ylabel('Power filtered') % y axis label
34
35

```

Line 14 shows the frequency cut off in hertz for the low cut off. Line 15 shows the high cut off frequency in hertz. Line 16 shows the transition band, the transition band is the band of frequencies between passband and stopband edges. The narrower the transition band, the more taps are required to implement the filter. (A “small” transition band results in a “sharp” filter.) Lines 17, 18 and 19 normalise the frequencies, this command changes the signal from a continuous time variable, t , with units of seconds and is replaced by the discrete integer variable, n , with units of samples. Line 20 calculates the number of taps, the number of FIR taps, (often designated as “ N ”) is an indication of 1) the amount of memory required to implement the filter, 2) the number of calculations required, and 3) the amount of “filtering” the filter can do; in effect, more taps means more stopband attenuation, less ripple and narrower filters. (lowegian International, n.d.).

Window Type	Normalized Transition Width	Passband Ripple (dB)	Stopband Attenuation (dB)
Rectangular	$0.9/N$	0.7416	21
Hanning	$3.1/N$	0.0546	44
Hamming	$3.3/N$	0.0194	53
Blackman	$5.5/N$	0.0017	74

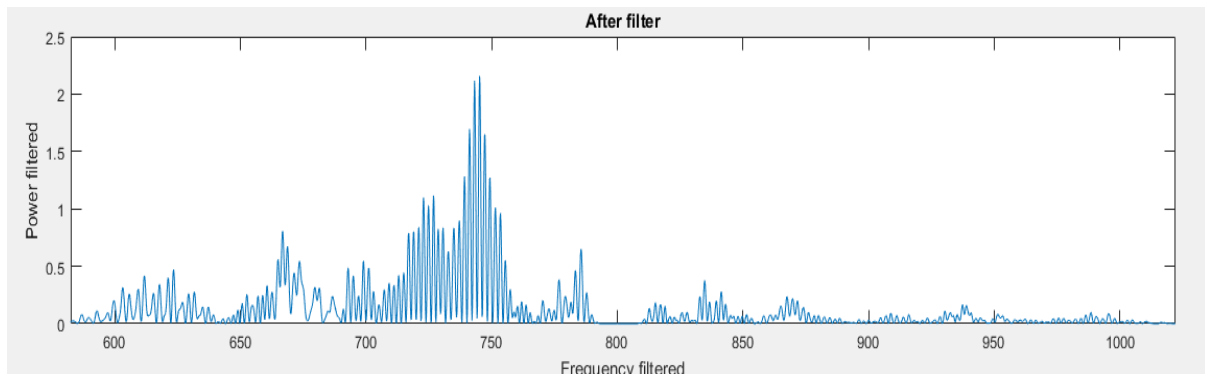
Examples of window functions and their corresponding properties are shown in the image above.

(dsp-u-lec08-fir-filter-design, 2010)

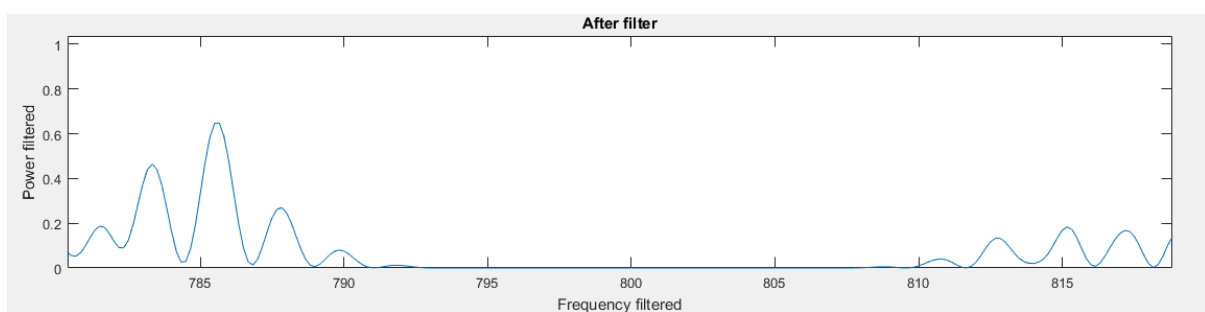
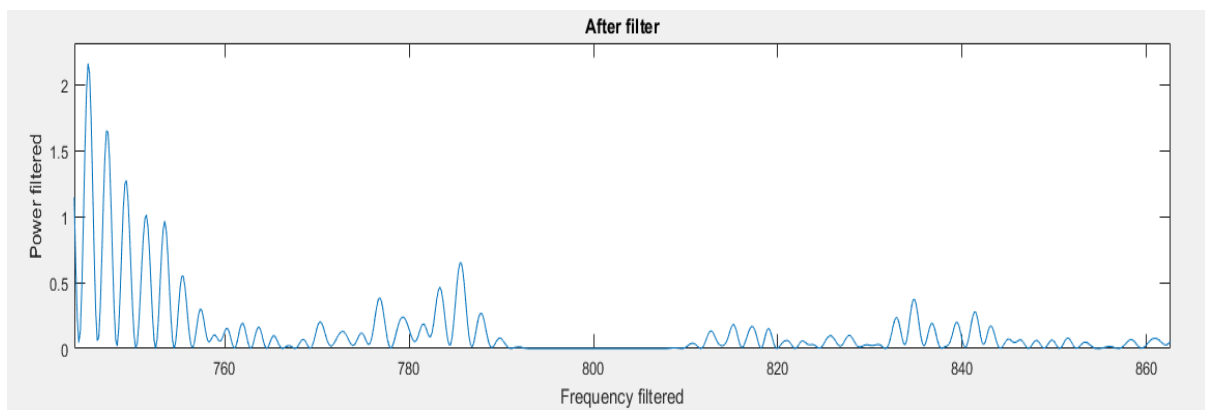
Line 21 divides the number of taps by 2 and then rounds to the nearest whole number. Line 22 multiplies the number of taps in N1 by 2 to make it an even number of taps. Line 23 implements the windowed, linear phase FIR filter, this is hamming window based. Line 24 applies the filter to the data sequence. Line 25 applies the fast Fourier transform (FFT) of the data, this allows us to view the signals in the frequency domain. Line 26 returns the length of the longest dimension of the data file. Line 27 shows the frequency range of the filtered file. Line 28 shows the power of the filtered file. Lines 29 to 33 are associated with plotting a chart showing the filtered frequency (X axis) against the filtered power (Y axis) and their associated axis titles.

Filtered data

The data after the filter has been applied is plotted below.

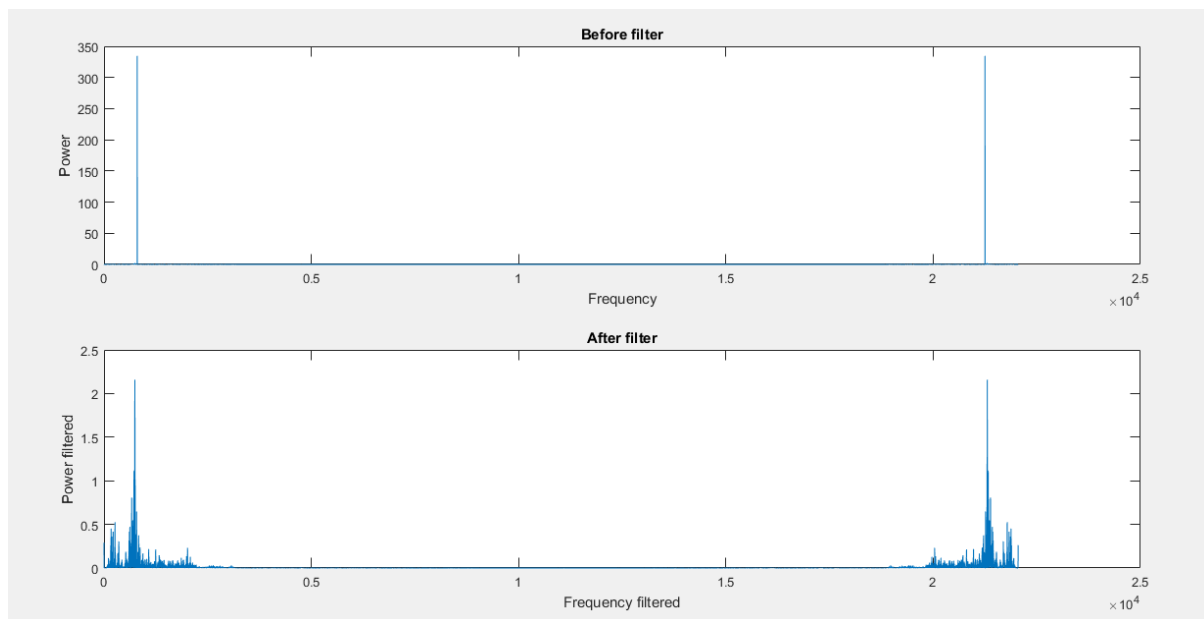


The image above shows that where the filter has been applied, between 790 and 810 Hz, the data has been filtered out. When the audio file is played after the filter has been applied, the corrupting tone can no longer be heard.

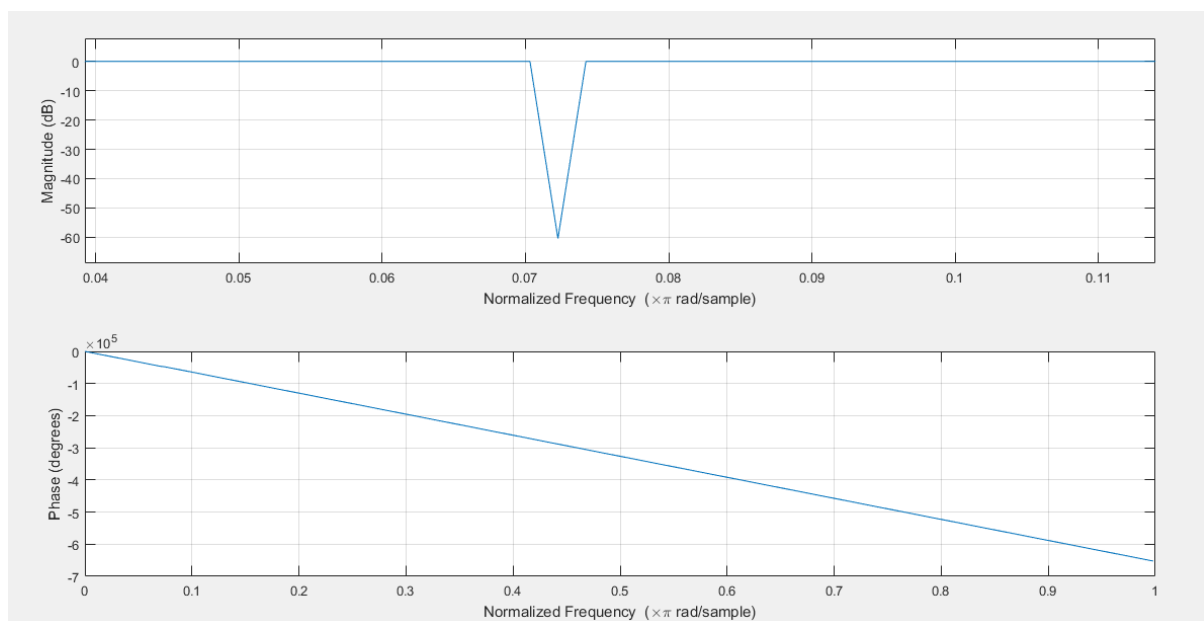


The two images above are a closer snap shot of the frequency of the suspected corrupt tone. The image clearly shows where the signal is filtered between 790 to 810 Hz.

The image below shows a comparison of the data before and after the FIR filter has been applied.



The final image below shows the filter response in the top image. It shows an attenuation of 60dB.



The bottom image above shows the phase response of the filter. This shows a linear response as we would expect with a FIR filter. I feel that the filter was effective as the tone could not been heard when playing the audio file. Also, the filter response and filtered chart plots show what we would expect the chart to show. The corrupt tone at 800Hz was removed and the speech passage was not affected

Assignment Task 2

Background

The evolution of mobile communications places an ever increasing demand on the capacity of mobile networks. The networks have evolved from the 1970's and 1980's when they were first introduced. The radio signals used by 1G were analogue, meaning the voice of a call is modulated to a higher frequency rather than being encoded to digital signals. Analogue signals degrade over time and space meaning that voice data can very often lack quality within a call.

In comparison, digital is a representation of analogue stored as signals, meaning larger amounts of data can be carried more effectively. The second generation saw the introduction of GSM (Global System for Mobile Communication) technologies as a standard in the early 90's. It allowed for digital voice and data to be sent across the network and allowed users to roam for the first time. 2G also used Signalling and Data Confidentiality and Mobile Station Authentication to ensure improved security and privacy of telephone calls. The advance in technology from 1G to 2G introduced many of the fundamental services that we still use today, such as SMS, internal roaming, conference calls, call hold and billing based on services e.g. charges based on long distance calls and real time billing.

Between the year 2000 and 2003, an upgrade in technologies introduced the packet network which provided high speed data transfer and internet and became known as 2.5G. The standards included GPRS (General Packet Radio Service) and EDGE (enhanced Data Rates in GSM). GPRS supports flexible data transmission rates and provides continuous connection with the network. It also allows for the service provider to charge for the amount of data that is sent rather than their connection time.

The next generation saw the arrival of 3G. Introduced commercially in 2001, the goals set out for third generation mobile communication were to facilitate greater voice and data capacity, support a wider range of applications, and increase data transmission at a lower cost. For the first time, this generation supported high speed wide band internet access as well as fixed wireless internet access and allowed for video calls, chatting and conferencing, mobile TV, video on demand services, navigational maps, email, mobile gaming, music and digital services such as movies. Significantly greater security features were introduced within 3G, including Network Access and Domain Security and Application Security.

Initiated in 2010, the fourth generation is an all IP based network system. Its purpose is to provide high speed, high quality and high capacity to users while improving security and lower the cost of voice and data services, multimedia and internet over IP.

The major benefit of an IP based network is that it is able to seamlessly handover, for voice and data to GSM, UMTS and CDMA2000 technologies from the previous different generation's infrastructure. 4G introduced the LTE standard which only support packet switching and an all IP Network. There are a significant amount of infrastructure changes needed to be implemented by service providers in order to supply because voice calls in GSM, UMTS and CDMA2000 are circuit switched, so with the adoption of LTE, carriers will have to re-engineer their voice call network.

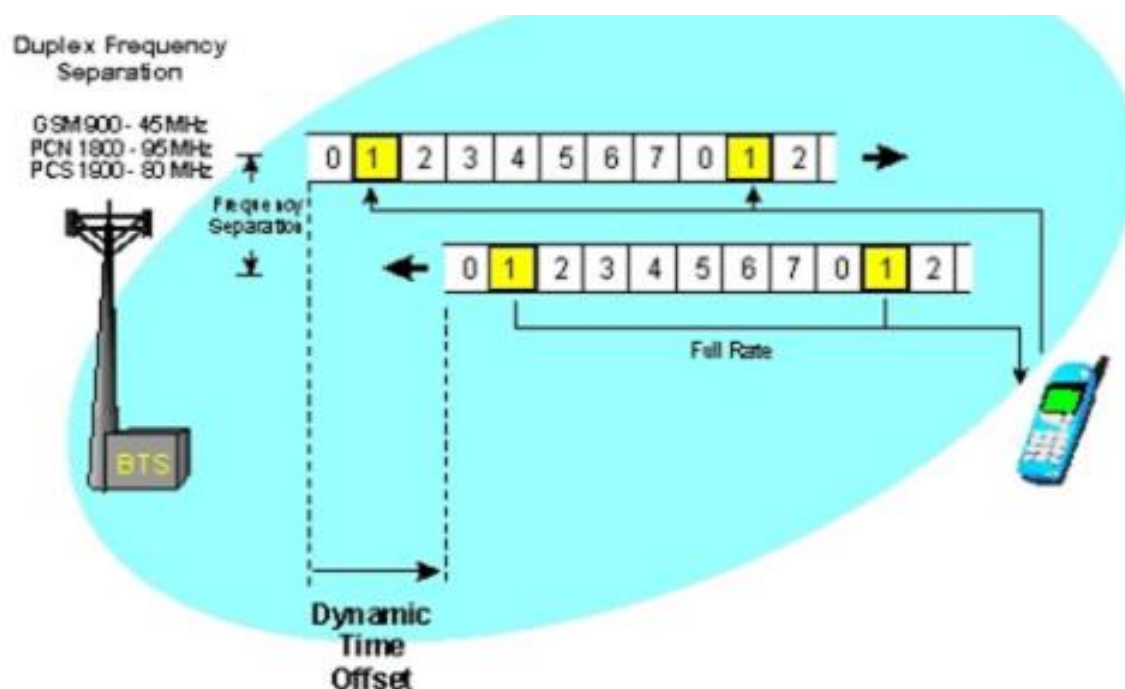
The next generation of commercial cellular network is set to greatly increase internet connectivity speeds. One of the main benefits of 5G is set to be IoT (Internet Of Things) which would make the most of the higher speed of connectivity to allow for seamless integration of devices on a scale never been achievable before. 5G would see data speed rates of between 1Gbps to 10Gbps. (evolution-mobile-communication-1g-5g, 2017)

Time Division Multiple Access

Time division multiple access (TDMA) is a process of sharing a single radio channel by dividing the channel into time slots that are shared between simultaneous users of the radio channel. When a mobile radio communicates with a TDMA system, it is assigned a specific time position on the radio channel. By allowing several users to use different time positions (time slots) on a single radio channel, TDMA systems increase their ability to serve multiple users with a limited number of radio channels. (GSM-tutorial-TDMA-time-division-multiple-access, 2009)

TDMA transmission can reduce the power on the transmitted signal to reduce the likelihood of error in the received signal. TDMA is a discontinuous transmission and it does not transmit silence. TDMA uses frequency hopping, it transmits at different frequencies every burst. TDMA uses sector antennas so there is a smaller area of transmission. TDMA transmits at 200KHz per channel.

TDMA allows a mobile station's radio component to listen and broadcast only in its assigned time slot. During the remaining time period, the mobile station may apply network measurements by detecting surrounding transmitters in different frequencies. This feature allows inter-frequency handover, which differs from code division multiple access (CDMA), where frequency handover is difficult to achieve. TDMA is used in most 2G cellular systems, while 3G systems are based on CDMA. (time-division-multiple-access-tdma, 2019)



Wide Band Code Division Multiple Access

Wide band code multiple access (WCDMA) and UMTS Universal Mobile Telecommunications System, is the 3G successor to the GSM family of standards including GPRS and EDGE. 3G UMTS uses a totally different radio interface based around the use of Direct Sequence Spread Spectrum as CDMA or Code Division Multiple Access. Although 3G UMTS uses a totally different radio access standard, the core network is the same as that used for GPRS and EDGE to carry separate circuit switched voice and packet data. UMTS uses a wideband version of CDMA occupying a 5 MHz wide channel. Being wider than its competition CDMA2000 which only used a 1.25MHz channel, the modulation scheme was known as wideband CDMA, or WCDMA / W-CDMA. This name was often used to refer to the whole system. In order to create and manage a system as complicated as UMTS or WCDMA it is necessary to develop and maintain a large number of documents and specifications. For UMTS or WCDMA, these are now managed by a group known as 3GPP - the Third Generation Partnership Programme. This is a global co-operation between six organisational partners - ARIB, CCSA, ETSI, ATIS, TTA and TTC.

The UMTS WCDMA system offered a significant improvement in capability over the previous 2G services.

3G UMTS SPECIFICATION SUMMARY

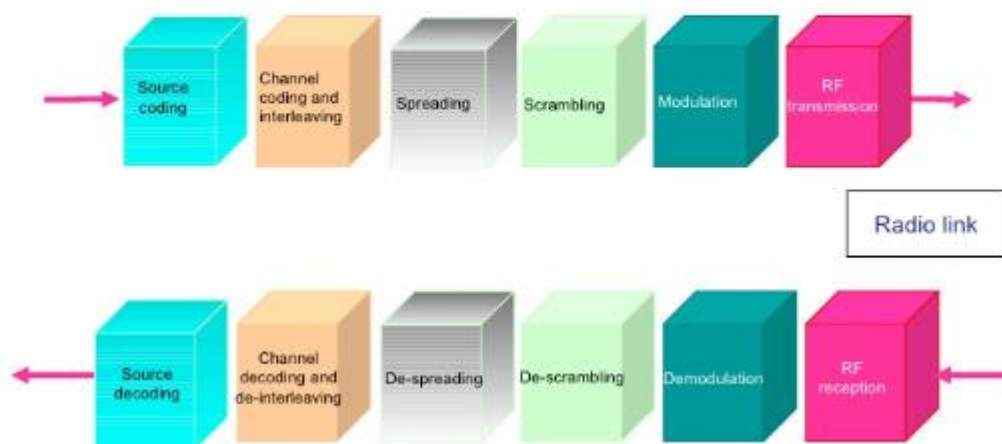
PARAMETER	SPECIFICATION
Maximum data rate	2048 kbps low range 384 kbps urban and outdoor
RF channel bandwidth	5 MHz
Multiple access scheme	CDMA
Duplex schemes	FDD and also TDD

UMTS uses Wideband CDMA - WCDMA - as the radio transmission standard. It employs a 5 MHz channel bandwidth. Using this bandwidth it has the capacity to carry over 100 simultaneous voice calls, or it is able to carry data at speeds up to 2 Mbps in its original format. However with the later enhancements of HSDPA (High-speed Downlink Packet Access) and HSUPA (High-speed Uplink Packet Access) included in later releases of the standard the data transmission speeds have been increased to 14.4 Mbps.

Many of the ideas that were incorporated into GSM have been carried over and enhanced for UMTS. Elements such as the SIM have been transformed into a far more powerful USIM (Universal SIM). In addition to this, the network has been designed so that the enhancements employed for GPRS and EDGE can be used for UMTS. In this way the investment required is kept to a minimum.

A new introduction for UMTS is that there are specifications that allow both Frequency Division Duplex (FDD) and Time Division Duplex (TDD) modes. The first modes to be employed are FDD modes where the uplink and downlink are on different frequencies. The spacing between them is 190 MHz for Band 1 networks being currently used and rolled out. The TDD mode where the uplink and downlink are split in time with the base stations and then the mobiles transmitting alternately on the same frequency is particularly suited to a variety of applications. (articles/connectivity/3g-umts/what-is-umts-wcdma-tutorial, 2019)

Basic Diagram of the WCDMA System



Frequency re-use

Frequency reuse is the process of using the same radio frequencies on radio transmitter sites within a geographic area, which are separated by sufficient distance to cause minimal interference with each other. Frequency reuse allows for a dramatic increase in the number of customers that can be served (capacity) within a geographic area on a limited amount of radio spectrum (limited number of radio channels).

Frequency planning is the assignment (coordination) of radio channel frequencies in wireless systems that have multiple transmitters to minimize the amount of interference caused by transmitters that operate on the same frequency. Frequency planning is used to help ensure that combined interference levels from nearby transmitters that are operating on or near the same frequency do not exceed a certain interference (desired signal to interference) level compared to the desired signal.

The ability to reuse frequencies depends on various factors, including the ability of channels to operate in with interference signal energy attenuation between the transmitters. A frequency plan is the assignment of radio frequencies to radio transmission sites (cell sites) that are located within a defined geographic area. The frequency plan may use ratios that are different dependent on the number of transmitting sites to the number of antennas (sectors) on each site. A common frequency reuse plan for GSM is the ability to reuse a radio frequency on every 4th site that has three 120 degree sectors each – 12 total sectors. This plan is commonly called “4/12”.



Image above shows frequencies with enough separation distance not to cause interference. (tutorial/GSM-tutorial-frequency-reuse, 2009)

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'M' file link



DSPassignment1.m