5G Notes

Data Compression

A Guide on Data Compression and 2G Stat sheet

Audio Signal Compression

> *As long as I get it, I get it.*

We have mentioned previously that the digital 2G signals were compressed before transmission to pack and transport more data than analog. This is done using a technique called **"Companding".**

It is a word made up of two different words **Compression and Expanding**. The aim is pretty simple. Reduce the size of data i.e. the amount of 0s and 1s required to represent the data in a way that it doesn't damage the information itself. It is not always accurate and most of the times we are trading size with noise which means that compressing the data might make it noisier than before.

International Telecom Union has recommended two Companding algorithms named **u(Mu)-law** and **A-law** Companding. Instead of worrying about which one is better, think of them as co-existing algorithms adapted in different parts of the world. For example, US and Japan used Mu-law while European countries used A-law and so on.

There are continuous and discrete versions of both but we will only focus on the discrete ones since they are useful for digital communication. Since all of the students may not be from electronics background, I have kept the focus on essence or crux of the logic behind the process.

With that out of the way, let's get started!

Human ears are smart enough to recognize high amplitude (loud) speech patterns even if they are noisier compared to rest of the sounds. Therefore, we can comprehend DJs, dance on party songs and recognize words from horn mics even if they are noisiest. On the other hand, low amplitude (less loud) speech patterns need to be clear. You might have asked someone to whisper again and again until their words were clear to you.

Putting it into technical perspective, our ears are good at removing noise from larger sound signals compared to smaller ones. This happens because speech recognition or sound decoding is a logarithmic process. We decode the voices in terms of curves.

This property or characteristic of human auditory system is used to make the signals smaller while transporting them and turning them back to larger ones after reception.

To make the signals smaller, Compression algorithms mimic the curves of the signal and map them onto smaller signals. Pretty much like how we change an HD image to a thumbnail without changing its aspect ratio. Once the signals have been transported (sent and received) a reverse algorithm of the compression (called expanding) enlarges those signals by mimicking the curves again.

As you might have guessed, this can cause errors.

While compressor was mimicking the original signal, it might have lost some of the small signal curves whereas when expanding algorithm mimics the compressed signal to enlarge it; it might miss some curves again.

At the end, the expanded signal will most likely not be EXACTLY SAME as the original signal. But that's the thing. As long as it is close to the original signal, our ears can recognize the speech! To make sure it stays close to the original signal, the algorithms play more with larger signals and put the smaller signals almost as they were. The larger signals an algorithm can compress the more dynamic range it is said to have.

As far as Mu-Law and A-Law are concerned, they are used with digital brother of the modulation schemes we have seen. In digital communication, we can't use FM (Frequency Modulation) or AM (Amplitude Modulation) so we have to use PCM which stands for Pulse Coded Modulation.

As for the differences, Mu and A law use different granularity to mimic curves and they are represented with little differences mathematically. But the bottom line is, both do their job fairly well and both are suitable for respective demographic factors like population and spectrum availability.

So, this is how digital data is compressed for mobile communication.

Happy Learning! :)