VIDEO 13.1. FUNDAMENTALS OF DIGITIZATION

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RAMESH YERRABALLI: Let's take a look at the concept of digitization.

JONATHAN VALVANO: The world we live in is inherently analog.

That means the signals are continuous, both in time and space.

However, computers are inherently digital.

So Professor Yerraballi, how do we capture analog signals

in a digital form?

RAMESH YERRABALLI: That is right.

So we have computers.

For us, it's a micro controller, which is inherently digital,

because it stores information in the form of binary data.

And we have the analog world, which conveys information

as a continuous time varying signal.

And we have to interface these two.

So our first kind of interfacing is how do we read information that's analog

and store it in digital form?

So what we need is a conversion, which

we call analog

to digital conversation, ADC, which does that act for us.

The second possibility is when we want to affect the world that is analog.

And the affectation is performed by, again, a microcontroller.

So we have the information in digital form.

And we want to convey it to the outside world, which is analog.

And we do that by introducing a module, which we will call a DAC.

A DAC is a digital to analog converter.

In some cases, the DAC could be part of the microcontroller.

But for flexibility, we will use external circuitry

that will do the job for us.

So let's take a look at some examples.

So when I look at examples of analog stimuli

that have to be captured in digital form,

I can think of simple things like temperature sensors.

JONATHAN VALVANO: Sound.

RAMESH YERRABALLI: Sound, which is where we're doing recording.

We're capturing sound.

And we can think of pressure sensors.

JONATHAN VALVANO: There are signals in the body, Bio-potentials,

you could measure.

RAMESH YERRABALLI: OK.

Biosensors-- we can think of proximity sensors.

JONATHAN VALVANO: Oh, I don't want my robot to hit the wall.

RAMESH YERRABALLI: Yeah.

And we have optical sensors.

Now, the world of embedded systems is all

about being able to work with sensors.

But we also have the counterpart, which is actuators.

So if we call these sensors, which is the input part,

then the output is actuators.

That is, we produce a stimuli which affects the world.

So again, we can think of examples like controlling our motors

so that we send voltages that power our motors.

JONATHAN VALVANO: Sound could be an output, too.

RAMESH YERRABALLI: Speakers.

We could have lights-- dimmers, for example.

What else?

JONATHAN VALVANO: Pacemaker uses an output to trigger the heart.

RAMESH YERRABALLI: OK.

JONATHAN VALVANO: So it's another biosensor, a bio actuator.

RAMESH YERRABALLI: Bio actuator.

And our anti-lock breaks are outputs.

And we want the braking to be controlled.

It's not an on off thing, but it's something that has to be a variable.

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So that's one.

JONATHAN VALVANO: Just like temperature was an input,

it could be an output-- so our heating and air conditioning.

RAMESH YERRABALLI: So A/C could be.

So we have examples of sensors and actuators.

So in summary, then, we have the analog world

interfacing with our digital computers.

And we have our ADCs and DACs that accomplish that.

So this module is going to be focusing on DACs.

JONATHAN VALVANO: What's next?

RAMESH YERRABALLI: So next goal is to understand

what the fundamentals of digitization are.

So we saw that the purpose of digitization

is to take an analog signal and digitize it,

which means to store it in a computer.

So we have an analog signal that's continuously varying in time.

And for now, we will think of it as a change in changing voltage.

And we want to digitize this.

So digitization is done using two concepts.

One is our amplitude quantization.

So by quantization, we mean discretizing the signal.

So we have continuous.

The opposite of continuous, as far as a computer is concerned, is discrete.

So when we store our information in the computer,

we store them as digital values.

So our amplitude quantization is talking about separating our amplitude

into levels.

So we'll call these levels.

So the counterpart to amplitude quantization

is to do time quantization, which is also referred to as sampling.

So time quantization is capturing the signal periodically,

or as we say, sampling the signal, so that we

can capture the essence of the signal faithfully.

So we have periodic capture of samples, which we call as time quantization.

So let's look at what the trade-offs are when we do amplitude quantization

and time quantization.

So the way we like to look at this is if I have more levels, from a computing

standpoint, I have to represent this by more bits per sample,

which means I have more levels.

If I have only two levels, I would represent all samples by just 0 or 1.

But more levels means more bits per sample.

From an analog standpoint, what this means is I have better precision.

So we'll define the term "precision."

The term "precision" is telling us exactly the number

of levels we can represent in a signal.

So I am assuming that I have around 16 levels here.

If I have 16 levels, I have four bits per sample.

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So the precision is 16.

Now, what I also have is this concept of resolution.

Resolution is the smallest change I can represent in the analog signal.

So resolution is the smallest change, which

means it's the difference between levels that I can actually capture.

So what we are saying is this captures precision.

Now, what we also see, then, is if we have more samples--

then the rate of capture is high-- then what we are saying is,

from a computer's standpoint, we need faster processing.

Whether we are producing these samples or we're captioning these samples,

we have to do more processing per unit time.

What it means from an analog standpoint is the more samples I capture,

the more faithful the digital representation

is with respect to the analog signal.

As a matter of fact, we will capture this idea

of faithfulness using a theorem called the Nyquist theorem, which

for us simply tells us how faithful is faithful enough?

That is, how many samples per second is best

to capture the essence of the analog signal?

So in summary, then, digitization is about taking a continuous time varying

signal and discretizing it so that we capture it

by these discrete points that attempt to make

a faithful representation of the signal.

An **analog signal** is one that is continuous in both amplitude and time. Neglecting quantum physics, most signals in the world exist as continuous functions of time in an analog fashion (e.g., voltage, current, position, angle, speed, force, pressure, temperature, and flow etc.) In other words, the signal has amplitude that can vary over time, but the value cannot instantaneously change. To represent a signal in the digital domain we must approximate it in two ways: amplitude quantizing and time quantizing (or Sampling). From an amplitude perspective, we will first place limits on the signal restricting it to exist between a minimum and maximum value (e.g., 0 to +3V), and second, we will divide this amplitude range into a finite set of discrete values. The **range** of the system is the maximum minus the minimum value. The **precision** of the system defines the number of values from which the amplitude of the digital signal is selected. Precision can be given in number of alternatives, binary bits, or decimal digits. The **resolution** is the smallest change in value that is significant. Furthermore, with respect to time one considers analog signals to exist from time equals minus infinity to plus infinity. Because memory is finite, when representing signals on a digital computer, we will restrict signal to a **finite time**, or we could have a finite set of data that are repeated over and over.

Figure 13.1 shows a temperature waveform (solid line), with a corresponding digital representation sampled at 1 Hz and stored as a 5-bit integer number with a range of 0 to 31 ^{o}C. Because it is digitized in both amplitude and time, the digital samples (individual dots) in Figure 13.1 must exist at an intersection of grey lines. Because it is a time-varying signal (mathematically, this is called a function), we have one amplitude for each time, but it is possible for there to be 0, 1, or more times for each amplitude.

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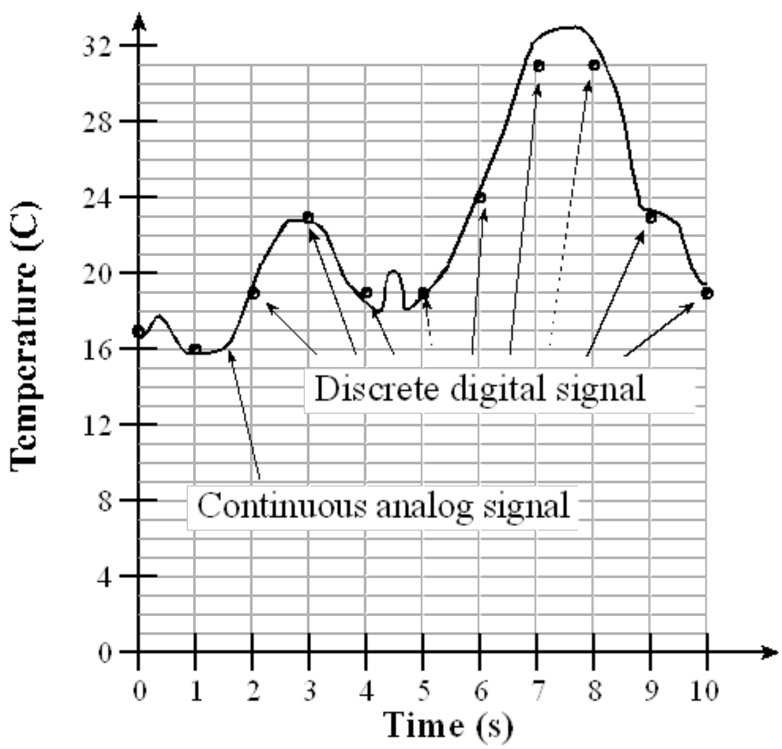

Figure 13.1. An analog signal is represented in the digital domain as discrete samples

The second approximation occurs in the time domain. Time quantizing is caused by the finite sampling interval. For example, the data are sampled every 1 second in Figure 13.1. In practice we will use a periodic timer to trigger an analog to digital converter (ADC) to digitize information, converting from the analog to the digital domain. Similarly, if we are converting from the digital to the analog domain, we use the periodic timer to output new data to a digital to analog converter (DAC). The **Nyquist Theorem** states that if the signal is sampled with a frequency of f_S, then the digital samples only contain frequency components from 0 to ½ f_S. Conversely, if the analog signal does contain frequency components larger than ½ f_S, then there will be an **aliasing** error during the sampling process (performed with a frequency of f_S). Aliasing is when the digital signal appears to have a different frequency than the original analog signal. Also note, the digital data has 11 values at times 0 to 10, but no information before time=0, and no information after time=10.

CHECKPOINT 13.1

Why can’t the digital samples represent the little wiggles in the analog signal?

Hide Answer

Because the frequency components of the wiggles are higher than ½ the sampling rate. The Nyquist Theorem is violated.

CHECKPOINT 13.2

Why can’t the digital samples represent temperatures above 31 oC?

Hide Answer

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Because temperatures above 31 ^{o}C are beyond the range, which is defined in this example as 0 to 31 ^{o}C.

CHECKPOINT 13.3

What range of frequencies is represented in the digital samples when the ADC is sampled once every millisecond?

Hide Answer

If the sampling rate is 1000 Hz, according to the Nyquist Theorem, the digital data can reliably represent frequencies from 0 to 500 Hz.

CHECKPOINT 13.4

If I wanted to create an analog output wave with frequency components from 0 to 11 kHz, what is the slowest rate at which I could output to the DAC?

Hide Answer

According to the Nyquist Theorem we would have to output to the DAC at 22 kHz.

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