Istanbul Technical University
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BLG438E Digital Signal Processing Lab
Experiment 3

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Results From Experiment

Exp 3.1

Sinusoidal wave with 1 kHz frequency generated by function generator was sampled in a time controlled interrupt with 2 kHz frequency. Then, sampled signal was given to oscilloscope. For the interrupt, Timer 0 in CC5515 was used. Registers and used values of Timer 0 are shown below:

Address	Register	Description	Value
1810h	TCR	Control register (32 bit)	0x802EU
1812h	TIMPRD1	Period register 1 (16 bit)	(98304000 / 4096U)/2000U
1813h	TIMPRD2	Period register 2 (16 bit)	0

TCR value denotes the prescaler divider as 4096, which results 100MHz/4096=24,4kHz frequency for interrupt counter.

TIMPRD1 value is (98304000 / 4096U)/2000U, equals to 12. Counter value/frequency gives interrupt period, 12/24,4kHz   0,5ms for this case, which corresponds to 2 kHz.
For 5 kHz sampling (0,2 ms period) required counter value is 4,88, which can be obtained from approximately (98304000 / 4096U)/5000U as TIMPRD1 value.

In this experiment, output signal could not observed correctly for different input and sampling frequencies, probably due to a technical fault about instruments.

Exp 3.2

Program to inverse a sine wave (other functions or parameter adjustments are not given):

```
interrupt void TINT_isr(void) {
    AIC_read2(&right, &left);
    right *= -1;
    left *= -1;
}

int main() {
...
...
        while(1){
                AIC_write2(right, left);
            }
    }
```

Program to change amplitude of input wave (other functions or parameter adjustments are not given):

```
interrupt void TINT_isr(void) {
    AIC_read2(&right, &left);
    right *= A;
    left *= A;
}

int main() {
...
...
        while(1){
                AIC_write2(right, left);
            }
}
```

Shannon Sampling Theorem

Shannon Sampling Theorem indicates that to sample a continious input signal, the sampling frequency must be at least double of the input signal. In case of sampling with lower than the double of the input signal frequency (undersampling), a different uncorrect signal(aliased) will be observed.

Adequately Sampled Signal

Aliased Signal Due to Undersampling
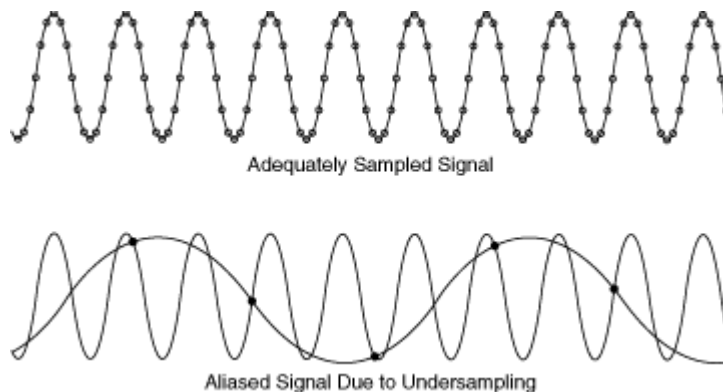
Quantization Error

Quantization error is the difference between analogue signal and digital signal which is obtained by sampling of this analogue signal. It is observed as when the analogue input cannot be mapped to its exact value, due to quantization process truncates or rounds the analogue input.

Quantization error resuls as quantization noise, which is observed as an extra unwanted signal. Increasing the level of quantization lowers the power of this unwanted signal.