Communication Systems II - Laboratory

Experiment 1

Pulse Amplitude Modulation

Objective

1. To investigate the process of *sampling* and the implications of its different practical implementations.
2. To investigate the possibility of message *signal reconstruction* from a sampled signal waveform through *filtering*.
3. To describe how PAM receiver works.
4. To examine the working of the blocks which make up the demodulator: reception amplifier, clock generator, and demodulator.

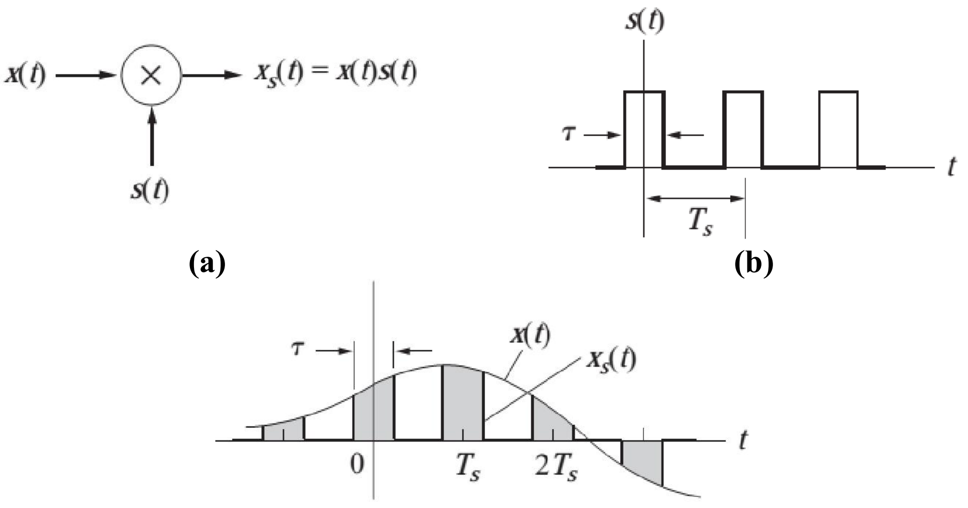
Apparatus

1. T20A board of the ElettronicaVeneta experimental kit.
2. DC power supply.
3. Oscilloscope.
4. Connecting wires.

Theory

Continuous amplitude signals could be classified as either continuous-time or discrete-time signals. Discrete-time signals might be produced by ideally sampling an analog information-bearing (message) signal. It involves the process of multiplying the message signal by a periodic train of impulses to produce an ideally sampled signal. Practical sampling, on the other hand, is effected either by *natural sampling* or by the process of *flat-top sampling* which is more formally known as *pulse amplitude modulation* (PAM).

Natural sampling simply involves mixing the message signal with a pulse train of a relatively small duty cycle, in which case the pulse amplitude of the PAM signal will reflect the shape of the original message signal within its time span. A typical functional diagram and the corresponding input and output signal waveforms illustrating the process of natural sampling are shown in Fig. 1. Flat-top sampling, on the other hand, involves the process of naturally sampling the output of a sample-and-hold (S/H) block with the message signal at its input. The S/H process is illustrated by the input/output waveforms shown in Fig. 2, while a typical message signal and the corresponding PAM signal waveforms are shown in Fig. 3.



(a) **(b)**

(c)

Fig. 1 Natural Sampling: (a) functional diagram; (b) sampling pulse train; (c) typical message and sampled signal waveforms.

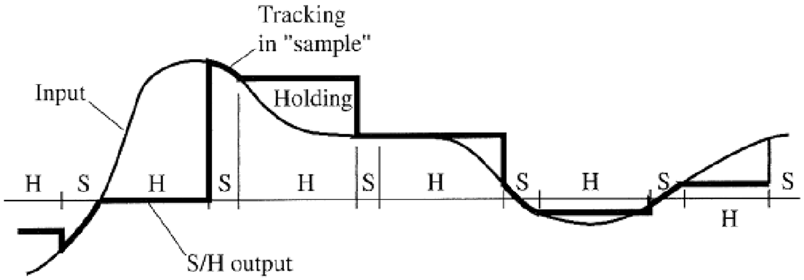


Fig. 2 Typical input and output signal waveforms of an ideal sample and hold circuit.

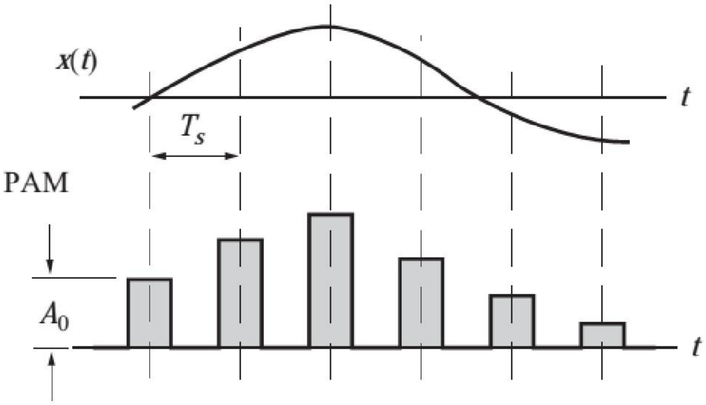


Fig. 3 Typical message and pulse-amplitude modulated (PAM) signal waveforms.

It can be shown that the process of sampling in the time domain (ideal or practical) involves replicating the message signal spectrum in the frequency domain as shown in Fig. 4 for the case of naturally sampling a bandlimited message signal $x(t)$. It is evident though that the nature of the spectrum of the sampled signal $x_s(t)$ depends explicitly on the value of the *sampling frequency* $f_s = 1/T_s$ relative to that of the message signal, W. If f_s is greater than or at least as much as twice the message bandwidth (known as the *Nyquist rate*) as shown in Fig. 4-b then the message signal may simply be reconstructed through lowpass filtering. This of course is based on the assumption that the reconstruction filter has a sharp cutoff characteristic to reject any contribution from nearby replicas of the message spectrum. If, on the other hand, f_s is

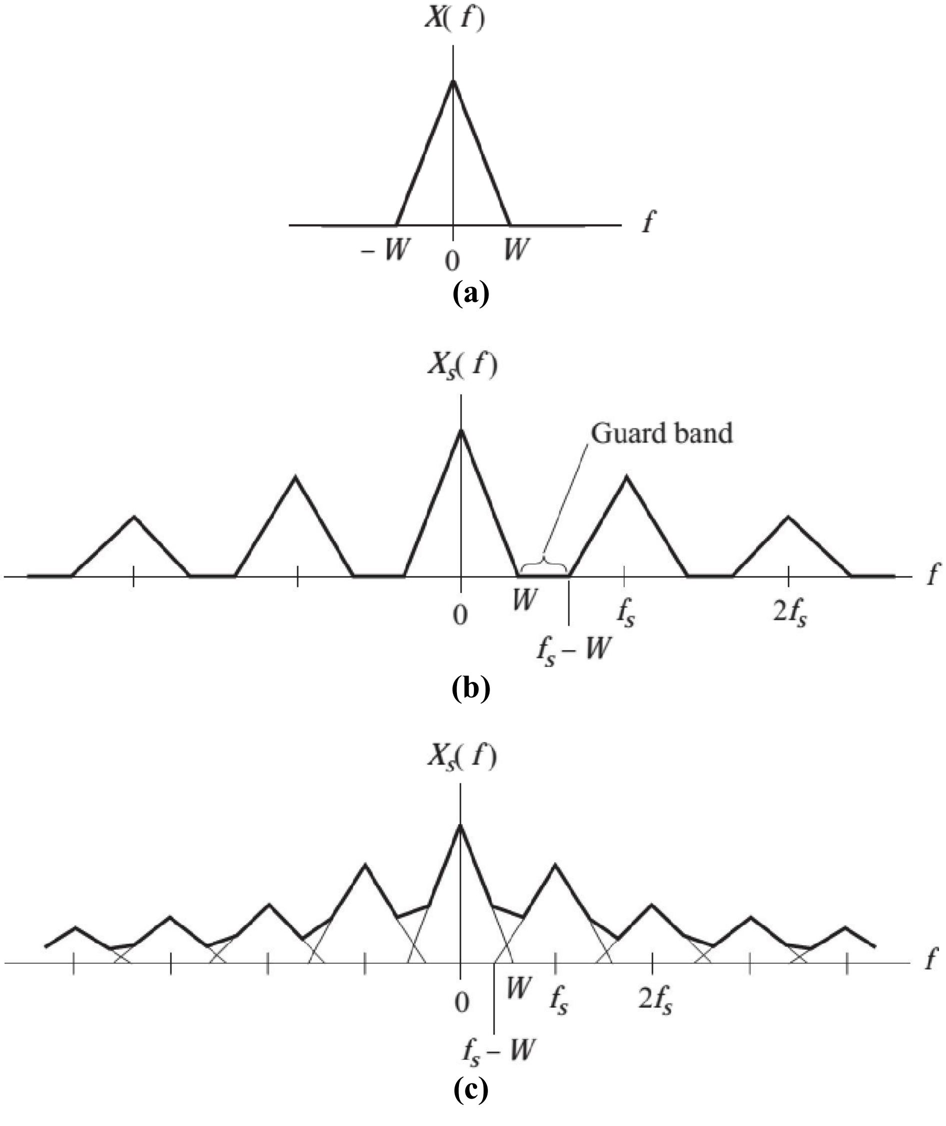


Fig. 4 Typical natural-sampling spectra: (a) message spectrum bandlimited to W; (b) oversampled signal spectrum ($f_s > 2W$); (c) undersampled signal spectrum ($f_s < 2W$).

less than twice the message bandwidth (as shown in Fig. 4-c), *aliasing* occurs and distortion-free reconstruction of the message signal is impossible. Since message signals cannot be strictly bandlimited, aliasing could not be really avoided unless the message signal bandwidth is intentionally limited by incorporating a lowpass *anti-aliasing* filter at the front end of the sampling network.

Another source of distortion yet is the process of flat-top sampling itself when employed for the generation of PAM signals. It can be shown that flat-top sampling is equivalent to passing an ideally sampled waveform, $x_\delta(t)$, through a network having the transfer function $P(f) = \mathcal{F}[p(t)]$, where $p(t)$ is the shape of the pulse involved in the sampling process. The high-frequency rolloff characteristic of a typical $P(f)$ acts like a lowpass filter and attenuates the upper portion of the message spectrum as shown in Fig. 5. This loss of high frequency content is called *aperture effect*. The larger the pulse duration or *aperture* τ, the larger the effect. Aperture effect can be corrected in reconstruction by including an equalizer whose effect is to opposite the aperture effect. However, little if any equalization is needed when the duty cycle (τ/T_s) is very small.

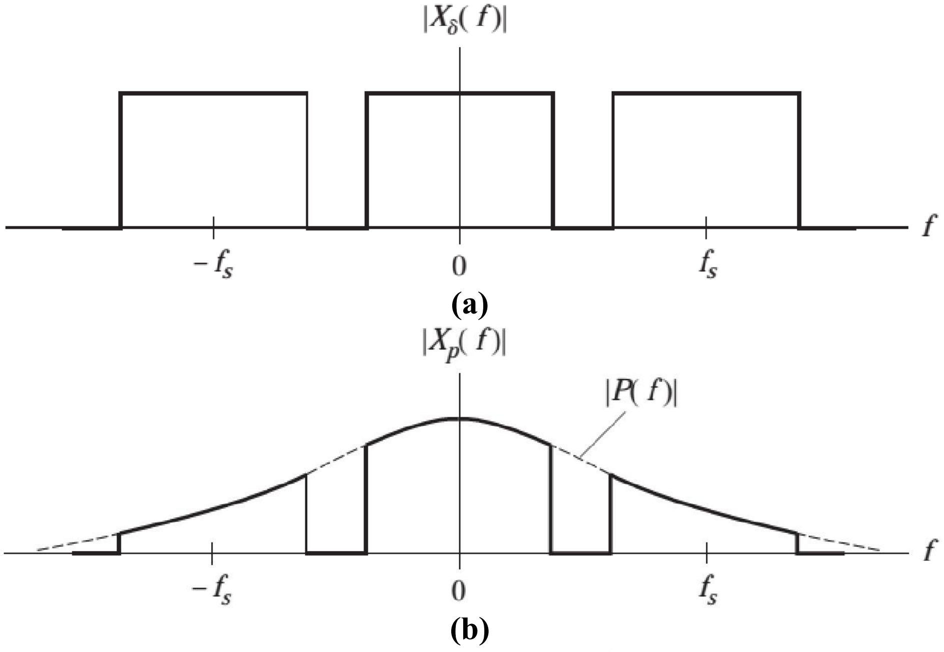


Fig. 5 (a) Spectrum for ideal sampling when $X(f) = \Pi(f/2W)$; (b) aperture effect in flat top sampling.

As we saw before, it is enough to use lowpass filter to demodulate PAM signals but this simple solution doesn’t guarantee good connection quality and can't be used in PAM time-division multiplexing (TDM). In order to improve the overall

PAM link a relatively more sophisticated approach is utilized in practice in which the received PAM signal is used to reconstruct an S/H signal similar to that used in the construction of the flat-top PAM signal at the receiver as depicted in Fig. 1. Then, the inherent step discontinuities in the reconstructed S/H signal could be smoothed out with a simple lowpass filter whose cutoff characteristics need not be as sharp as that required in case of message reconstruction by direct lowpass filtering the received PAM signal.

In the basic block diagram of Fig.6 the received PAM pulses are used to regenerate the sampling pulse train at the receiver to be used then along with the received PAM signal to drive the S/H block in time in order to regenerate the required S/H signal. The sampler output is kept at steady level until the following sample arrives, thereby generating a step signal which approximates the starting signal. The signal reconstructed from the step signal has wider amplitude than the signal reconstructed directly from the PAM pulses. It also comprises fewer harmonics than the starting signal, which makes its filtering easier. In the experiment kit, The PAM coming from the transmitter is amplified and subsequently applied to two sections: the sampling pulse regenerator and the demodulator (S/H block). The demodulator output signal is filtered through a lowpass filter which produces a demodulated analogue signal.

The regeneration of the sampling pulses for the demodulator is carried out as follows. The amplified PAM signal passes through a limiting circuit which reduces the signal amplitude variations. The next bandpass filter (adjusted at 8 or 12 kHz according to the sampling frequency adopted during the transmission) separates the sampling frequency related component. Such a component gets to a PLL circuit which generates a synchronous sampling signal with the PAM signal pulses it receives. The next circuit adjusts the phase of the pulses coming from the PLL, for them to coincide with the maximum amplitude of the PAM pulses going to the demodulator (S/H block).

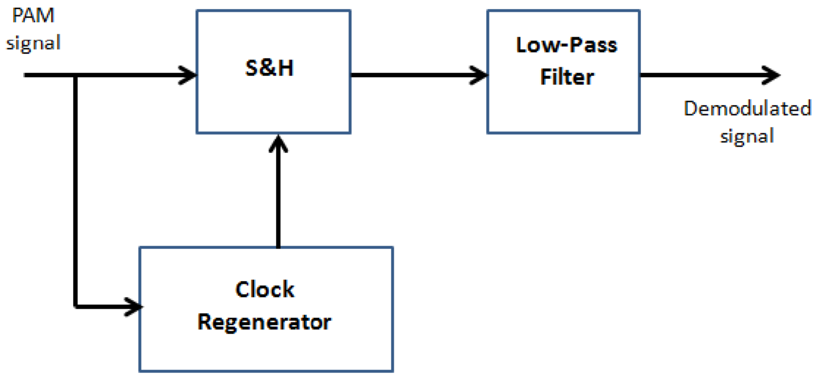


Fig. 6 PAM Receiver.

Procedure:

Part I: Natural Sampling

1- Use the PAM block of the T20A board to generate a naturally sampled signal with the sampling signal set to 8 kHz frequency with a pulse width of 10 μs, and the message signal to a 1 kHz sinusoid of 1.6 V_{pp}.
2- Display and sketch the sampling pulse train, the message, and the naturally sampled signals.
3- Use the 3.4 kHz filter in the R_X section of the T20A to reconstruct the message signal from the naturally sampled one. Display and sketch the reconstructed message signal.
4- Use the cascade of the 3.4 kHz and 5 kHz filters in the R_X section of the T20A to reconstruct the message signal from the naturally sampled one. Display and sketch the reconstructed message signal.
5- Repeat steps 1-4 above with the 5 kHz message frequency.

Part II: Flat-Top Sampling

1. With the message signal the same as that used in Part I, use the T_X section of the T20A board to generate a flat-top sampled (PAM) signal with same sampling signal settings used in I-1 above.
2. Display and sketch the timing and output signals of the S/H block, and the generated PAM signal in this case.
3. Use the 3.4 kHz filter in the R_X section of the T20A to reconstruct the message signal from the flat-top sampled one. Display and sketch the reconstructed message signal.
4. Use the cascade of the 3.4 kHz and 5 kHz filters in the R_X section of the T20A to reconstruct the message signal from the flat-top sampled one. Display and sketch the reconstructed message signal.
5. Repeat steps 1-4 above with the 5 KHz message frequency.

Part III: PAM Transceiver

1- Use the PAM block of the T20A board to generate a flat-top sampled signal with the sampling signal set to 8 kHz frequency with a pulse width of 10 μs, and the message signal to a 1 kHz sinusoid of 1V_{pp} taking into consideration the effect of loading.
2- Display and sketch the sampling pulse train, the message, and the flat-top sampled signals.
3- Use the R_X section of the T20A to reconstruct the message signal from the flat-top sampled one.
4- Use the amplifier block in the R_X section of the T20A to display and sketch the amplified PAM signal.
5- Use the Limiter block in the R_X section of the T20A to display and sketch the limited amplified PAM signal.

6- Use the bandpass filter block in the R_X section of the T20A to obtain a sinusoidal waveform having the same frequency at the PAM pulses at the receiver input. Display and sketch the filtered signal.
7- Use the PLL block in the R_X section of the T20A which gives you a square wave having the same frequency at the PAM pulses at the receiver input. Display and sketch the output signal.
8- Display and sketch the amplified signal (output of the amplifier) and output signal from the Phase Adjust.
9- Display and sketch the PAM signal at the input of pulse regeneration step and output of PAM demodulator (S/H).
10- Use the Phase Adjust Probe in order to obtain the maximum step signal amplitude at the S/H output.
11- Use the 3.4 kHz filter in the R_X section of the T20A to reconstruct the message signal from the flat-top sampled one. Display and sketch the reconstructed message signal.

Discussion

1. Give appropriate mathematical descriptions of ideal, flat-top and naturally sampled signals both in the time and frequency domains and explain the differences between them.
2. Explain the reasons behind distortion received in the reconstructed message waveform in parts I and II of the experiment when the message frequency is varied.
3. Compare between demodulating PAM signal using lowpass filtering only and demodulating PAM signal using (S/H).
4. Explain why the phase adjust regulator varies the amplitude of the demodulated signal (use figures to boost your explanation).
5. Consider the circuit shown in Fig.7. Draw the signals at point a and b.

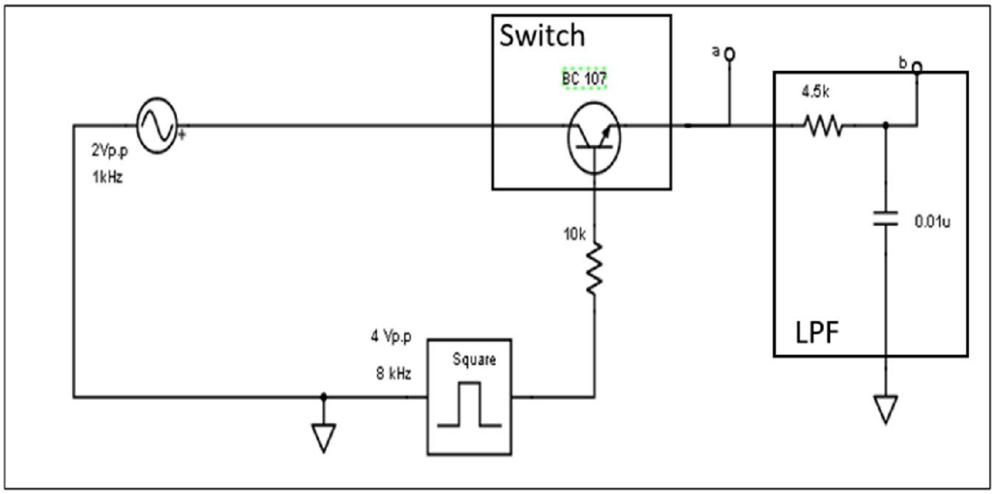


Fig.7.

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References

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[2] *EV-T10A and EV-T10B Training Kit Manuals*, Elettronica Veneta & Inel Spa, Treviso, Italy