Design of IIR Filter Using Chebyshev

Mandeep Singh Saini[1] and Dr. Kuldeep Kaur[2]

[1]*M.Tech. (E.C.E), Guru Nanak Dev Engineering College, Ludhiana .P.T.U Regional Centre, (Punjab) India.*

[2]*Ph. D (Asst. Prof.), Faculty Member of B.A.M. Khalsa College, Garhshankar. (Punjab)*

Abstract - **In this paper we examine the performance of IIR Chebyshev filters. Higher filter order is disadvantages because the cost of filter is increased and more multipliers are required. But consider a filter of the same order without ripples in the pass band and stop band with the advantage of providing steeper transitions between pass band and stop band. For comparison, Notice the wider transitions that result as a tradeoff. Hence this type of filter plays very important role in spectral analysis of different types of signal. In spectral analysis applications, a small main lobe width of the window function in frequency domain is required for increasing the ability to distinguish two closely spaced frequency components.**

Keywords:-Butterworth, Chebyshev, Cauer, IIR, FIR

[1] INTRODUCTION

a) Need of IIR Filters

Chebyshev Type II filters are sometimes called inverse Chebyshev filters because of their relationship to Chebyshev Type-I filter.

The magnitude response of a Chebyshev type-I filter is equiripple in the pass band and monotonic in the stop band and the magnitude response of a Chebyshev type II filter is monotonic in the pass band and equiripple in the stop band.

The primary advantage of IIR filters over FIR filters is that they typically meet a given set of specifications with a much lower filter order than a corresponding FIR filter. Although IIR filters have nonlinear phase, data processing IIR filters they require a very small number of multipliers to implement, they are inherently stable and have low round off noise sensitivity and no limit cycles. Furthermore, it is possible to achieve almost linear phase designs. [7]

TABLE 1.1

IIR CHEBYSHEV FILTER TYPES

Filter Design	Description
Chebyshev type I	The magnitude response of a Chebyshev type I filter is equiripple in the pass band and monotonic in the stop band.
Chebyshev type II	The magnitude response of a Chebyshev type II filter is monotonic in the pass band and equiripple in the stop band.

b) Chebyshev type I

Chebyshev filters are analog or digital filters having a steeper roll-off and more pass band ripple (type-I) or stop band ripple (type- II) than Butterworth filters.

Digital Domain

[z, p, k] = chebyshev1(n, R, W) designs an order n Chebyshev low pass digital Chebyshev filter with normalized pass band edge frequency W_p and R dB of peak-to-peak ripple in the pass band. It returns the zeros and poles in length n column vectors z and p and the gain in the scalar k [z, p, k] = chebyshev1(n,R,W_p,'f_{type}') designs a high pass, low pass, or band stop filter, where the string 'ftype' is one of the following.

i. 'high' for a high pass digital filter with normalized pass band edge frequency W_p
ii. 'low' for a low pass digital filter with normalized pass band edge frequency W_p
iii. 'stop' for an order 2*n band stop digital filter if W_p is a two-element vector, $W_p = [w_1\ w_2]$. The stop band is $w_1 < \omega < w_2$.

Normalized pass band edge frequency *is* the frequency at which the magnitude response of the filter is equal to R dB. For chebyshev1, the normalized pass band edge frequency

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W_p is a number between 0 and 1, where 1 corresponds to half the sample rate, π radians per sample. Smaller values of pass band ripple R lead to wider transition widths (shallower roll off characteristics).

If W_p is a two-element vector, $W_p = [w_1\ w_2]$, chebyshev 1 returns an order 2*n band pass filter with pass band $w_1 < \omega < w_2$.

With different numbers of output arguments, chebyshev 1 directly obtains other realizations of the filter. To obtain the transfer function form, use two output arguments as shown below.

[b,a] = chebyshev1(n, R,W_p) designs an order n Chebyshev low pass digital Chebyshev filter with normalized pass band edge frequency W_p and R dB of peak-to-peak ripple in the pass band. It returns the filter coefficients in the length n+1 row vectors b and a, with coefficients in descending powers of *z*.

$$H_z = \frac{b(1) + b(2)Z^{-1} + b(n+1)Z^{-n}}{1 + a(2)Z^{-1} + a(n+1)Z^{-n}} \quad 2.1$$

[b,a] = chebyshev1(n,R,W_p,'f_{type}') designs a high pass, low pass, or band stop filter, where the string 'f_{type}' is 'high', 'low', or 'stop', as described above.

To obtain state-space form, use four output arguments as shown below [7].

[A,B,C,D] = chebyshev1(n,R,W_p) or

[A,B,C,D] = chebyshev1(n,R,W_p,'f_{type}')

Where A, B, C, and D are

$$x[n+1] = Ax[n] + Bu[n]$$
$$y[n+1] = Cx[n] + Du[n]$$

and *u* is the input, *x* is the state vector, and *y* is the output.

c) Chebyshev type II

Chebyshev Type II filters are monotonic in the pass band and equiripple in the stop band. Type II filters do not roll off as fast as type I filter, but are free of pass band ripple.

Digital Domain

[z,p,k] = chebyshev II (n,R,W_{st}) designs an order n low pass digital Chebyshev Type II filter with normalized stop band edge frequency W_{st} and stop band ripple R dB down from the peak pass band value. It returns the zeros and poles in length n column vectors z and p and the gain in the scalar k. Normalized stop band edge frequency is the beginning of the stop band, where the magnitude response of the filter is equal to -R dB. For chebyshev II, the normalized stop band edge frequency W_{st} is a number between 0 and 1, where 1 corresponds to half the sample rate. Larger values of stop band attenuation R lead to wider transition widths (shallower roll off characteristics).

If Wst is a two-element vector, $\mathbf{w_{st}} = [\boldsymbol{\omega_1}\ \boldsymbol{\omega_2}]$, chebyshev-II returns an order 2*n band pass filter with pass band $\boldsymbol{\omega_1 < \omega < \omega_2}$

[z,p,k] = chebyshev II (n,R,W_{st},'f_{type}') designs a high pass, low pass, or band stop filter, where the string 'ftype' is one of the following:

- 'high' for a high pass digital filter with normalized stop band edge frequency W_{st}
- 'low' for a low pass digital filter with normalized stop band edge frequency W_{st}
- 'stop' for an order 2*n band stop digital filter if W_{st} is a two-element vector, $W_{st} = [w_1\ w_2]$. The stop band is $w_1 < \omega < w_2$ Width different numbers of output arguments, chebyshev2 directly obtains other realizations of the filter. To obtain the transfer function form, use two output arguments as shown below.

[b,a] = chebyshev II(n,R,W_{st}) designs an order n low pass digital Chebyshev Type II filter with normalized stop band edge frequency Wst and stop band ripple R dB down from the peak pass band value. It returns the filter coefficients in the length n+1 row vectors b and a, with coefficients in descending powers of *z*.

$$H_z = \frac{b(1) + b(2)Z^{-1} + b(n+1)Z^{-n}}{1 + a(2)Z^{-1} + a(n+1)Z^{-n}} \quad 2.1$$

- [b,a] = chebyshev II(n,R,W_{st},'f_{type}') designs a high pass, low pass, or band stop filter, where the string 'ftype' is 'high', 'low', or 'stop', as described above
- To obtain state-space form, use four output arguments as shown below.

[A,B,C,D] = chebyshevII(n,R,W_{st}) or

[A,B,C,D] = chebyshev II (n,R,W_{st},'f_{type}')

where A, B, C, and D are

$$x[n+1] = Ax[n] + Bu[n]$$
$$y[n+1] = Cx[n] + Du[n]$$

and u is the input, x is the state vector, and y is the output

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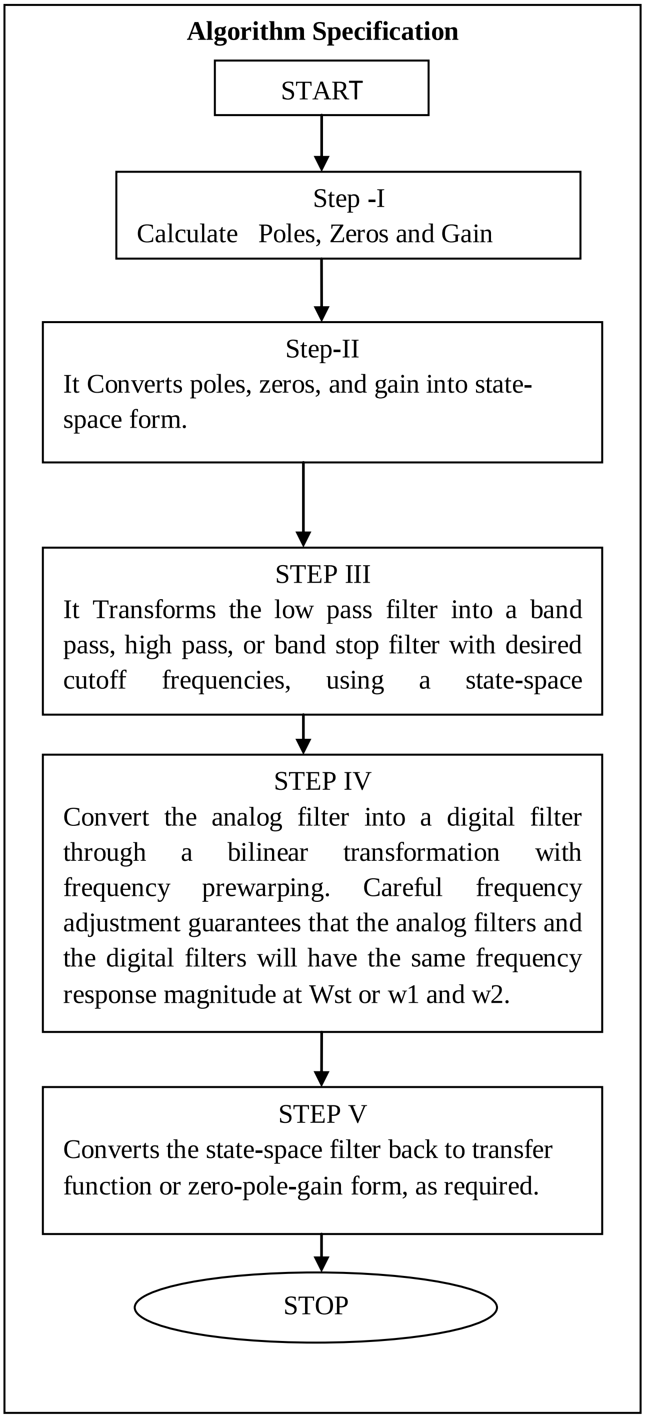
[II] PROPOSED FILTER DESIGN



[III] RSEULTS & DISCUSSION

Digital filters used in audio for adjusting the frequency content of a sound signal. Parametric equalizers provide capabilities beyond those of graphic equalizers by allowing the adjustment of gain, center frequency, and bandwidth of each filter. In contrast, graphic equalizers only allow for the adjustment of the gain of each filter.

These filters have the drawback that because of their low order (Low Pass filter order=4), they can present relatively large ripple or transition regions and may overlap with each other when several of them are connected in cascade. Figure 3.1 shown LPF and Figure 3.2 showed HPF.

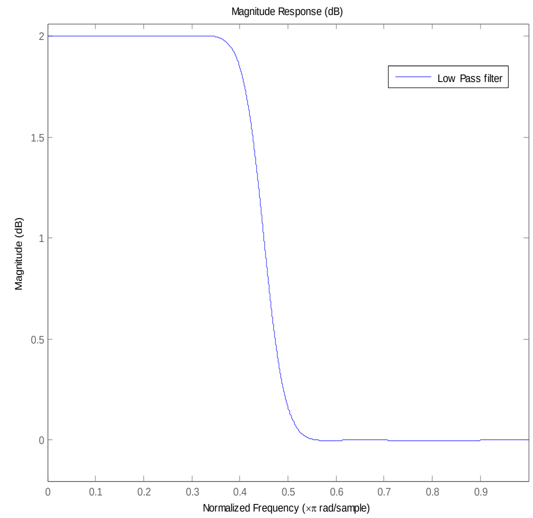


Fig. 3.1 Magnitude response of a Low Pass filter

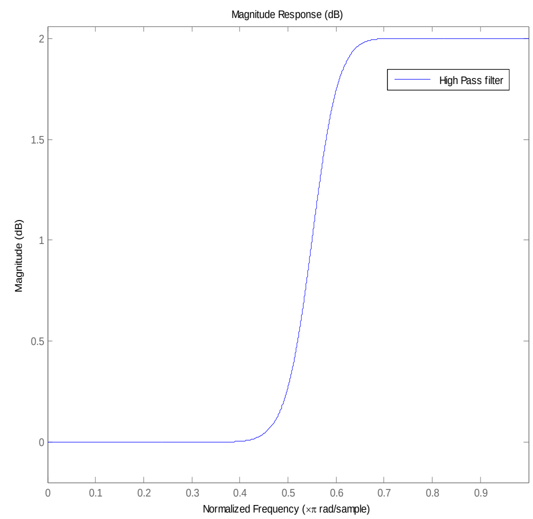


Fig. 3.2 Magnitude response of a High Pass filter

Chebyshev Pass Band Filter order = 04) designs provide much more control over the shape of each filter.

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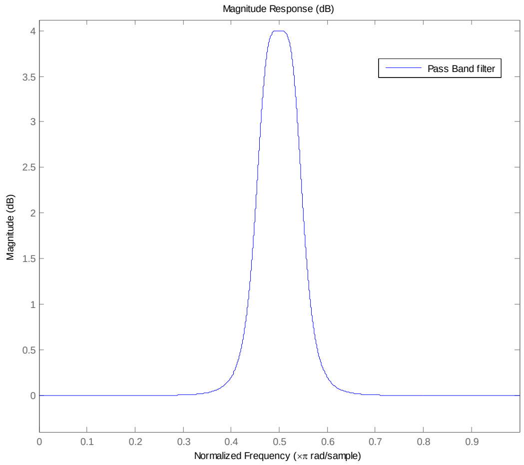
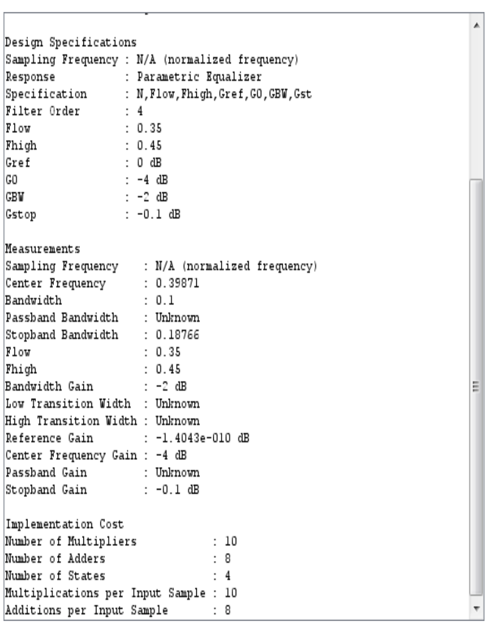


Fig. 3.3 Magnitude response of a Chebyshev Band Pass filter

Notice that we have specified stop band filter is designed with filter order = 4, lower Frequency at Gain GBW= 0.45, high Frequency at Gain GBW= 0.55 and bandwidth is 0.1 Notice how the fourth-order filter is closer to an ideal brickwall filter when compared to the second-order design. Obviously the approximation can be improved by increasing the filter order even further. The price to pay for such improved approximation is increased implementation cost as more multipliers are required.



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Design Specifications
Sampling Frequency : N/A (normalized frequency)
Response           : Parametric Equalizer
Specification      : N,Flow,Fhigh,Gref,G0,GBW,Gst
Filter Order       : 4
Flow               : 0.35
Fhigh              : 0.45
Gref               : 0 dB
G0                 : -4 dB
GBW                : -2 dB
Gstop              : -0.1 dB

Measurements
Sampling Frequency    : N/A (normalized frequency)
Center Frequency      : 0.39871
Bandwidth             : 0.1
Passband Bandwidth    : Unknown
Stopband Bandwidth    : 0.18766
Flow                  : 0.35
Fhigh                 : 0.45
Bandwidth Gain        : -2 dB
Low Transition Width  : Unknown
High Transition Width : Unknown
Reference Gain        : -1.4043e-010 dB
Center Frequency Gain : -4 dB
Passband Gain         : Unknown
Stopband Gain         : -0.1 dB

Implementation Cost
Number of Multipliers            : 10
Number of Adders                 : 8
Number of States                 : 4
Multiplications per Input Sample : 10
Additions per Input Sample       : 8
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TABLE 3.1
IIR FILTER TYPES

FILTER TYPE	ORDER	MUL.	ADDER	STATES	MPIS	APIS
Chebyshev Filter	04	10	08	04	10	08

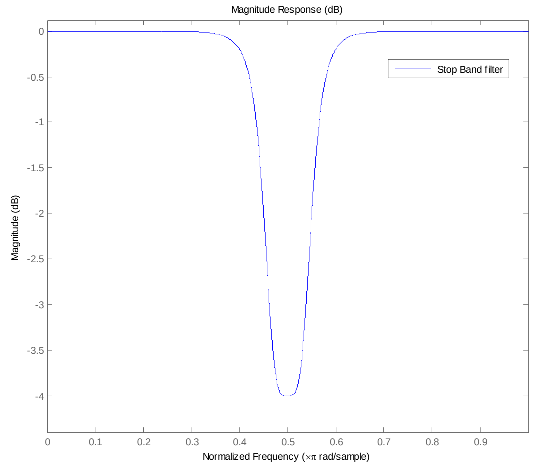

Fig. 3.4 Magnitude response of a Chebyshev-II Band Stop filter

[IV] CONCLUSION

Theory and practice prove, digital audio signal Processing system using IIR digital Chebyshev filter,
The Chebyshev IIR Stop Pass Filter is examined from the above simulation result it is found that the designed IIR filter has lesser main lobe width so it can be used in speech processing applications such as speech filtering, noise reduction, frequency boosting and digital audio equalizing etc. In speech filtering; filter are used to modify the frequency response of a speech signal according to require applications of speech processing.

[V] APPLICATION

These types of filter is uses in Audio Equalizer System to control the voice signal shown in Fig. 5.1

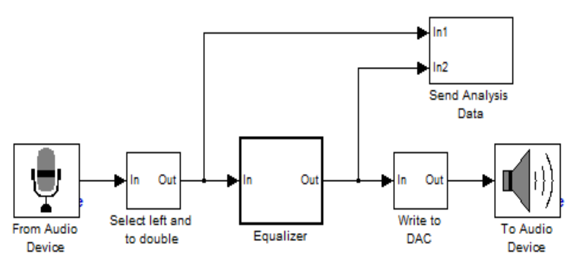

Fig. 5.1 Audio Equalizer System

From Audio Device
Record the sound data from analog system audio device.
Select Left and to Double

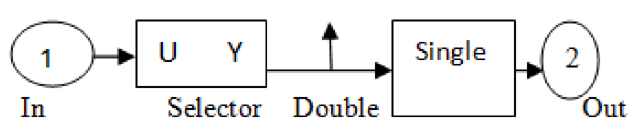

Fig. 5.2 Select Left and to Double

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Input

Provide an input port for a subsystem or model. For Triggered Subsystems, 'Latch input by delaying outside signal' produces the value of the subsystem input at the previous time step.

For Function-Call Subsystems, turning 'On' the 'Latch input for feedback signals of function-call subsystem outputs' prevents the input value to this subsystem from changing during its execution.

Selector

Select or reorder specified elements of a multidimensional input signal. The index to each element is identified from an input port.

Data type conversion

Convert the input to the data type and scaling of the output. The conversion has two possible goals. One goal is to have the Real World Values of the input and the output is equal. The other goal is to have the Stored Integer Values of the input and the output is equal. Overflows and quantization errors can prevent the goal from being fully achieved.

Output

Provide an output port for a subsystem or model. The 'Output when disabled' and 'Initial output' parameters only apply to conditionally executed subsystems. When a conditionally executed subsystem is disabled, the output is either held at its last value or set to the 'Initial output'.

Equalizer

Coefficients are slewed between changes. Each band is implemented using a different bi-quad filter structure.

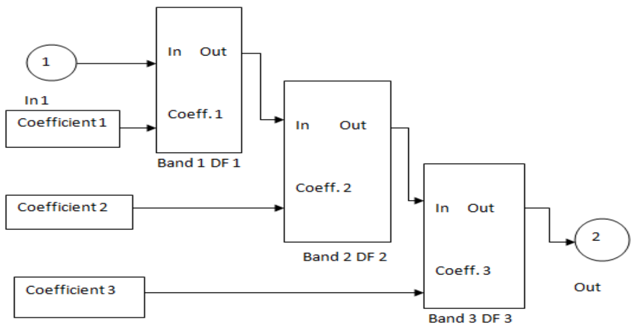

Fig. 5.3 Three band Equalizer

Send Analysis Data

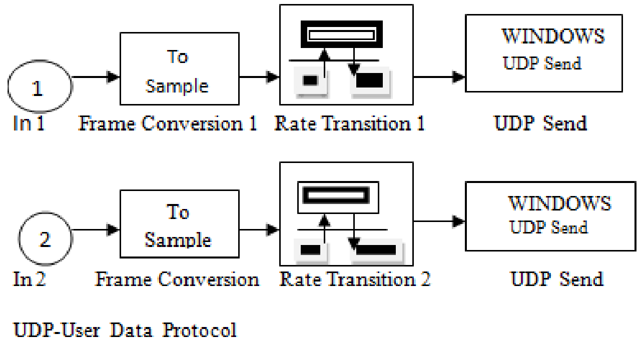

Fig. 5.4 Analysis data

Frame conversion

Set the sampling mode of the output signal.

Rate Transition

Handle transfer of data between ports operating at different rates. Configuration options allow you to trade off transfer delay and code efficiency for safety and determinism of data transfer. The default configuration assures safe and deterministic data transfer. The block's behavior depends on option settings and/or the sample times of its input and output ports. Updating the block diagram causes text on the block's icon to indicate its behavior as follows:

ZOH: Zero Order Hold

1/z: Unit Delay

Buf: Copy input to output under semaphore control

Db_buf: Copy input to output, using double buffers

Copy: Unprotected copy from input to output

NoOp: No Operation

Mixed: Expanded to multiple blocks with different behaviors

User Datagram Protocol

The User Datagram Protocol (UDP) is a transport protocol layered on top of the Internet Protocol (IP) and is commonly known as UDP/IP. It is analogous to TCP/IP. A convenient way to present the details of UDP/IP is by comparison to TCP/IP as presented below:

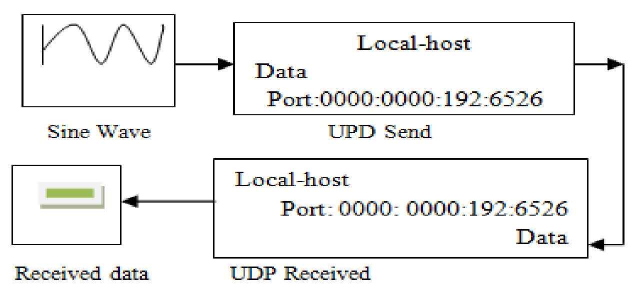

Fig. 5.5 Basic UPD communization

Connection Versus Connectionless

TCP is a connection based protocol, while UDP is a *connectionless* protocol. In TCP, the two ends of the

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communication link must be connected at all times during the communication. An application using UDP prepares a packet and sends it to the receiver's address without first checking to see if the receiver is ready to receive a packet. If the receiving end is not ready to receive a packet, the packet is lost.

Stream Vs Packet

TCP is a *stream-oriented* protocol, while UDP is a *packet-oriented* protocol. This means that TCP is considered to be a long stream of data that is transmitted from one end to the other with another long stream of data flowing in the other direction. The TCP/IP stack is responsible for breaking the stream of data into packets and sending those packets while the stack at the other end is responsible for reassembling the packets into a data stream using information in the packet headers.UDP, on the other hand, is a packet-oriented protocol where the application itself divides the data into packets and sends them to the other end. The other end does not have to reassemble the data into a stream. Note, some applications might indeed present the data as a stream when the underlying protocol is UDP. However, this is the layering of an additional protocol on top of UDP, and it is not something inherent in the UDP protocol itself.

TCP Is a Reliable Protocol; While UDP Is Unreliable

The packets that are sent by TCP contain a unique sequence number. The starting sequence number is communicated to the other side at the beginning of communication. Also, the receiver acknowledges each packet, and the acknowledgment contains the sequence number so that the sender knows which packet was acknowledged. This implies that any packets lost on the way can be retransmitted (the sender would know that they did not reach their destination because it had not received an acknowledgments). Also, packets that arrive out of sequence can be reassembled in the proper order by the receiver.

Further, time-outs can be established, because the sender will know (from the first few packets) how long it takes on average for a packet to be sent and its acknowledgment received. UDP, on the other hand, simply sends the packets and does not keep track of them. Thus, if packets arrive out of sequence, or are lost in transmission, the receiving end (or the sending end) has no way of knowing.

Digital to Analog conversion

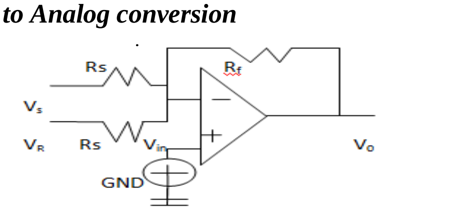

Fig. 5.6 Digital to Analog conversion

Fig. shown 5.6 using IC 741 Non- Inverter Amplifier Let it has two input Left Channel V_s and Right Channel V_R and V_0 is then output is given as:

V_s 0 Volt=Logic 0 5Volt =Logic 1 Digital input	V_R 0 Volt=Logic 0 5Volt =Logic 1 Digital input	$V_0 = \left(1+\frac{Rf}{Rs}\right)(V_S+V_R)$ Logic 0= 0 Volt Logic 1= 5 Volt (Where $R_f = R_S = 1$) Analog output
0	0	0 volts
0	1	5 volts
1	0	5 volts
1	1	10 volts

$$\frac{V_0}{(V_S+V_R)} = \left(1+\frac{Rf}{Rs}\right) \quad if\ Rf = 0$$

Then voltage Gain is unity.

Element-wise gain (y = K.*u) or matrix gain (y = K*u or y = u*K).

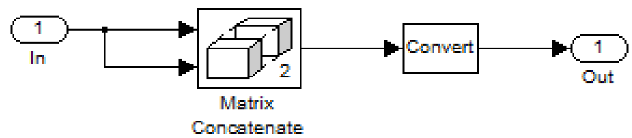

Fig. 5.7 Digital to Analog conversion

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MANDEEP SINGH SAINI was born in Punjab, India on 26 Jan, 1975. He has completed his B.E in Electronics Engineering from Nagpur University (Maharashtra State) in 2001. He has also completed Master of Technology (M. Tech) from Guru Nanak Dev Eng. College, Ludhiana (Punjab) in 2012. His current research and teaching interest are in Digital Signal Processing and Very Large Scale Integration Design. He has also worked has M. Tech Research Guide with more than 24 Journal and conferences.

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