Reconfigurable Architecture to enhance QoS for VoIP networks

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Abstract- **Reconfigurable systems provide improved performance by adapting to computations which are not feasible with current processor architecture.As Voice over IP networks (VoIP) are delay sensitive, maintaining quality of service in these networks is essential. Since VoIP is offering real time services, degradation of QoS due to the channel can be rectified only at the receiver end . In this paper, a reconfigurable architecture that can be implemented in VoIP networks at the receiver end is proposed to enhance the Quality of Service in VoIP networks. The architecture is proposed for two parameters that improves QoS , namely echo cancellation and Packet loss concealment. The runtime reconfiguration to implement the efficient algorithms for these parameters are discussed.**

Key words: Reconfigurable architecture, QoS , Echo Cancellation,Packet loss Concealment ,runtime reconfiguration

I. Introduction

VoIP refers to the transmission of voice over packet switched IP networks. Current IP networks which are based on best effort services, lack stringent QoS control. Congestion is inevitable in IP networks and may result in packet loss, delay and delay jitter, which directly impact the quality of VoIP applications. Echo is another aspect to be nullified, to improve the quality of voice. So, the current VoIP architecture must be enhanced by some QoS guaranteeing mechanisms to ensure QoS in VoIP applications. This paper focuses on QoS enhancement mechanism by implementing reconfigurable architecture.

In this paper, a model to implement reconfigurable hardware is proposed. In section II, we give an overview of QoS parameters for VoIP and in section III, we present an overview and implementation of reconfigurable architecture for VOIP to enhance QoS. In, section IV , simulated results of echo cancellation and PLC algorithms are discussed and switching network needed for reconfigurable architecture is synthesized and its design summary is given.

II. QoS parameters of VoIP

Echo is the repetition of a waveform due to reflection from points where the characteristics of the medium through which the wave propagates changes. Echo can severely affect the quality and intelligibility of voice conversation. The perceived effect of an echo depends on its amplitude and time delay. In general, echoes with an appreciable amplitude and a delay of more than 1 ms are noticeable. Echoes become increasingly annoying and objectionable with the increasing round-trip delay and amplitude in particular for delays of more than 20 ms. Hence echo cancellation is an important design consideration of VoIP systems[1].

To support the transmission of real time voice traffic over IP networks, four main performance parameters should be considered: total end to end delay, packet delay variations (inter arrival jitter), lost or damaged packets, which cause voice frame erasure and out of order packet delivery.

A. Echo cancellation

The echo canceller is basically an adaptive linear filter[1]. The coefficients of the filter are adapted so that the energy of the signal on the line is minimized. This is mainly due to the practical difficulties associated with the adaptation and stable operation of adaptive IIR filters .The echo canceller can be an IIR filter or FIR filter.

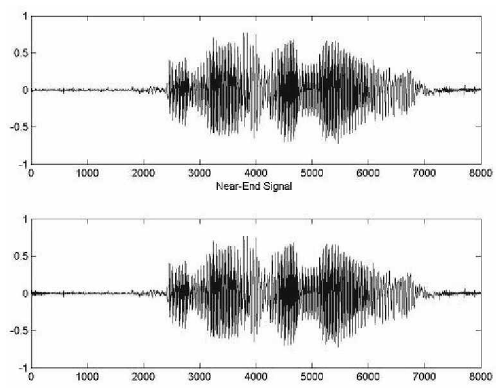

Fig.1 Original and echo cancelled speech signal

In practice, echo cancellers are based on FIR filters. Most echo cancellers use variants of the LMS adaptation algorithm. The drawback of LMS is that it can be sensitive to the eigen value spread of the input signal and is not particularly fast in its convergence rate.

Modified NLMS algorithm:

In this algorithm, an adaptive individual step-size is assigned to each filter coefficient. The step-sizes are calculated from the last estimate of the filter coefficients in such a way that a larger coefficient receives a larger increment, thus increasing the convergence rate of that coefficient.This has the effect that active coefficients are adjusted faster than non-active coefficients. Hence, MNLMS converges much faster than NLMS for sparse impulse responses.*x(n)* is Far-end signal, *y(n)* is the received signal and *h(n)* is Echo with background noise possibly including near-end signal.Estimated echo path. is $\hat{h}(n)=[\hat{h}_0(n).....\hat{h}_{L-1}(n)]^T$

The excitation vector is $x(n)=[x(n).....x(n-L+1)]^T$ and L is the length of the adaptive filter. The MNLMS algorithm is described by the following equations:

$$e(n)=y(n)-\hat{h}^T(n-1)x(n) \qquad (1)$$

$$\hat{h}(n)=\hat{h}(n-1)+\mu e(n).\left[\frac{1}{x(n-1)\delta/[D(n)x(n)]+x^T(n)}\right] \qquad (2)$$

D(n) is a diagonal matrix which adjusts the step-sizes of the individual taps of the filter, μ is the overall step-size parameter, and δ is a regularization parameter which stabilizes the solution when speech is used as the input (far-end) signal.

The diagonal elements D(n) are calculated as follows:

$$d_t(n+1)=L\tau_t(n+1)/\sum_{i=0}^{L-1}\tau_i(n+1) \qquad (3)$$

$$\tau_t(n+1)=\max[\rho\max\{\delta_p,|\hat{h}_0(n)|,.....,|\hat{h}_{L-1}(n)|\},|\hat{h}_t(n)|] \qquad (4)$$

The MNLMS algorithm has faster convergence rate than NLMS algorithm

B. Packet loss concealment(PLC)

Packet loss is an important problem with regard to the deployment of Internet real-time services. Loss concealment algorithms typically add a delay of at least that corresponding to one packet length, because the algorithm is triggered only when a missing packet has been detected. If the packet following the missing packet is needed only for detection and not for the concealment operation itself, the concealment algorithm could be started immediately after the receipt of the previous packet and prepare a replacement packet without any indication if the packet under consideration will really be lost.

1) Waveform substitution: The replacement of a missing signal segment by another segment which is generated from correctly received speech is called “waveform substitution". The procedure is: identification of gaps in the signal as either a missing packet or silence (when silence detection is enabled) using sequence number and timestamps, buffering of recently received signal segments, signal processing to replace the missing segment.

2) Silence Substitution: The replacement of a missing signal segment by silence period is called “silence substitution".

3)Packet Repetition: The repetition of the most recently received packet is the simplest method to approximate the missing waveform. It is only necessary to buffer a copy of the last packet. Because the packetization interval L is not related to the speech pitch period p, discontinuities in the signal occur. Together with a typically reverberating sound caused by exactly the same speech material to be played twice, this method results in a only slightly improved speech quality as compared to silence substitution[6,2].

4)LP-based waveform substitution: The well-known technique of linear prediction is an interesting candidate to alleviate the packet loss problem. Fig.2. shows the approach based on linear prediction. When a packet is correctly received, the PCM signal x(n) (represented by its z transform X(z)) is used to compute the LP filter coefficients. The difference signal (D(z)) is then fed to the LP synthesis filter (^D (z) = D(z)) which uses the computed filter coefficients, resulting in an output signal (^X (z) = X(z)) which is identical to the input signal (assuming ideal filters).

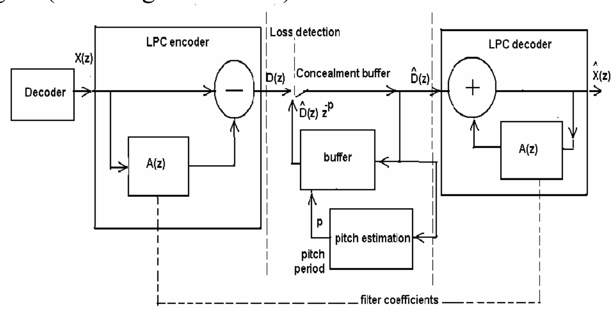

Fig.2.LP-based waveform substitution

Additionally to these operations, the pitch period p is estimated and a segment of the LP difference signal corresponding to this period is buffered. When a packet loss is detected, no LP analysis is performed, however the previous difference signal (^D (z)z_p) which has been buffered is used as the replacement excitation to excite the LP synthesis filter using the previous filter parameters. For codecs which are based on a linear prediction or transform coding, it is possible that the decoder algorithm is run with repeated or estimated parameters. Fig.4 shows LPC based concealed speech signal.

Interleaving

A simple method to increase the audibility of a loss-distorted signal is interleaving, i.e. sending parts of the same signal segment in different packets, thus spreading the impact of loss over a longer time period. Particularly for voice this property has been reported to be useful in terms of enhanced speech quality due to the long-term correlation property. Interleaving always needs buffering of generated data at the sender and re-sequencing at the receiver, thus introducing a higher latency.

Fig.3 shows the interleaving of “units" (e.g. voice frames): a number of units are associated to a group (here the group size is G = 12). Units, which are in a certain distance of each other (interleaving distance D = 3), are packetized together (packet size P = G/D = 4). In the event of a loss, the burst loss of P units is traded against P isolated losses of unit

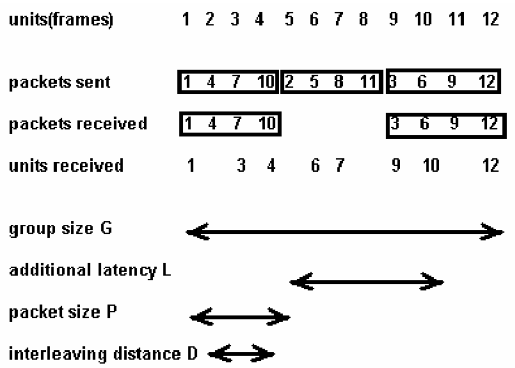

Fig.3 Interleaving technique

size. The additional latency introduced is thus L = (P - 1)D + 1 - P = (P - 1) (D - 1) = 6 units. This delay is added permanently to the play-out delay, because units have to be buffered at the sender before being interleaved and finally packetized. Note that the mean bandwidth of the flow is not changed (no redundant data is generated), however the flow exhibits more burstiness: Packet departure times for the non-interleaved case are after the generation of unit 4, 8 and 12 respectively. When interleaving is used the earliest departure of the three packets of the group is after unit 10, 11 and 12 respectively.

III. Reconfigurable architecture

Reconfigurable computing systems use FPGAs or other programmable hardware to accelerate algorithm execution by mapping compute-intensive calculations to the reconfigurable substrate The strength of reconfigurable processor is the ability to customize hardware for a specific program's requirements. Algorithm execution is partitioned between a main algorithm and reconfigurable coprocessor. Coprocessor takes care of the few regular kernels of computations that are responsible for a large fraction of execution time and energy, while main processor executes all remaining original algorithm tasks and manages reconfigurable specific tasks like reconfiguration, input data uploading and output data downloading[3,9,10].

Runtime reconfiguration is based upon the concept of virtual hardware. The Physical hardware is much smaller than the sum of resources required for each configurations. Instead of reducing number of configurations that are mapped, we swap them in and out of actual hardware as they are needed. Run time reconfigurable systems are able to optimize based on values that are known only at run time. Partially reconfigurable models are more suitable for the VoIP applications, as the configurations for applying various QoS coefficients do not occupy the full reconfigurable hardware, only a part of a configuration requires modification.

B. Architecture:

By analyzing the VoIP algorithms and the existing architecture, we can divide them into two groups of basic computation structures : Filtering (for echo cancellation), and loss concealment. The following principles must be complied to improve performance: 1) more than one to investigate the parallelism inherent in the algorithms; 2) communication bandwidth must be wide enough to avoid routing conflict; 3) memory must be available for Reconfigurable Function Unit (RFU) to work less wastage of cycles.

The architecture (Fig.4) consists of mainly a) Controller b) Address Generator c) Configuration memory d) Programmable I/O e) Switching matrix with memory f)

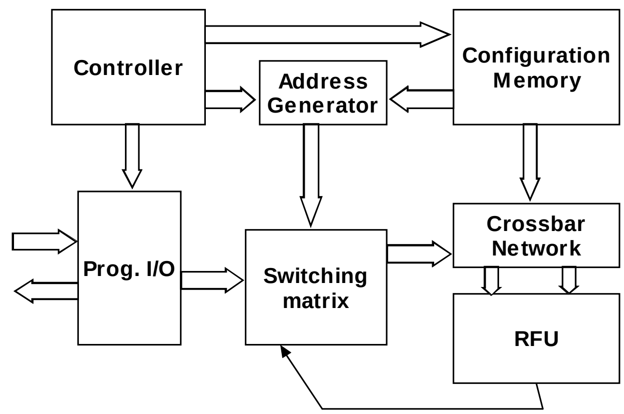

Fig.4Architecture

Crossbar network g) RFUs. Except Controller and memory , all the units are reconfigurable. The function of controller is to load the memory with configuration words and addressing them. Address Generator consists of a simple logic to generate the address sequence needed by the targeted algorithms.

Programmable I/O adapts the chip to interface with other devices. Configuration memory are stored with configuration words meant for RFUs. The memory and crossbar network

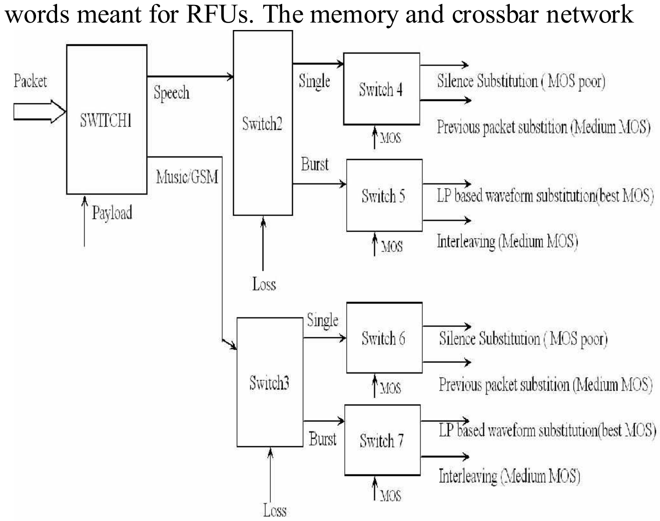

Fig.5 Switching Network

can provide data for RFUs and RFUs can process these data without wasting time.

***Implementation of Reconfiguration for packet loss concealment*:**

The packet loss can be detected from the sequence number of the packets received and the loss may be bursty. Single packet loss can be easily concealed where as bursty loss requires efficient algorithm.Loss concealment technique to be applied may vary according to payload(speech from telephone , music, digitized speech from PC,GSM etc.). The payload type can be extracted from the RTP(Realtime Transport Protocol) packet(Fig.6). RTP is the protocol used for real time
services over IP networks.

As packet loss is bursty in IP networks, LP based waveform substitution is the preferred loss concealment technique. Mean Opinion Score(MOS) is another factor used to implement different loss concealment techniques. MOS is a technique used to measure QoS in IP networks[15].

V=2	Pad	Ext	CSRC count	Marker	Payload type	Sequence No.
Timestamp						
Synchronization Source Identifier (SSRC)						
Contributing Source Identifiers (CSRC)						
Data variable						

Fig.6 RTP packet format

The schematic of the switching network that includes crossbar network for packet loss concealment is shown in fig.5. Reconfigurable Function Unit(RFU) consists of modules
to implement different PLC techniques. Selection of the technique is based on the type of payload, loss and MOS .This can be realized in Xilinx Virtex Pro FPGA.

IV. Simulation

The switching network used in the reconfigurable architecture was simulated and synthesized using Xilinx ISE 7.1i and the design summary is as follows.

Design Information

Command Line : I:/Xilinx/bin/nt/map.exe -ise j:\switch\switch.ise -intstyle
ise -p xcv600e-bg432-6 -cm area -pr b -k 4 -c 100 -tx off -o switch_map.ncd
switch.ngd switch.pcf
Target Device : xcv600e
Target Package : bg432
Target Speed : -6
Mapper Version : virtexe -- $Revision: 1.26.6.3 $
Mapped Date : Sat May 21 17:35:42 2005

Design Summary

Number of errors: 0
Number of warnings: 3
Logic Utilization:
Number of 4 input LUTs: 47 out of 13,824 1%
Logic Distribution:
Number of occupied Slices: 45 out of 6,912 1%
Number of Slices containing only related logic: 45 out of 45 100%
Number of Slices containing unrelated logic: 0 out of 45 0%
*See NOTES below for an explanation of the effects of unrelated logic
Total Number of 4 input LUTs: 47 out of 13,824 1%
Number of bonded IOBs: 205 out of 316 64%
IOB Flip Flops: 200
Number of GCLKs: 1 out of 4 25%
Number of GCLKIOBs: 1 out of 4 25%
Total equivalent gate count for design: 1,882
Additional JTAG gate count for IOBs: 9,888

The echo cancellation algorithms were simulated (fig7) in

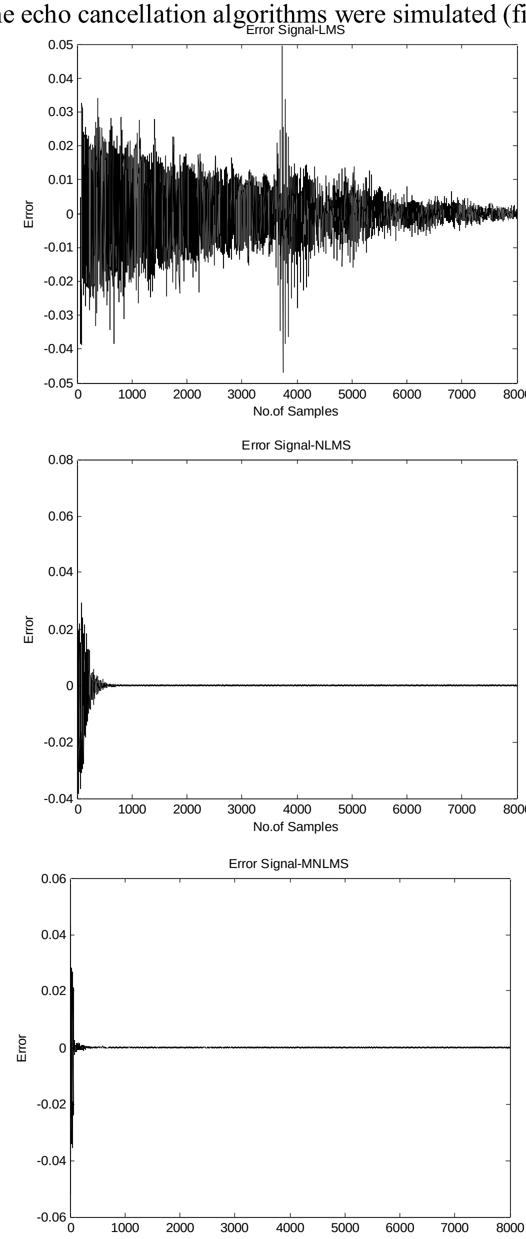

Fig.7 Error signals of LMS,NLMS,MNLMS

MATLAB to show to variations in the convergence rate.. Simulation of packet loss concealment using LP based waveform substitution was done in MATLAB and is shown in fig.8

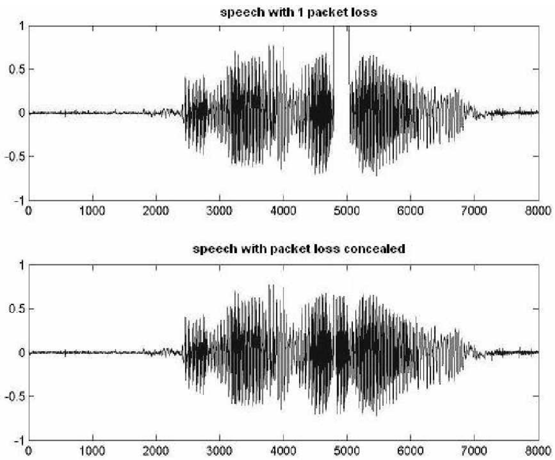

Fig. 8 Original and packet concealed speech signal Using LP based waveform substitution

Conclusion

Since the Quality of Service in VoIP networks is degraded by the various factors like packet loss, echo and jitter, a suitable architecture is required to enhance QoS. Reconfigurable architecture is the only solution and is not yet implemented for VoIP applications. This architecture is used not only to improve QoS but also for power management as the modules will be loaded in the FPGA dynamically.

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