

# 11095 SPH

## G.726A and G.711

# Agenda

- **Speech Coding Overview**
- **Speech Coding Solutions**
- **G726A**
  - Basic Idea into working of the Algorithm
  - Data Structures and Library API
  - LAB #1 and #2
    - With **ADC/PWM PICtail™ Plus** Daughter Board
    - With **dsSPEAK™** Speech Processing Reference Design Board
- **G711**
  - Basic Idea into working of the Algorithm
  - Data Structures and Library API
  - LAB #3
    - with **ADC/PWM PICtail Plus** Daughter Board

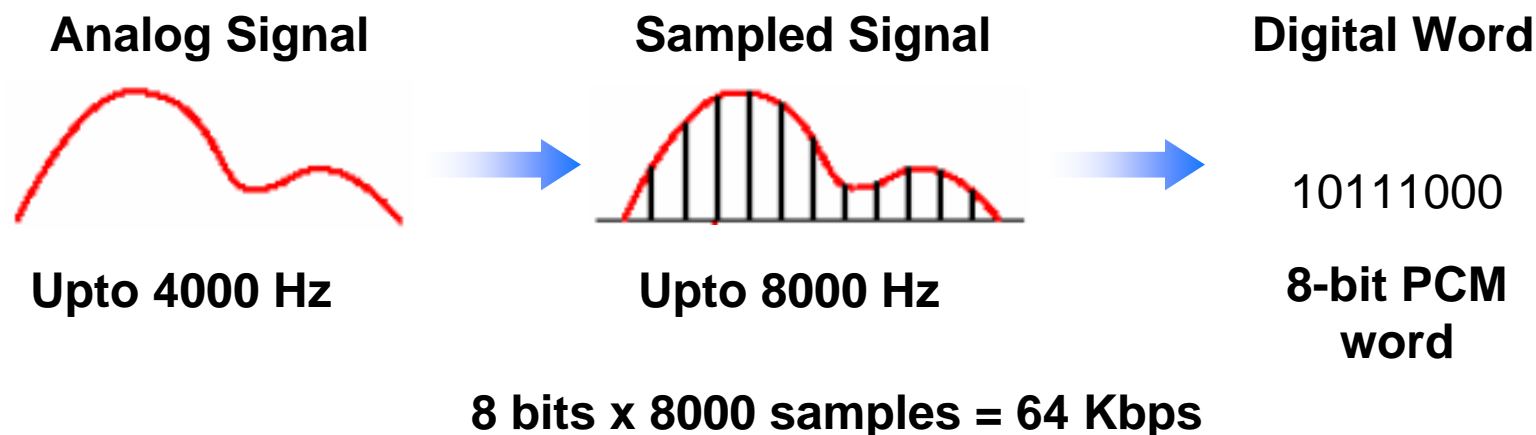
# Learning Objective

**When you finish this class you will:**

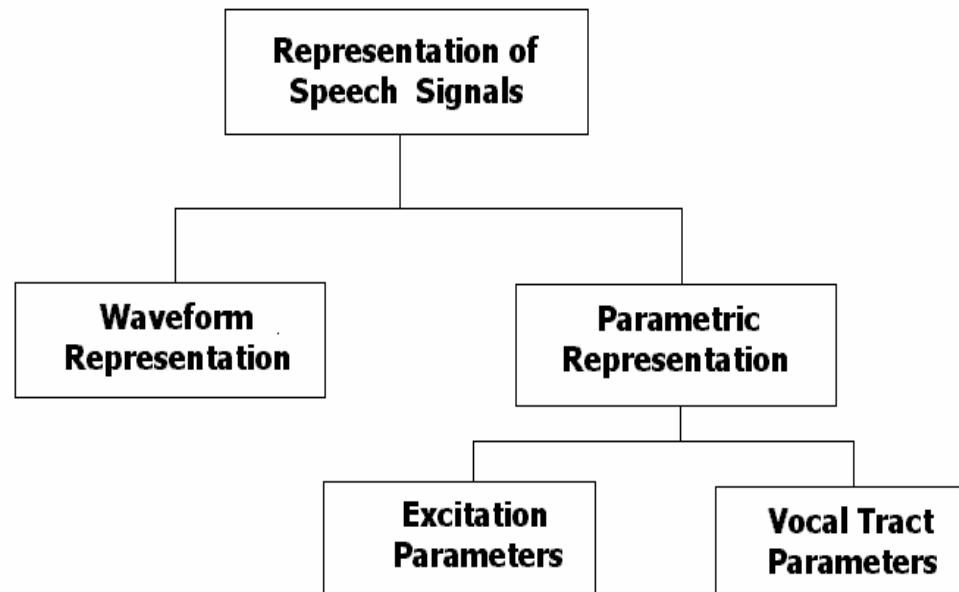
- **Understand features of G711 and G726A speech processing solutions**
- **Look at some common applications**
- **Learn how to use the software functions and demos**

# Speech Coding Overview

- **Data compression – speech**
- **Standard PCM encoding of analog signals**
  - 8 KHz sampling rate – 8-16 bits/sample
  - Requires 64-128 Kbps



# Representation of Speech Signal

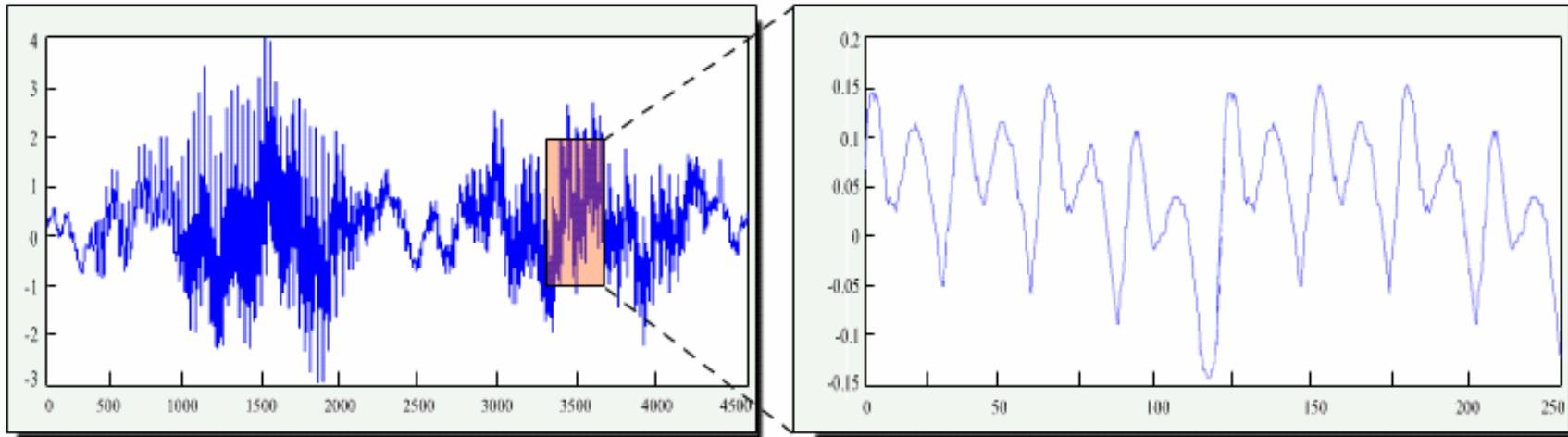


- **Waveform representations** are concerned with simply preserving the wave shape of the analog speech.
- **Parametric representations** are concerned with representing the speech signal as the output of a model for speech production

# Waveform Coders

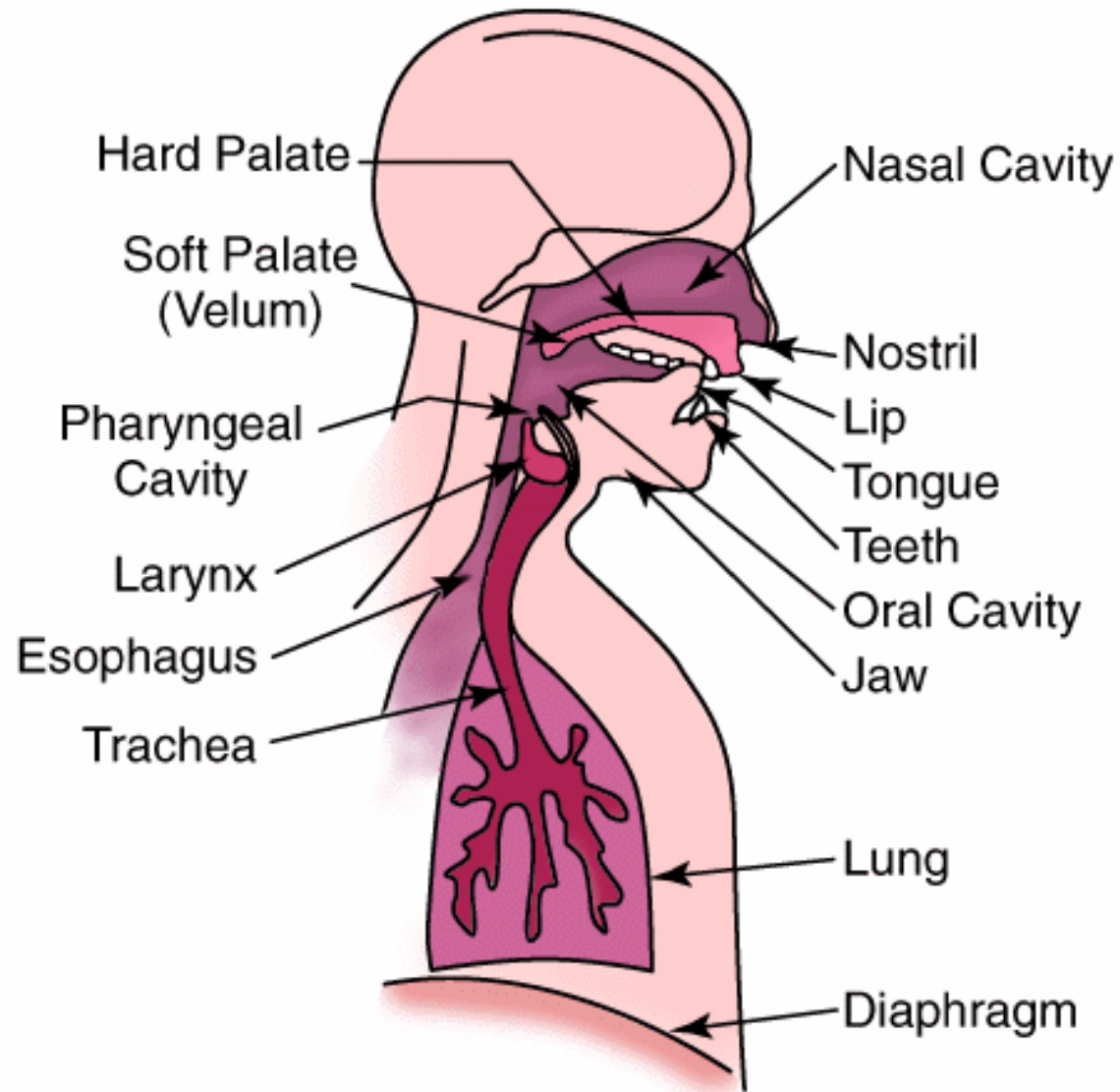
- **Waveform coders encode the shape of the waveform**
- **Faithful reconstruction of the time-domain waveform**
- **Non-speech specific coders, represent nonspeech sounds (music, background noise).**
- **Operate at medium-rates**
- **Examples: PCM,DPCM,ADPCM, Mu/A law**

# Parametric Coders



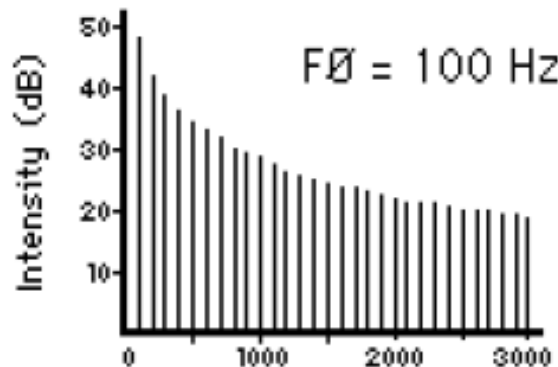
- Speech specific Coders (VOCODERS)
- Speech though random, consists of quasi-stationary signals (Voiced Sounds)
- Parameters of the voiced portion sent instead of the whole waveform
- Compression rates as low as 4 Kbps have been achieved
- Examples: CELP, ACELP

# Know Your Vocal Apparatus

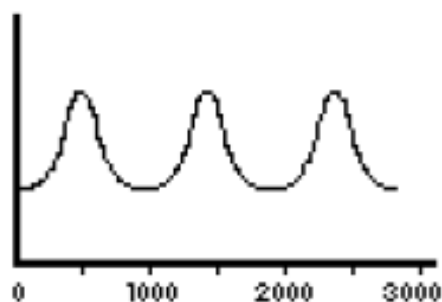




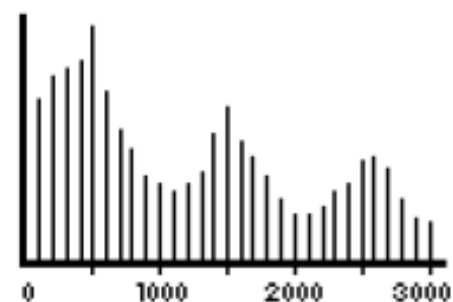
# Know Your Vocal Apparatus



Source spectrum

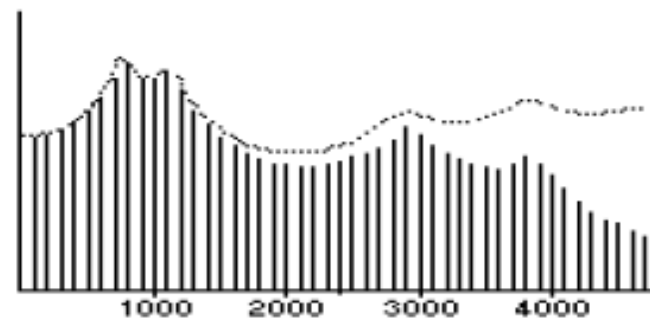


Filter Function

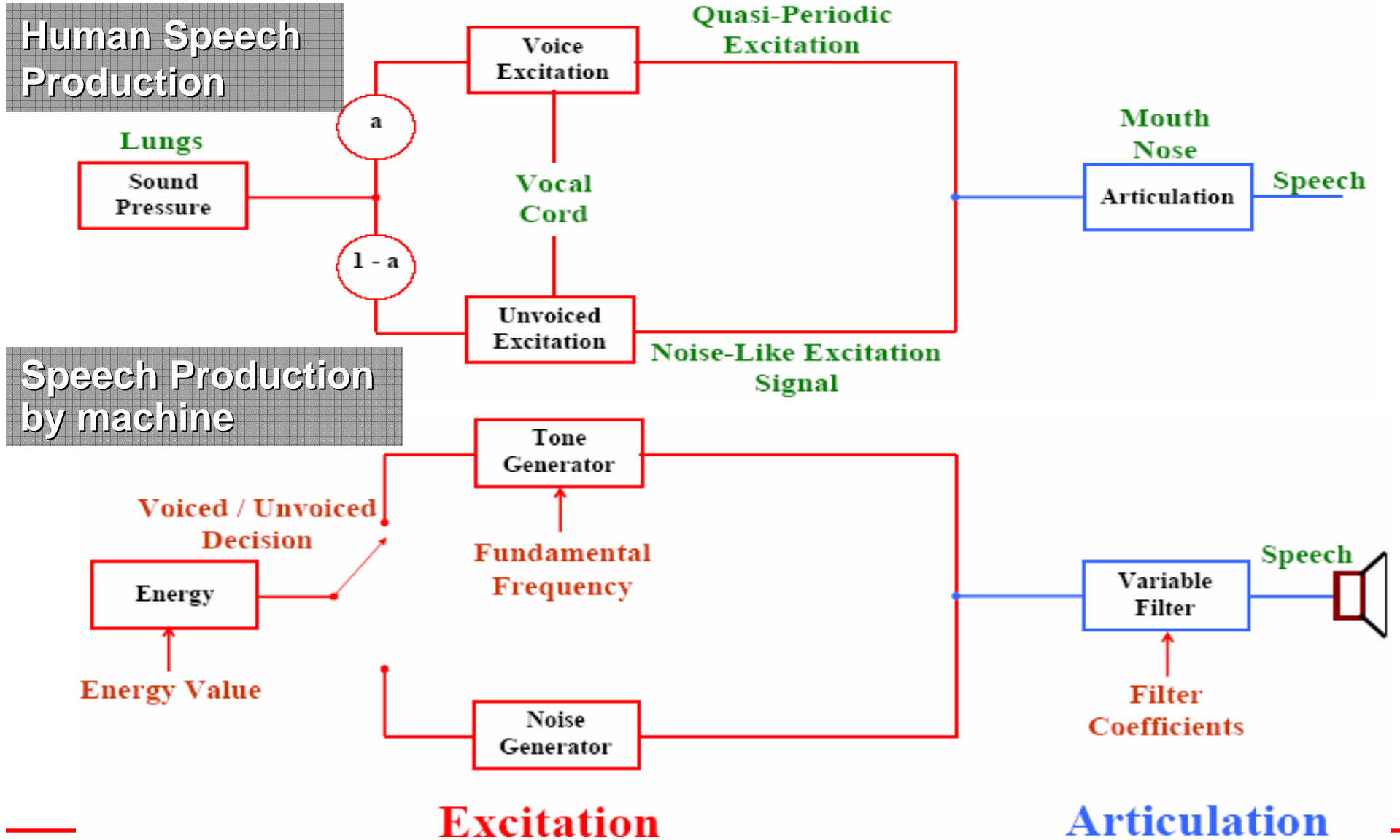


Output Spectrum

AH



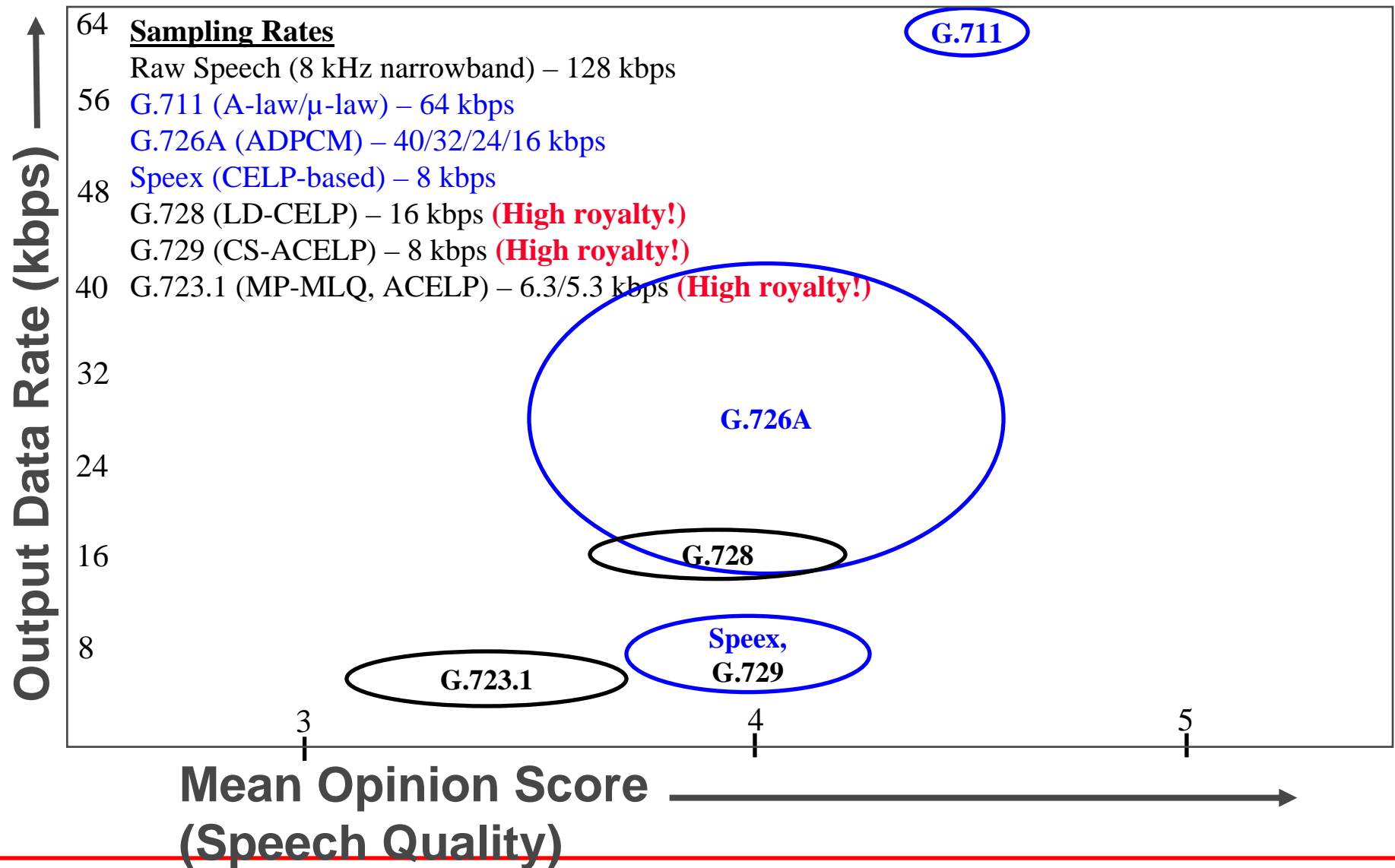
# Model Based Coders



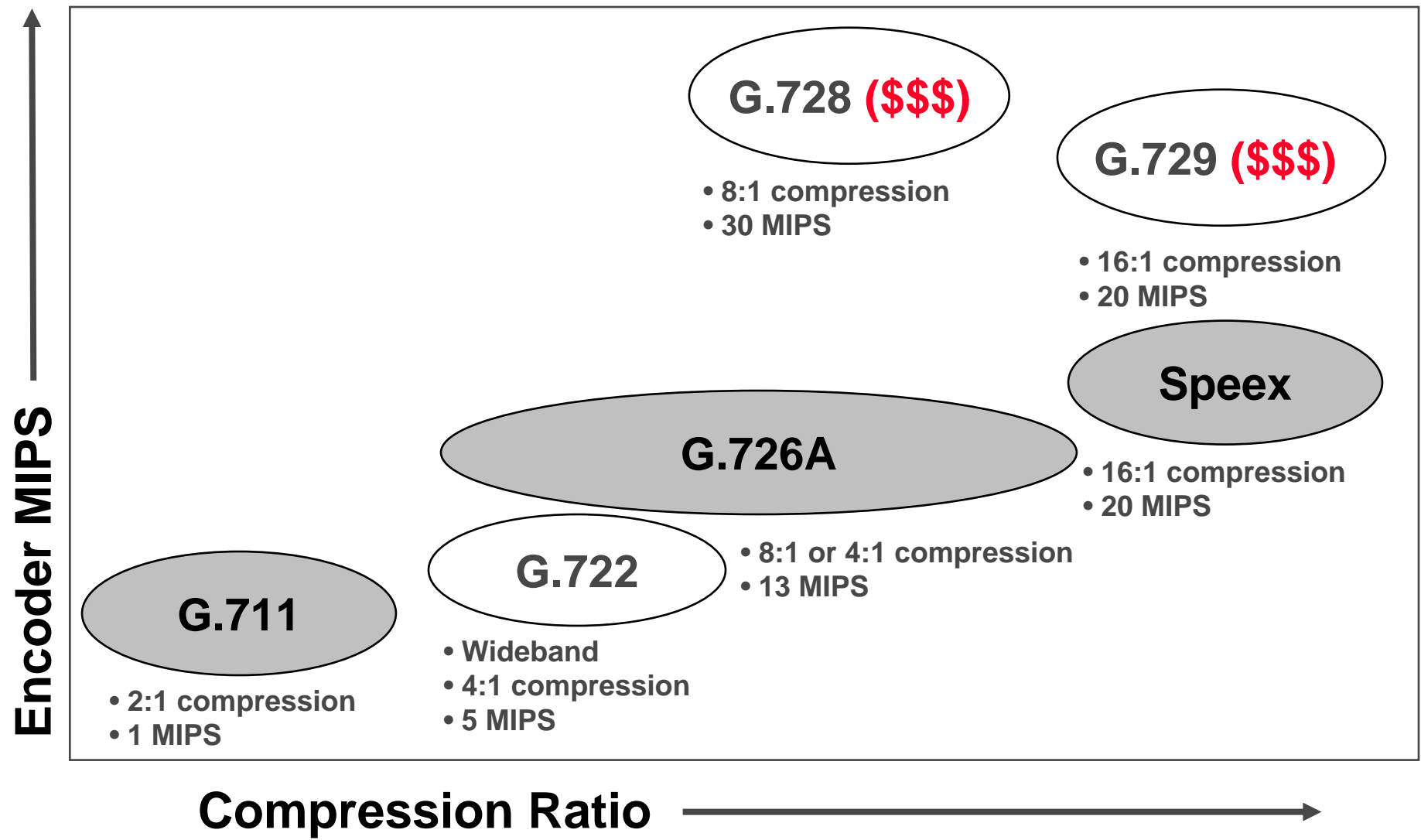
# Speech Coding Solutions

- **Speech Compression (Encoding)**
  - Reduce amount of data required to represent a speech signal, thus reducing communication or storage requirements
- **Speech Decompression (Decoding)**
  - Reconstruct original speech signal from compressed data
- **Suite of speech coding solutions**
  - Speex, G.726A, G.711
  - G.711 also supports PIC24H/PIC24F

# Speech Coding Solutions



# Speech Coding Solutions



# Why use dsPIC<sup>®</sup> DSC for Speech Processing?

- **High processor speed (40 MIPS)**
- **DSP instructions, bit manipulation and data shifting**
- **Fast, deterministic interrupts**
- **Peripherals**
  - Codec interface (DCI)
  - ADC and PWM for alternative low-cost speech I/O interfaces
  - Serial Peripheral Interface (SPI) to transmit and receive compressed data

# Speech Coding Solutions Resource Requirements

	<b>G.711</b>	<b>G.726A</b>	<b>Speex</b>
<b>MIPS</b>	<b>1</b>	<b>13</b>	<b>20</b>
<b>Flash (KB)</b>	<b>3.5</b>	<b>6</b>	<b>30</b>
<b>RAM (KB)</b>	<b>3.5</b>	<b>4</b>	<b>7</b>

# Speech Coding Benefits and Applications

- **Benefits**

- Reduces communication bandwidth
- Reduces storage requirements

- **Sample Applications**

- Communication Systems (full-duplex)
  - Digital radios and walkie-talkies
  - Voice-over-IP phones
- Record-and-Playback Systems (half-duplex)
  - Answering machines and voice recorders
- Playback-only Systems (simplex)
  - Toys, security systems (building evacuation), museum guides



# How to Obtain the Libraries?

- **Order from *buy.microchip.com***
  - 3-tiered Library Pricing for each library
    - **\$2500 for 5,000 unit license**
    - **\$4950 for 25,000 unit license**
    - **\$9750 for 100,000 unit license**
  - **Free** Evaluation License, full-feature package
  - G.711 library is **Free**

# Speech Coding Solution #1

## G.726A Speech Coding Library

# Overview

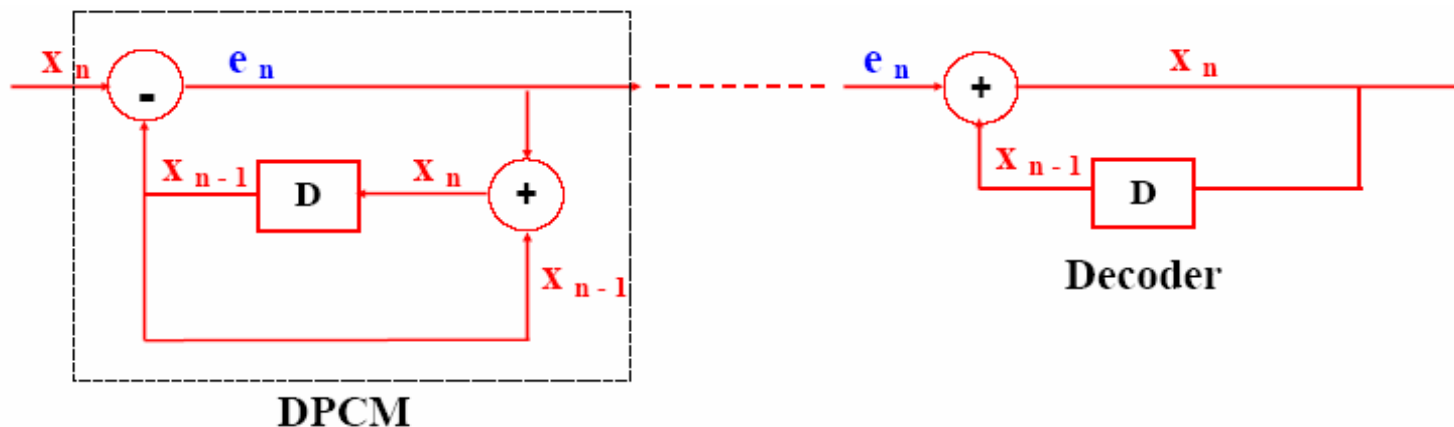
- **Based on Adaptive Differential Pulse Code Modulation (ADPCM)**
  - ITU-T standard, **but no royalties!!**
  - Similar to G.726, but no G.711 needed
  - Very popular for medium data rates
- **Mean Opinion Score (MOS):**
  - 4.5 (40 kbps)
  - 4.4 (32 kbps)
  - 4.1 (24 kbps)
  - 3.4 (16 kbps)

# Differential PCM

- **Motivation:**

- The motivation is that the differenced signal has a smaller amplitude swing compared with the original signal
- Reduces data by encoding a difference signal instead of input signal
- The difference is obtained by subtracting the estimate (Prediction) of the Input signal from the Input Signal

# First Order Prediction



- **Encoding**

$$x_1 \ x_2 \ \dots \ \dots \ x_N \longrightarrow e_1 \ e_2 \ \dots \ \dots \ e_N$$

$$e_1 = x_1 \quad e_n = x_n - x_{n-1}, \quad n=2, \dots, N$$

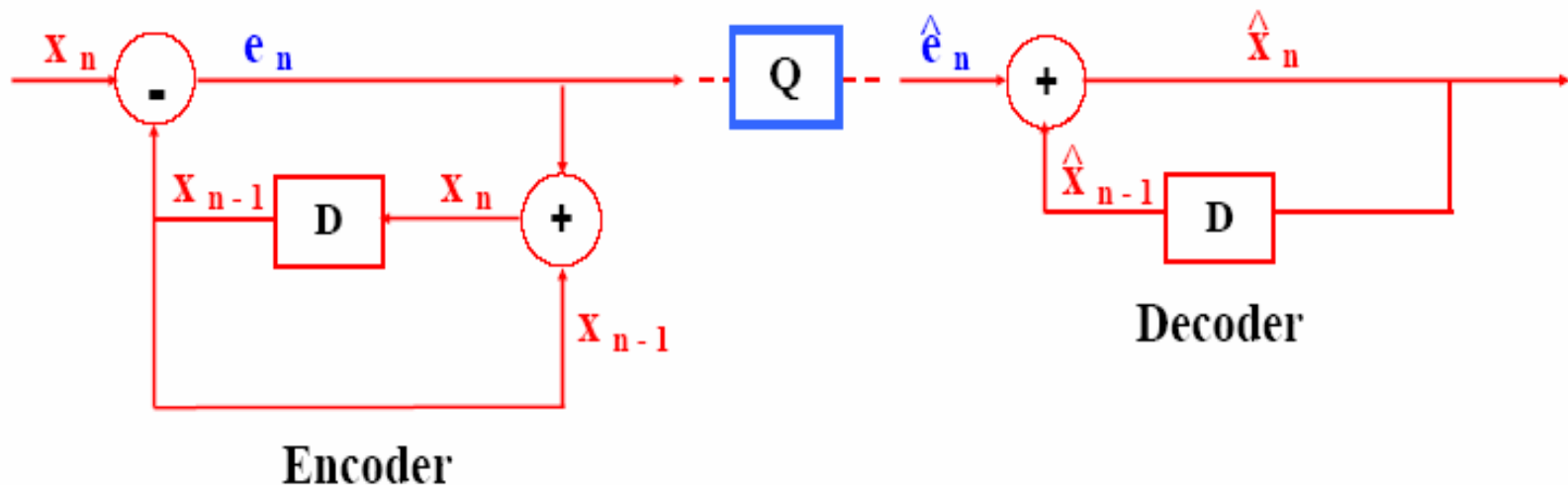
- **Decoding**

$$e_1 \ e_2 \ \dots \ \dots \ e_N \longrightarrow x_1 \ x_2 \ \dots \ \dots \ x_N$$

$$x_1 = e_1 \quad x_n = e_n + x_{n-1}, \quad n=2, \dots, N$$

# Prediction Meets Quantization

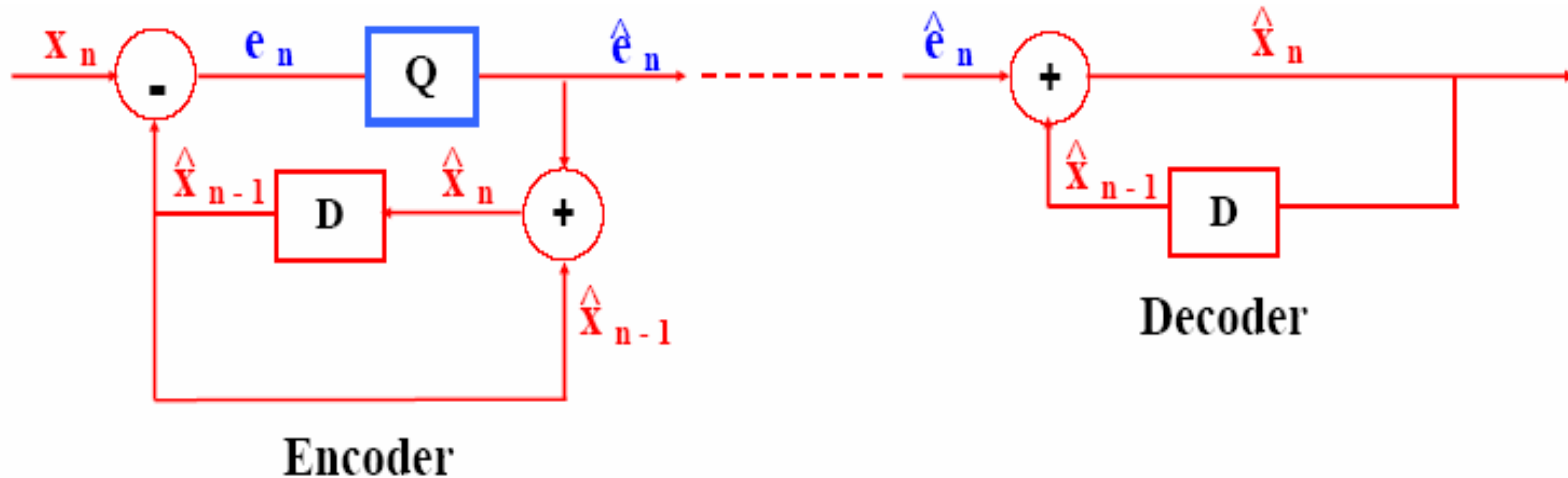
- Open-Loop DPCM



- Quantization is located **outside** the DPCM loop
- Prediction is based on the past unquantized sample

# Prediction Meets Quantization

- **Closed-Loop DPCM**

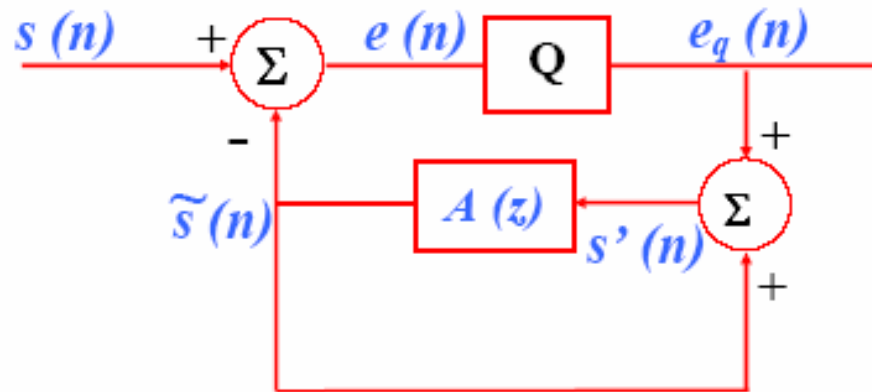


$X_n, e_n$ : unquantized samples and prediction residues

$\hat{X}_n, \hat{e}_n$ : decoded samples and quantized prediction residues

- Quantization is located **inside** the DPCM loop
- Prediction is based on the past decoded sample

# DPCM Encoder



**Encoder**

$$A(z) = \sum_{i=1}^P a_i z^{-i}$$

Digital Filter operating as a Linear Filter

$$\check{S}(n) = \sum_{i=1}^P a_i S'(n-i)$$

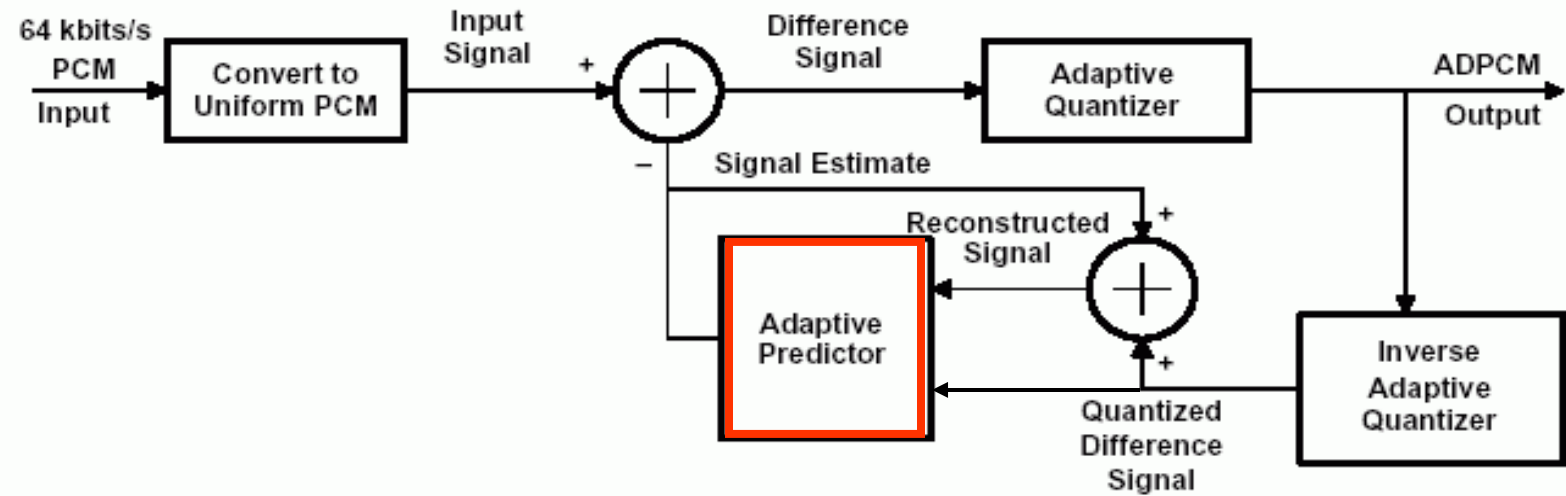
Current Sample is predicted by a linear combination of 'P' past samples



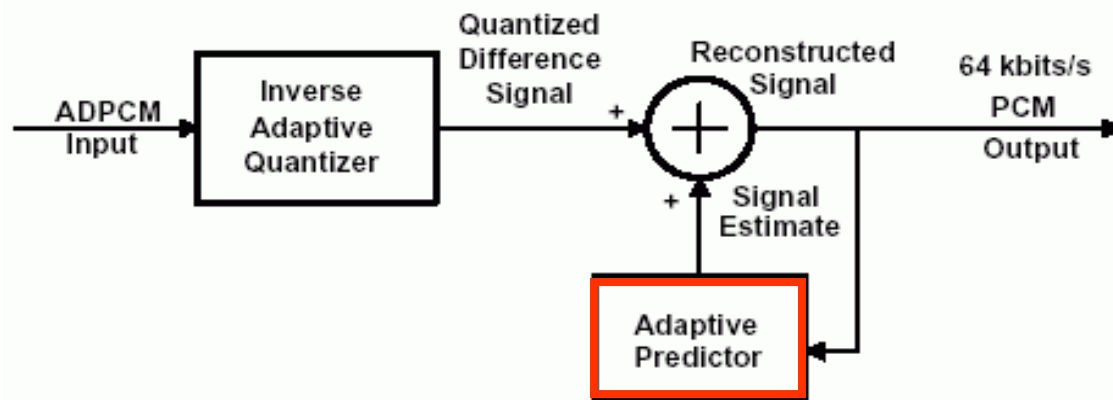
# G.726A Algorithm

- **Feedback-based ADPCM technique**
- **Two Adaptive Predictor Structures used**
  - Sixth order section that models Zeros
  - Second order section that models Poles
- **Adaptive Quantizer used**
  - 31, 15, 7 or 4 quantization levels (5, 4, 3 or 2 bits per sample)
  - Standardized lookup table

# ADPCM (G726A)



a) Encoder



b) Decoder

# G.726A Speech Coding Library Overview

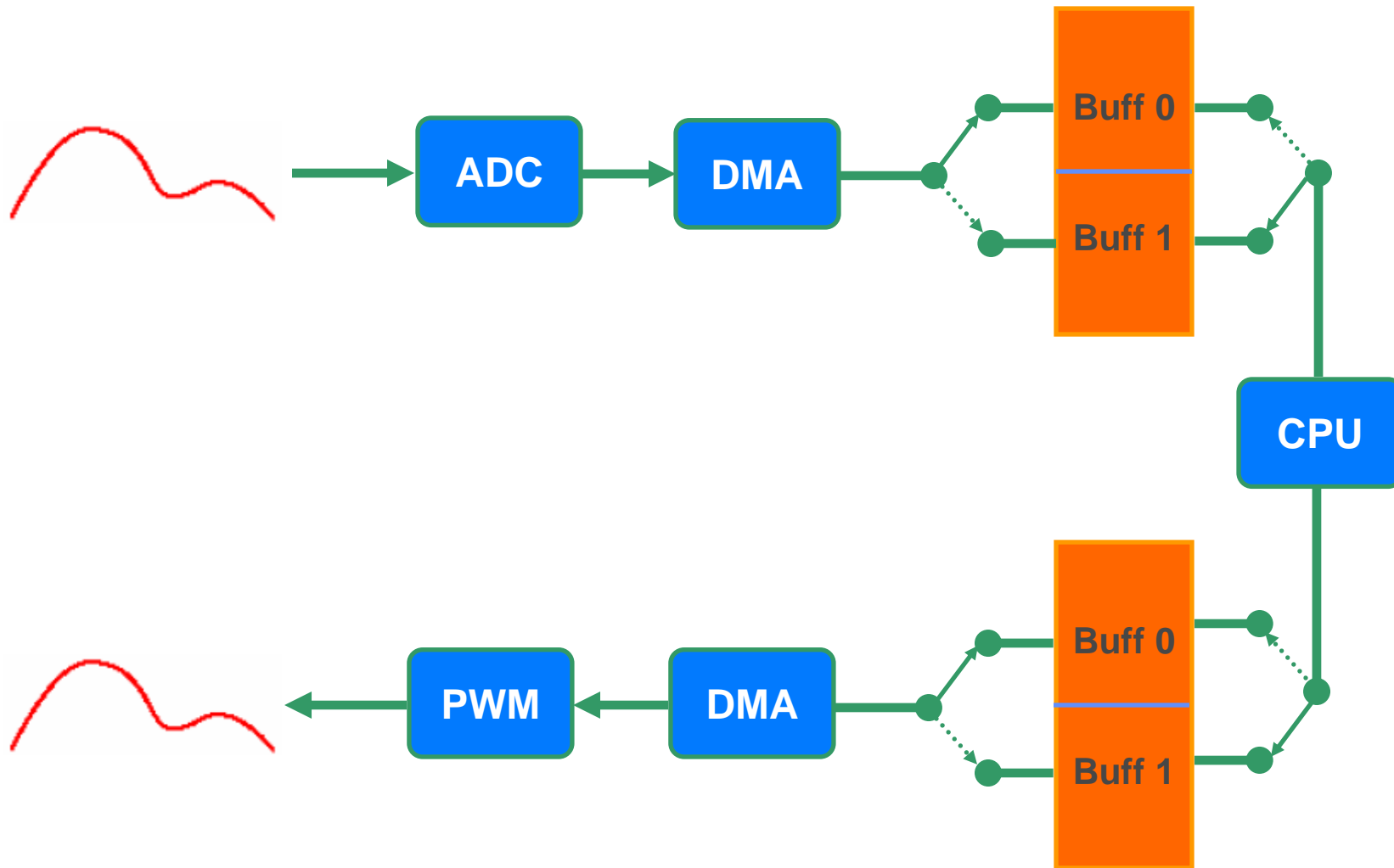
- **Encoder**

- Compression ratio: 3.2:1, 4:1, 5.33:1 or 8:1
- Speech input: 8 kHz, 16-bit mono
- Encoded output: 40/32/24/16 kbps

- **Decoder**

- Decoder input: 40/32/24/16 kbps
- Speech output: 8 kHz, 16-bit mono

# Loopback with ADC/PWM



# Data Structures

- Structure “*sADCChannelHandle*” contains user-defined speech buffer pointers and synchronization flags

## **sADCChannelHandle Structure:**

```
– typedef struct sADCChannelHandle {  
    int * buffer1;  
    int * buffer2;  
    volatile int bufferIndicator;  
    volatile int isReadBusy;  
} ADCChannelHandle;
```

# Encoder Data Buffers

- Encoder Data Buffer usage

Buffer	Frame 0 (32 msec)	Frame 1 (32 msec)	Frame 2 (32 msec)	Frame 3 (32 msec)
Buffer1	Filled by DMA	Processed by library	Filled by DMA	Processed by library
Buffer2	Idle	Filled by DMA	Processed by library	Filled by DMA

# Alternate Sampling Interface

- **On-chip ADC**
  - Reduces system cost
  - 12-bit dynamic range
  - Good intelligibility
  
- **Initialization**
  - ***ADCChannellnit ( )***
    - **Initializes DMA0 ,12-bit ADC & Timer3 modules.**
  
    - ***ADCChannellnit (ADCChannelHandle \* pHandle,int \* pBufferInDMA);***

# Alternate Sampling Interface (*cont....*)

## – *ADCChannelStart( )*

- Enables the ADC, DMA0 and Timer3 modules.
- *ADCChannelStart* (*ADCChannelHandle* \* *pHandle*);
- *AD1CON1bits.ADON = 0x01; //Enable A/D*  
*//converter module*

## ● Synchronization

## – *ADCChannellsBusy ( )*

- Polls the *isReadBusy* flag to check if a new frame is available
- *ADCChannellsBusy*(*ADCChannelHandle* \* *pHandle*);



# G726A Encoder API

- The G726A Encoder is initialized by calling the *G726\_encoder\_init()* function with the desired input bit-rate and the address of the instantiation of *G726\_state* structure.

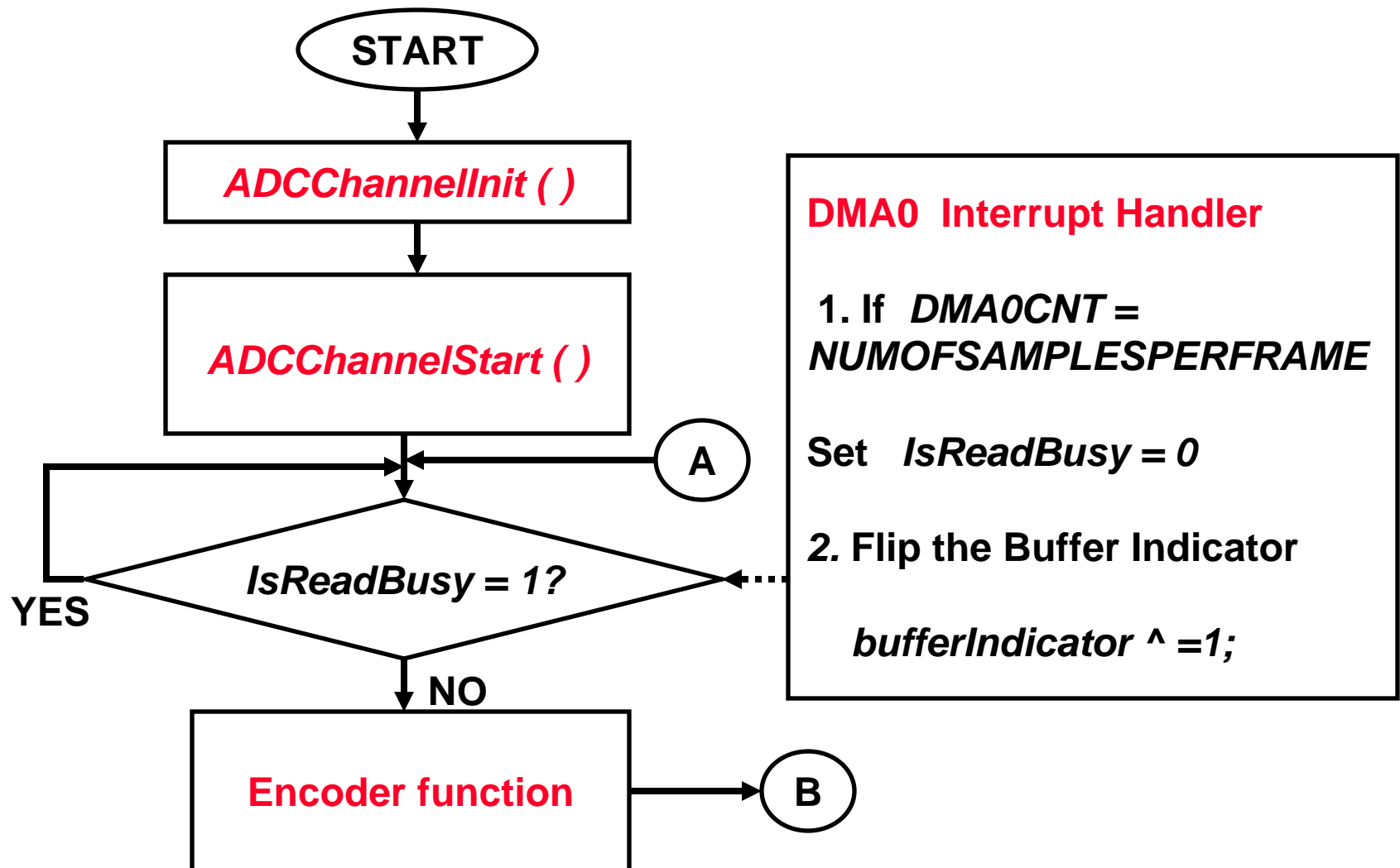
```
G726_state encoder_state;
```

```
G726_encoder_init ( &encoder_state, codecdata.rate );
```

- Speech Encoding is performed by the *G726\_encode()* function.

```
G726_encode( codecdata.sampleOpBuffer,  
             codecdata.sampleEncodeIpBuffer,  
             slen, codecdata.rate, &encoder_state );
```

# Encoder Application Flowchart



# Data Structures

- Structure “*sOCPWMHandle*” contains user-defined speech buffer pointers and synchronization flags

## **sOCPWMHandle Structure:**

```
– typedef struct sOCPWMHandle {  
    int * buffer1;  
    int * buffer2;  
    volatile int bufferIndicator;  
    volatile int isWriteBusy;  
}OCPWMHandle;
```

# Alternate Playback Interface

- **On-chip Output Compare**

- Reduces system cost
- Pulse-Width Modulation (PWM) mode

- **Intialization**

- ***OCPWMInit( )***

- Initializes **DMA1** and **Timer2** module.

- ***void OCPWMInit(OCPWMHandle \* pHandle,int \* pBufferInDMA);***

# Alternate Playback Interface (*cont....*)

## – *OCPWMStart( )*

- Enables **DMA1**, **Output compare** and **Timer2** module.
- *void OCPWMStart(OCPWMHandle \* pHandle);*
- *OC1CON= OCCON\_WORD; /\* Turn module on \*/*

## ● Synchronization

## – *OCPWMIsBusy ( )*

- Polls the *isWriteBusy* flag to check if a new frame is available
- *OCPWMIsBusy(OCPWMHandle \* pHandle);*

# G726A Decoder API

- The G726A Decoder is initialized by calling the *G726\_decoder\_init()* function with the desired input bit-rate and the address of the instantiation of *G726\_state* structure.

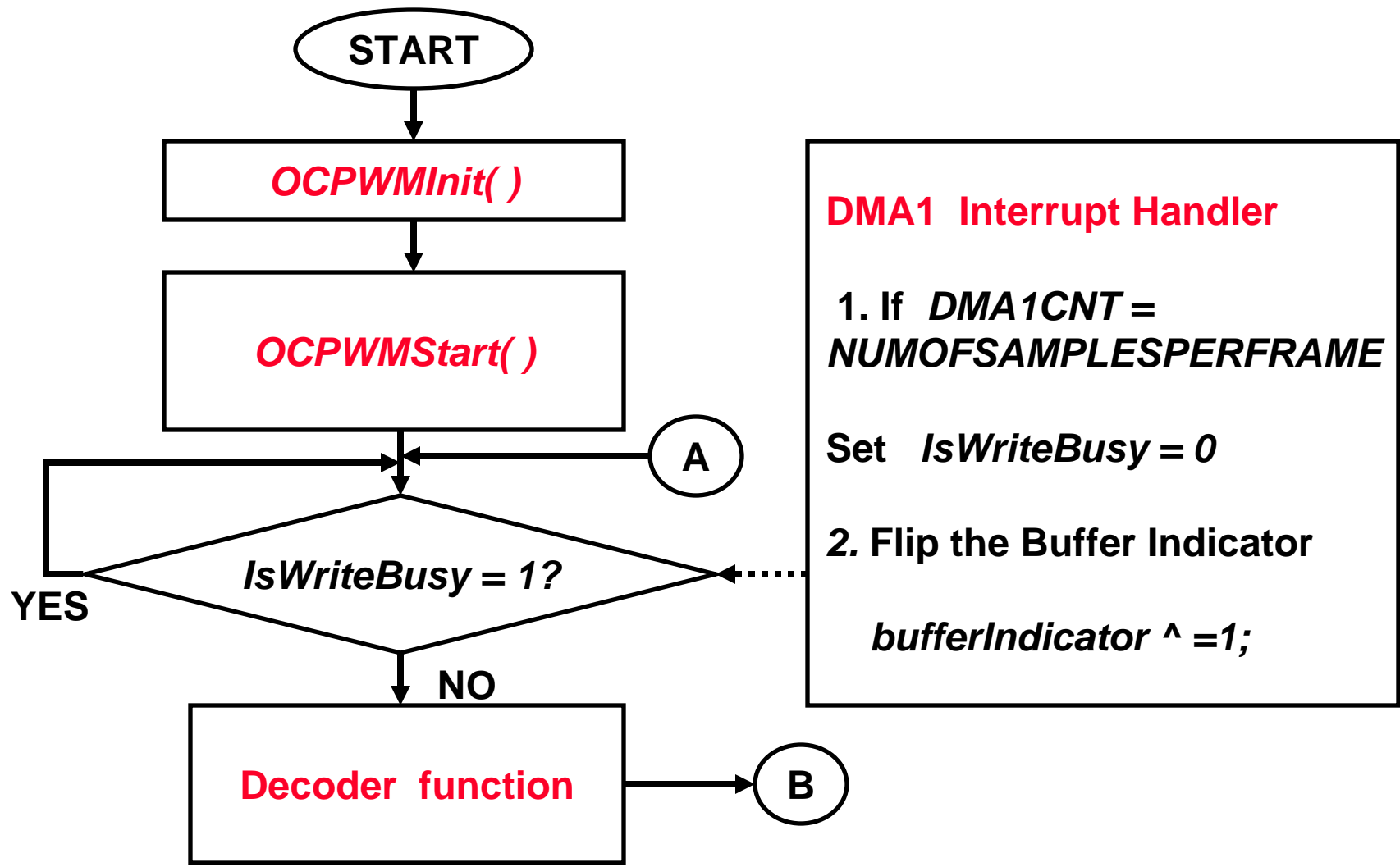
```
G726_state decoder_state;
```

```
G726_decoder_init ( &decoder_state, codecdata.rate );
```

- *G726\_decode()* function is called to perform decoding.

```
G726_decode ( codecdata.sampleEncodeOpBuffer,  
              codecdata.sampleDecodeIpBuffer,  
              slen, codecdata.rate, &decoder_state );
```

# Decoder Application Flowchart

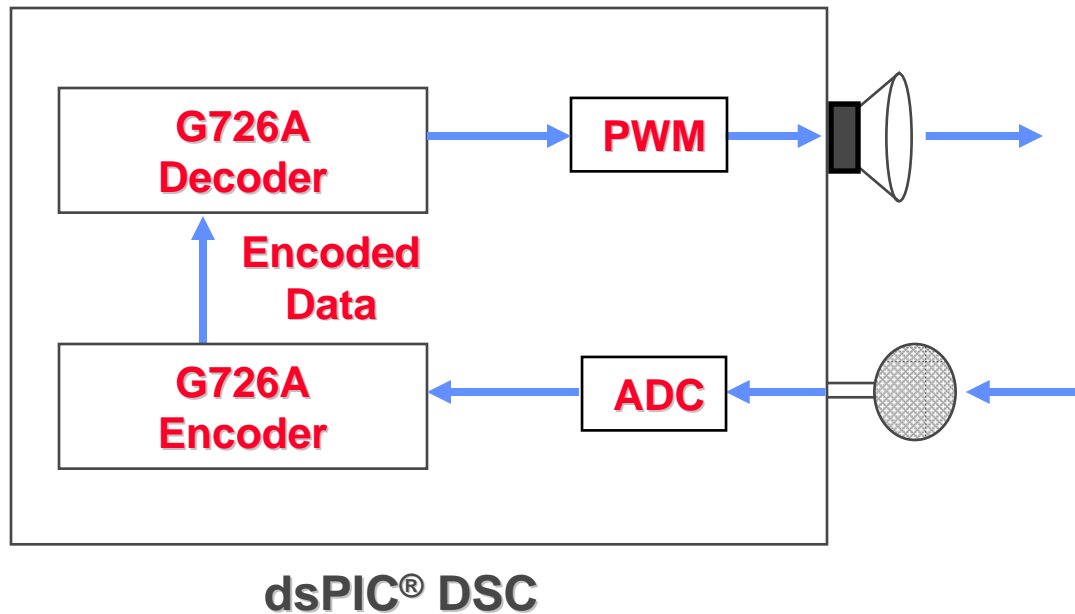


# Hands-on LAB #1

## Loopback with ADC/PWM using G.726A



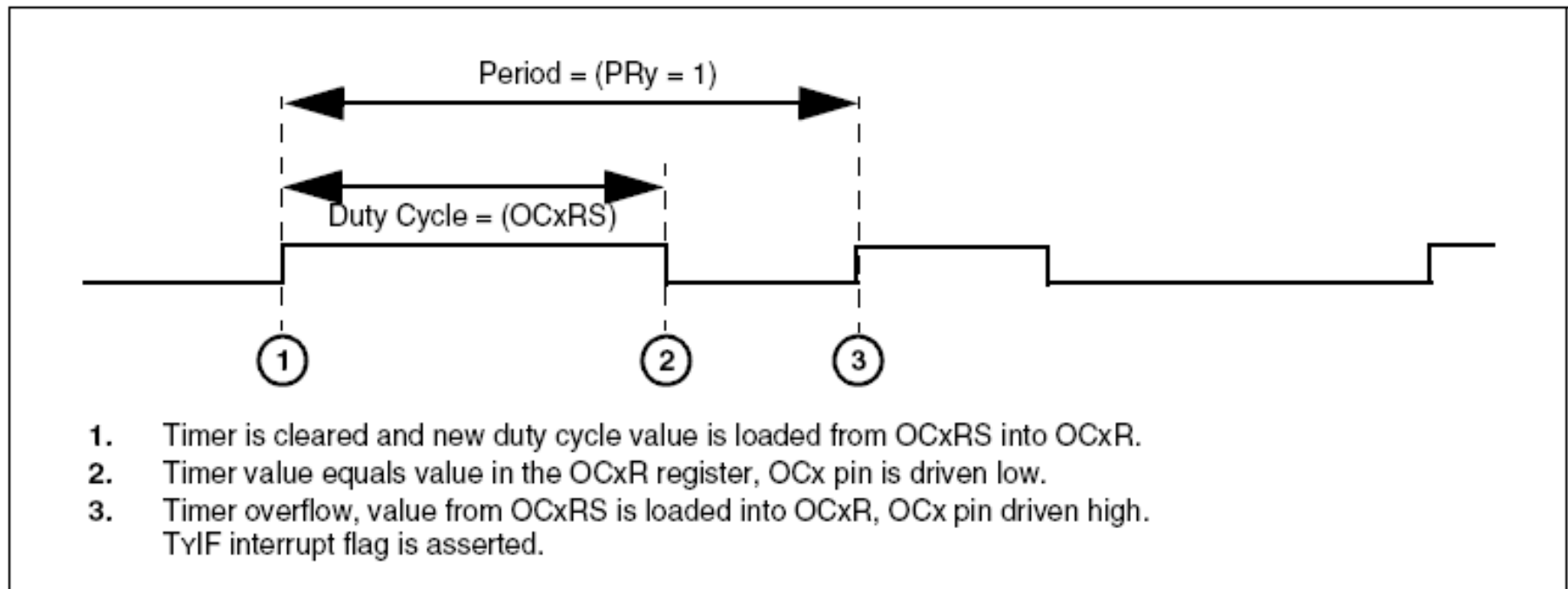
# G726A Loopback Demo with ADC/PWM



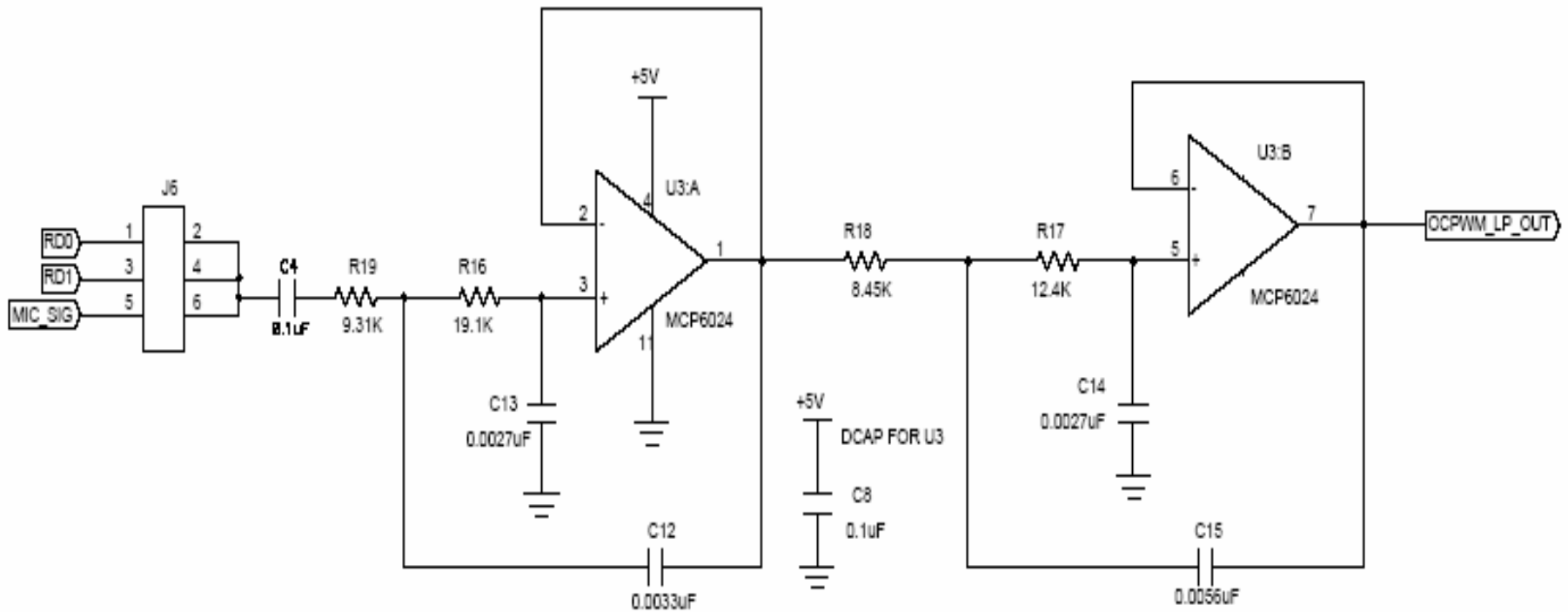
- **Microphone input:**
  - ADC
- **Speaker output:**
  - PWM



# PWM Timing Diagram



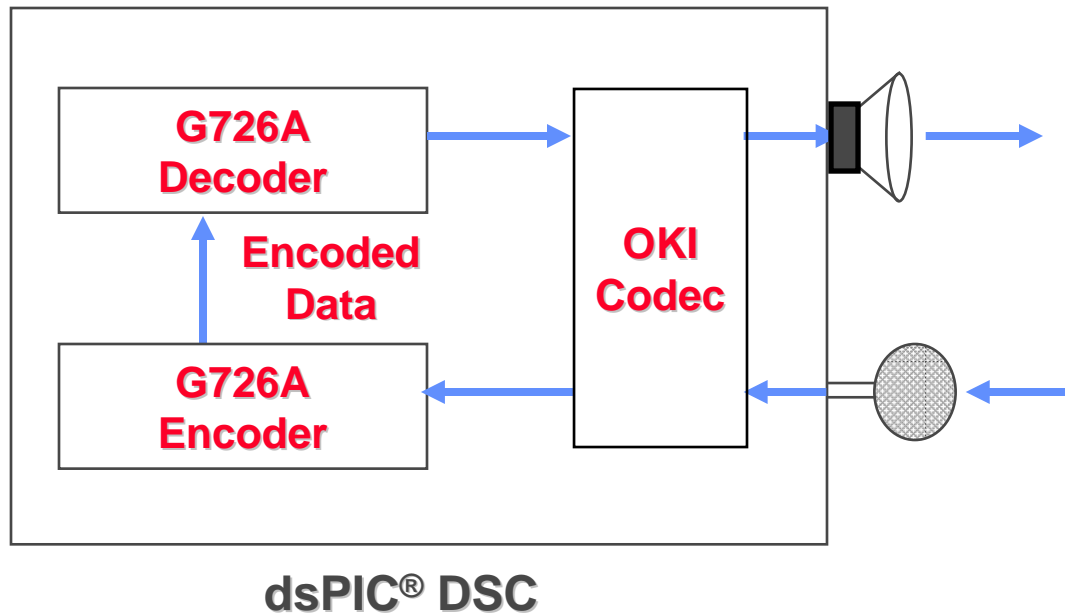
# OC PWM Filter



# Hands-on LAB #2

## Loopback with dsSPEAK™ Speech Processing Reference Design Using G.726A

# G726A Loopback Demo with dsSPEAK™ Speech Processing Reference Design



- **Microphone input:**
  - OKI codec (via DCI)
- **Speaker output:**
  - OKI codec (via DCI)

# Features of dsSPEAK™ Speech Processing Reference Design

- **16-bit dual channel voice band codec (OKI MSM7704-01)**
- **On board voice band filters**
- **5 W audio amplifier for external speaker**
- **Full duplex audio channel for connecting external audio signal**

# Speech Coding Solution #2

## G.711 Speech Coding Library



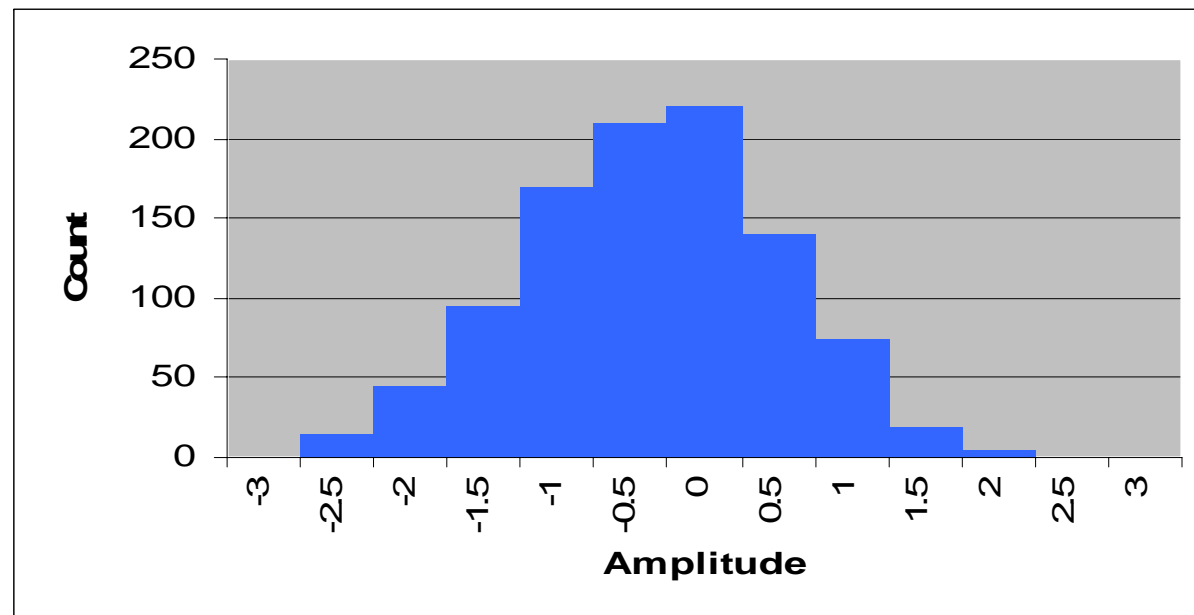
# Overview

- **Based on A-law and  $\mu$ -law companding**
  - ITU-T standard, **but no royalties!!**
  - Standard method for telephony: A-law for Europe,  $\mu$ -law for USA
  - Can interface with A-law/ $\mu$ -law codecs
- **MOS: 4.3 – 4.5**

# G.711

## ● Motivation

- Speech signals have the characteristic that small-amplitude samples occur more frequently than large-amplitude ones
- The Amplitude Distribution is non-uniform



# G.711

- **Basic idea:** assign smaller quantization step size for small-amplitude regions and larger quantization step size for large-amplitude regions
- **Two types of nonlinear compressing functions**
  - **Mu-law** adopted by North American telecommunications systems
  - **A-law** adopted by European telecommunications systems

# G.711 Suite - Alaw

- **A-law Equation**

$$\text{sign}(x) * A |x| / (1 + \ln(A)), \quad \text{for } 0 < |x| < 1/A$$

$$\text{sign}(x) * (1 + \ln(A |x|)) / (1 + \ln(A)), \quad \text{when } 1/A \leq |x| \leq 1$$

Typical value for A is **87.56**

**Note:** *A-law is a combination of logarithmic curve for large amplitudes and linear curve for small amplitudes.*

# G.711 Suite - $\mu$ law

- $\mu$ -law Equation

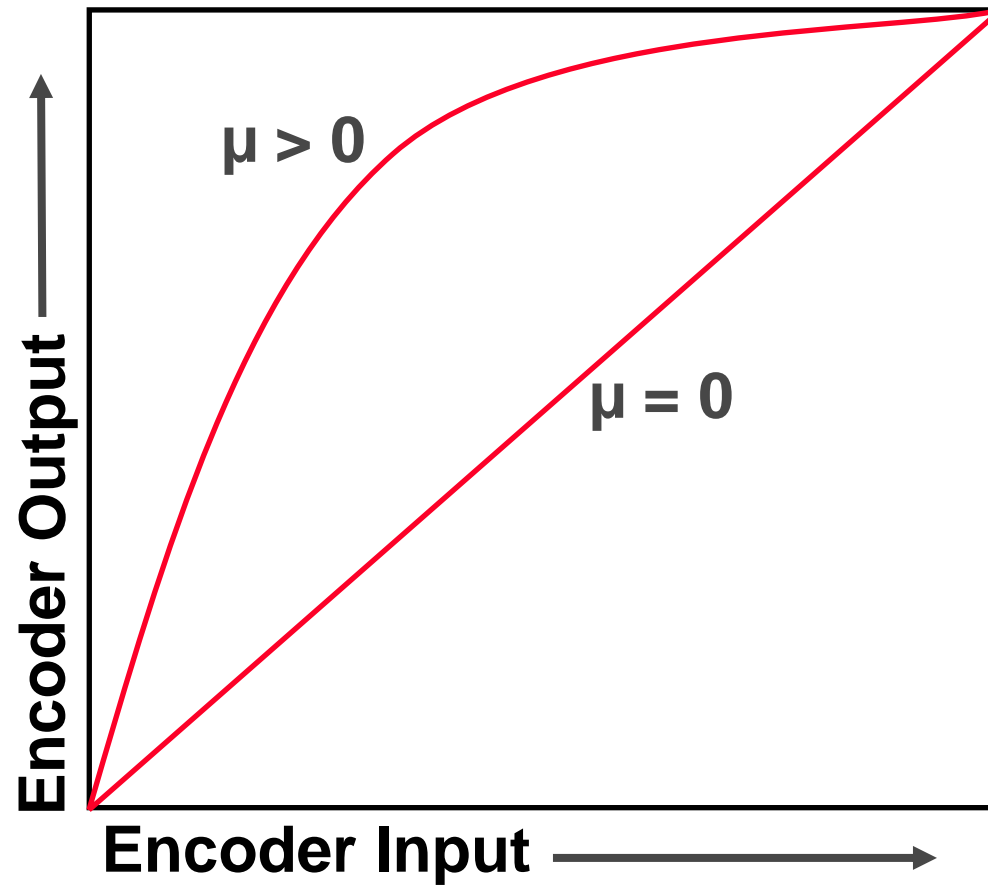
$$\text{sign}(x) * \ln(1 + \mu|x|) / \ln(1 + (\mu))$$

Typical value for  $\mu$  is **256**

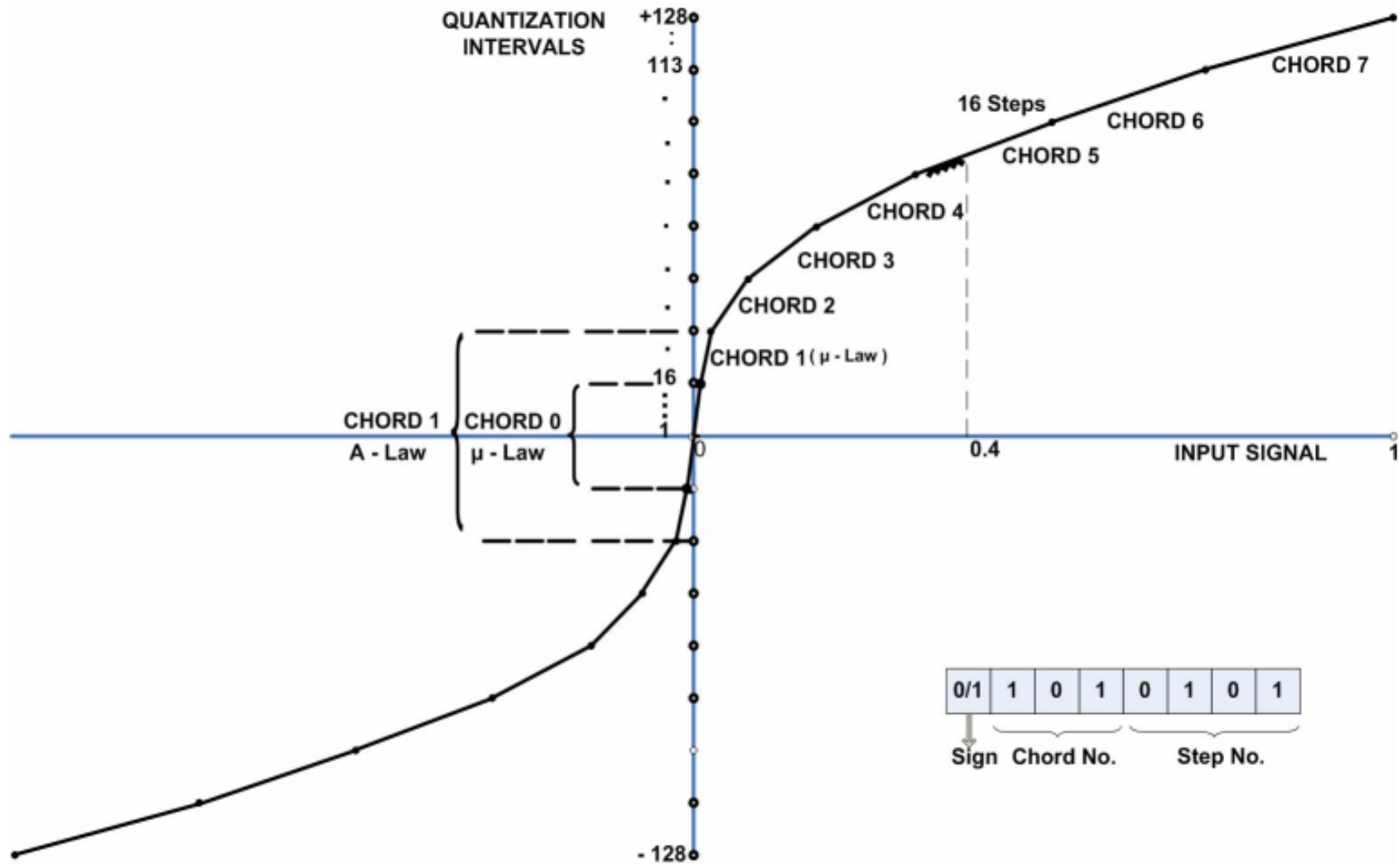
**Note:**  $\mu$ -law is not exactly logarithmic or linear in any range

*but is approximately linear for small amplitudes and logarithmic for large amplitudes.*

# G.711 Algorithm



# Piece-wise linear approximation



# Piece-wise linear approximation

- **$\mu$ -law** and **A law** use 8 linear segments on either side of analog zero
- **$\mu$  law** consists of 15 segment with two inner most segments with identical slope
- **A-law** Consists of 13 segments with four innermost segments having identical slope



# G.711 Speech Coding Library

- **Encoder – 2:1 compression ratio**
  - Speech input: 8 kHz, 16-bit mono
  - Encoded output: 64 kbps
  
- **Decoder**
  - Decoder input: 64 kbps
  - Speech output: 8 kHz, 16-bit mono

# Alternate Playback Interface

- **Intialization**

- ***OCPWMInit( )***

- Initializes **Timer2** module.

- ***OCPWMInit ( );***

- ***OCPWMStart( )***

- Enables **Output compare** and **Timer2** module.

- ***OCPWMStart( );***

# Alternate Playback Interface

- **Synchronization**

- ***OCPWMIsBusy ( )***

- Polls the *isWriteBusy* flag to check if a new frame is available
    - ***OCPWMIsBusy( );***

# G.711 Library API

- **A-law API**

Encode:

***G711Lin2Alaw(int \* input, char \* output, int size)***

Decode:

***G711Alaw2Lin(char \* input, int \* output, int size)***

- **Mu-law API**

Encode:

***G711Lin2Ulaw(int \* input, char \* output, int size)***

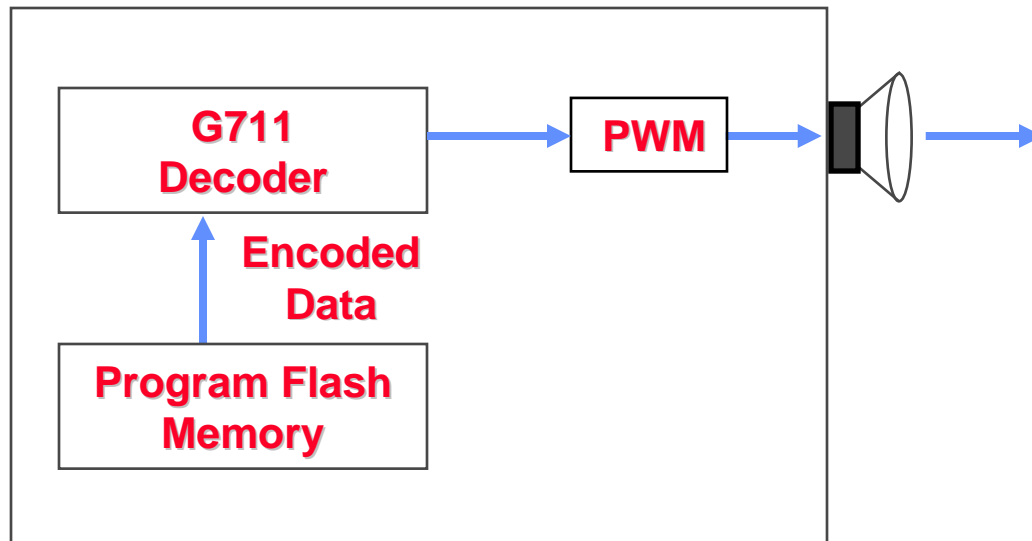
Decode:

***G711Ulaw2Lin(char \* input, int \* output, int size)***

# Hands-on LAB #3

## Playback with PWM using G.711

# Playback Demo PWM

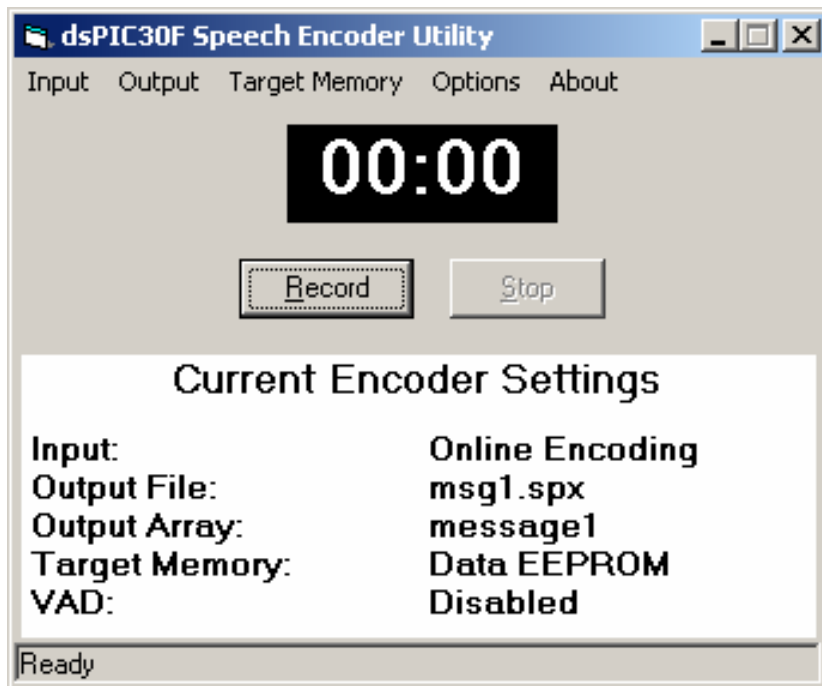


dsPIC® DSC

- **Decoder input memory source:**
  - Program Flash Memory
- **Speaker output:**
  - PWM

# Speech Encoding Utility

- Used mainly for making “playback” files
- Generates an encoded data file from PC microphone (live input) or WAV file
- User links encoded file into application project



- **Target memory for storing encoded speech array**
  - Program Flash
  - Data EEPROM
  - External Flash
  - RAM

# Summary

- dsPIC<sup>®</sup> DSC Speech Processing libraries provide computationally efficient software solutions for a wide range of speech and telephony applications
- Self-contained software packages with pre-compiled library functions and demo application
- Easy-to-use Application Programming Interface for integration into user application



# Dev Tools used in this class

- **Development Boards**
- **G726A Loopback demo with ADC/PWM**
  - Explorer 16 Board (DM240001)
  - Audio PICtail™ Plus Daughter Board (AC164129)
- **G726A Loopback demo with dsSPEAK™ Speech Processing Reference Design**
  - dsSPEAK Speech Processing Reference Design Board (DM300025)
- **G711 Playback demo with ADC/PWM**
  - Explorer 16 Board (DM240001)
  - Audio PICtail Plus Daughter Board (AC164129)
- **Development Tools**
  - MPLAB® ICD 2 In-Circuit Debugger **OR**
  - MPLAB REAL ICE™ In-Circuit Emulator

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