Digital Communications Engineering 1

(COMM2108)

Bandwidth Efficient Schemes

Bandwidth Efficient Schemes

- The bandwidth (BW) of a digital signal depends on the bit rate and the pulse shape used.
- The *dimensionality theorem* says that the bandwidth of a digital signal is bounded by

$$BW \geq \tfrac{1}{2} R_b$$

- where R_b is the bit rate.
- Note that this is a lower bound, in general the bandwidth will be very much greater.

Bandwidth Efficient Schemes

- Considering the case of PCM, the bit rate

$$R_b = nf_s$$

- where n is the number of bits in the code word and f_s is the sampling rate.
- To avoid aliasing

$$f_s \geq 2B$$

- where B is the bandwidth of the baseband analogue waveform,

$$R_b \geq 2nB$$

Bandwidth Efficient Schemes

- The resulting PCM signal will have a bandwidth

$$BW_{PCM} \geq nB$$

- Since typically n = 8,12,14 the bandwidth of the PCM signal will be considerably larger than that of the original analogue baseband signal.
- However, PCM is widely used despite the usual concerns about conserving bandwidth as the advantages of the digital format far outweigh the bandwidth penalty.

Bandwidth Efficient Schemes

- In many situations it is desirable to reduce the bandwidth occupied by a PCM signal, as spectrum is a limited and therefore valuable resource, especially for radio transmission systems.
- For example, the standard *A-law* companded telephony generates a *64 kbps* bit stream in order to digitally encode toll quality speech for telephony applications.
- However, this bit rate (and its associated transmission bandwidth) is far greater than needs be.
- Consider the case of the GSM cellular telephony standard where a special low data rate speech encoding technique is used to digitise speech into a *13 kbps* bits stream.

Bandwidth Efficient Schemes

- The key strategy is low data rate speech encoding is to remove *redundant* information from the speech waveform.
- Redundant information is unnecessary or superfluous information that may be removed from a waveform and still not adversely affect its intelligibility.
- By removing redundant information, the bit rate can be reduced and hence the bandwidth occupied by signal can also be reduced.
- This allows for more users to be accommodated within a given spectrum allocation which leads to an increased system capacity.

Bandwidth Efficient Schemes

- There is a high degree of *correlation* between closely spaced samples of human speech.
- Human speech is said to contain *redundant* information, i.e. the same or very similar information tends to reside in two or more adjacent samples.
- Consequently, valuable information is wasted in transmitting this unnecessary information.
- Moreover, if this redundant information could be removed from the signal, a more spectrally efficient signal should result.

Bandwidth Efficient Schemes

- Human speech contains significant redundancy which means that the changes in signal amplitude between adjacent sample values will be quite small.
- This inter-sample dependence can be exploited to remove the redundancy from speech waveforms.
- There are a number of techniques, known as *differential techniques*, that deliberately exploit this feature in order to realise a more efficient representation of the speech waveform.

Bandwidth Efficient Schemes

- The main <u>differential</u> techniques include:
  - Delta PCM
  - Differential PCM (DPCM)
  - Adaptive Differential PCM (ADPCM)
  - Delta Modulation (DM)
  - Adaptive Delta Modulation (ADM)
- The basic concept behind these differential techniques is to transmit information about the <u>changes</u> in sample value rather than the actual sample values themselves.

Bandwidth Efficient Schemes

- In *Delta PCM*, at the transmitter the difference between adjacent sample values is generated and encoded using PCM but with a reduced code word.
- Since the differences between sample values will be significantly less than the actual sample values, these differences can be encoded using fewer bits than would be the case with conventional PCM.
- At the receiver, a “summing” operation is implemented to reconstruct the sample values.
- The main disadvantage with this technique is that it cannot accommodate rapidly varying waveforms.

Bandwidth Efficient Schemes

- In *differential PCM (DPCM)* another feature of redundancy is exploited, namely that future sample values can be predicted (within certain confidence limits) from the current and past sample values.
- This technique is known as *linear prediction* where the value of next sample is inferred or estimated from past sample values.
- DPCM uses an algorithm called a *predictor* to predict the next sample value.
- In DPCM, it is the <u>difference</u> between the actual sample value and the predicted sample value that is transmitted.

Bandwidth Efficient Schemes

- If *g(t)* represents the input analogue waveform, then $g(kT_s)$ represents the sampled version of *g(t)* where *k* is the sample number and T_s is the sampling interval.
- The prediction or estimate of next sample value (generated by the predictor algorithm) is denoted $g'(kT_s)$.
- The error $\varepsilon(kT_s)$ between the actual sample value and the predicted sample value is calculated by the *predict-and-compare loop* to give

$$\varepsilon(kT_s) = g(kT_s) - g'(kT_s)$$

- This known as the *prediction error* or the *residue*.

Bandwidth Efficient Schemes

- The prediction error is then quantised,

$$\varepsilon_q(kT_s) = \text{quant}\,[\varepsilon(kT_s)]$$

- The quantised prediction error $\varepsilon_q(kT_s)$ in then encoded and <u>transmitted</u>.
- The quantised prediction error $\varepsilon_q(kT_s)$ is also used to correct the predicted sample value to give

$$g(kT_s) = g'(kT_s) + \varepsilon_q(kT_s)$$

- This is performed by the *predict-and-correct* loop. The corrected and quantised sample value is used by the predictor in forming the estimate for the next sample value.

Bandwidth Efficient Schemes

- At the receiver, an identical predictor is used to reconstruct the sample values using

$$g(kT_s) = g'(kT_s) + \varepsilon_q(kT_s)$$

- where *$g'(kT_s)$* has been generated by the predictor at the receiver and *$\varepsilon_q(kT_s)$* is the received quantised prediction error.
- Finally, as in all sampled systems, a reconstruction filter is used to recover the analogue waveform *$g(t)$* from the recovered sampled waveform *$g(kT_s)$.*
- These coders are known as *predictor-corrector* coders.

Bandwidth Efficient Schemes

- The predictor values is often calculated as a linear combination of the previous *N* sample values..
- In practice, this would be implemented as a transversal digital filter which generates the next sample value based upon a weighted sum of of the filters taps.
- *Adaptive Differential PCM (ADPCM)* is a more sophisticated version of DPCM where the predictor coefficients $C_0, C_1, \ldots, C_N$ (i.e. the weighting factors applied to the filter taps) are continuously modified or adapted to suit the changing signal characteristics.

Bandwidth Efficient Schemes

- The ITU-T has adopted an ADPCM technique which converts standard 64 kbps companded PCM (G.711) into a 32 kbps ADPCM signal (G.721).
- The G.721 encoder uses a (15 level) 4-bit code word to transmit the quantised prediction error.
- The subjective speech quality of error-free 32 kbps ADPCM is only slightly inferior to 64 kbps standard PCM.
- For bit error probabilities $>10^{-4}$, the subjective speech quality is better than standard PCM (i.e. it's more rugged).
- Other ADPCM standards from the ITU-T include the G.726 (16 kbps) and G.727 (40kbps) recommendations.

Bandwidth Efficient Schemes

- *Delta modulation (DM)* is a simplified form of DPCM that uses a very simple one-tap predictor.
- The predictor algorithm operates by assuming that the next sample value will be same as the current sample value, i.e.

$$g(kT_s) = g((k-1)T_s)$$

- This can be implemented as a one-sample delay element.
- The one-bit quantiser is essentially a comparator which reports on whether the predicted sample value is less than or greater than the actual sample value.

Bandwidth Efficient Schemes

- Delta modulation generates a staircase approximation to the input analogue waveform, i.e. at each time interval the quantised waveform amplitude can increase or decrease by a fixed amount Δ.
- The prediction error is given by

$$\varepsilon(kT_s) = g(kT_s) - g'(kT_s)$$

- The quantised prediction error is

$$\begin{aligned} \varepsilon_q(kT_s) &= \mathrm{quant}\,[\varepsilon(kT_s)] \\ &= \Delta\,\mathrm{sgn}[\,\varepsilon(kT_s)] \end{aligned}$$

Bandwidth Efficient Schemes

- At the receiver, the reconstructed sampled waveform

$$g(kT_s) = g'(kT_s) + \varepsilon_q(kT_s)$$

$$= g((k-1)T_s) \pm \Delta$$

- A reconstruction filter recovers the analogue waveform *g(t)* from the reconstructed sampled waveform $g(kT_s)$.
- Delta modulation as two unique features:
  – A 1-bit word which eliminates the need for word framing.
  – Simplicity in terms of hardware and algorithm structure.

Bandwidth Efficient Schemes

- Unlike PCM, there are two types of quantisation error that can occur in DM:
  - Slope overload distortion.
  - Granular (or idling) noise.
- *Slope overload distortion* occurs when the input analogue waveform *g(t)* changes too rapidly for the staircase approximation $g(kT_s)$ to follow.
- In order to avoid slope overload distortion, we need to ensure that

$$\frac{\Delta}{T_s} \geq \max\left|\frac{dg(t)}{dt}\right|$$

Bandwidth Efficient Schemes

- Slope overload distortion can be avoided by using a large value for Δ or a small value for T_s (i.e. by using a large sampling rate or oversampling).
- In contrast *granular noise* occurs when the step size Δ is too large relative to the local slope of the input waveform causing the staircase approximation to "hunt around" a locally flat segment of the input waveform.
- The resulting waveform is a square wave with a period half that of the sampling period.
- Granular noise is also known as *idling noise*.

Bandwidth Efficient Schemes

- As regards the choice of step size Δ there are conflicting requirements for acceptable granular noise and avoiding slope overload distortion.
- One could choose to use a small step size Δ and compensate by using a small T_s (i.e. sample much faster than the Nyquist rate), however with this approach all of the bandwidth saving would be lost.
- The choice of step size Δ is therefore a compromise between avoiding slope overload distortion and minimising the granular noise.

Bandwidth Efficient Schemes

- Assume we have a sinusoidal input signal waveform given by

$$g(t) = A\cos(2\pi f_o t)$$

- The maximum slope of the signal *g(t)* is given by

$$\frac{dg(t)}{dt} = -2\pi f_0 A\sin(2\pi f_o t)$$

$$\max\left|\frac{dg(t)}{dt}\right| = 2\pi f_0 A$$

Bandwidth Efficient Schemes

- To avoid slope overload distortion we require that

$$\frac{\Delta}{T_s} \geq 2\pi f_0 A$$

- This imposes a limit on the amplitude of the input signal

$$A \leq \frac{\Delta}{2\pi f_0 T_s}$$

- Therefore the maximum permissible signal power is

$$S = \tfrac{1}{2} A^2$$

$$= \frac{\Delta^2}{8\pi^2 f_0^2 T_s^2}$$

Bandwidth Efficient Schemes

- Using an earlier result from PCM systems, the noise power due to the quantisation process is

$$N_0 = \frac{\Delta V^2}{12}$$

- where ΔV = the quantisation interval.
- In the delta modulation system, the quantisation interval is

$$\Delta V = 2\Delta$$

$$N_0 = \frac{\Delta^2}{3}$$

Bandwidth Efficient Schemes

- The signal to noise ratio (*SNR*) before the reconstruction filter is

$$SNR = \frac{3}{8\pi^2}\left(\frac{f_s}{f_0}\right)^2$$

- where $f_s = 1/T_s$ is the sampling frequency.
- Next we need to consider the *SNR* after the reconstruction filter.
- Assuming that the average noise power is uniformly distributed over the frequency interval between $-f_s$ and $+f_s$.
- Also assuming that the reconstruction filter is a LPF with a bandwidth $W << f_s$.

Bandwidth Efficient Schemes

- The average noise power is

$$N_0 = \left(\frac{W}{f_s}\right)\frac{\Delta^2}{3}$$

- This gives a post reconstruction filter *SNR* of

$$SNR = \frac{3}{8\pi^2 W f_0^2} f_s^3$$

Bandwidth Efficient Schemes

- Under the assumption of no slope overload distortion, the maximum output signal to noise ratio of a delta modulator is proportional to f_s^3, i.e. the *SNR* increases by *9 dB* for every octave increase in f_s.
- With standard PCM, the *SNR* increases exponentially with the bit rate, i.e. it increase by *6 dB* for every additional bit added to the PCM code word.

Bandwidth Efficient Schemes

- In conventional delta modulation, the problem of keeping both quantisation (or qranular) noise and slope overload noise within acceptable levels may be solved by *oversampling*.
- However *oversampling* results in an increased bandwidth which negates the original bandwidth saving.
- An alternative strategy is to make the step size Δ variable,
  - By increasing Δ when slope overload noise would otherwise dominate, and
  - By decreasing Δ when granular noise would otherwise dominate.
- The step size Δ is varied according to the history of ε_q.
- This leads to *adaptive delta modulation (ADM).*

Bandwidth Efficient Schemes

- For example, if $\varepsilon_q = +\Delta$ or $-\Delta$ for several adjacent samples, then g(t) is rising/falling rapidly and therefore Δ needs to be increased in order to avoid slope overload distortion.
- Conversely, if $\varepsilon_q = \pm\Delta$ (i.e. alternates between $+\Delta$ and $-\Delta$), then g(t) is changing slowly and and therefore Δ needs to be decreased in order to minimise granular noise.
- Adaptive delta modulation (ADM) performs between *8* to *14 dB* better than standard DM is terms of *SNR*.
- ADM also has a larger dynamic range and can operate at lower bit rates, typically between *32 kbps* and *16 kbps*.

Bandwidth Efficient Schemes

- By increasing the <u>complexity</u> of the signal processing, it is possible to produce communications quality speech at bit rates significantly less than 64 kbits/sec.
- Speech encoded bit rates as low as 4.8 or 2.4 kbits/sec can be achieved.
- The two main techniques used are:
  – Waveform coding
  – Source coding (i.e. speech analysis and synthesis)

Bandwidth Efficient Schemes

- *Waveform Coding:* Human speech is reduced to a string of bits by operating on the waveform in one of two ways:
  - Differential Coding
  - Frequency Domain Coding

Bandwidth Efficient Schemes

- *Differential coding* exploits the high degree of correlation found between adjacent samples taken from human speech.
- Typical differential coding schemes include:
  - Delta PCM
  - Differential PCM (DPCM)
  - Adaptive Differential PCM (ADPCM)
  - Delta Modulation (DM)
  - Adaptive Delta Modulation (ADM)

Bandwidth Efficient Schemes

- *Frequency Domain Coding* divides the signal into a number of frequency bands and then codes each of the bands separately.
- Because speech energy is not evenly distributed across the frequency spectrum, it is possible to dynamically allocate more bits to those bands in which speech activity is significant at the expense of those bands where speech activity is absent.
- Two such techniques are:
  - Subband Coding (SBC)
  - Adaptive Transform Coding (ATC)

Bandwidth Efficient Schemes

- In *subband coding (SBC),* the input signal frequency band is divided into several subbands (typically 4 to 8 bands) and each subband is encoded using ADPCM.
- SBC is capable of producing communications quality voice encoding at 9.6 kbits/sec.
- For example, the ITU-T G.722 Recommendation for 0-7 kHz high quality speech encoding at 64, 56, and 48 kbits/sec.

Bandwidth Efficient Schemes

- In *adaptive transform coding (ATC),* the input signal is divided into short time blocks (called *frames*) that are transformed into the frequency domain using Fourier based techniques.
- The result is a set of transform coefficients that describe each frame. These coefficients are quantized and transmitted.
- ATC is capable of producing communications quality voice encoding at 7 kbits/sec.
- Typical ATC schemes include:
  - Discrete Fourier Transform (DFT)
  - Discrete Walsh-Hadamard Transform (DWHT)
  - Discrete Cosine Transform (DCT)

Bandwidth Efficient Schemes

- *Waveform Coding* is used to produce an approximation to the input signal and can generally be applied to any signal source.
- *Source Coding* based upon speech analysis and synthesis is very signal specific, i.e. it is designed for speech coding only.
- Exploits the way in which the human ear perceives speech, e.g. the human ear is only sensitive to short term amplitude variations in the speech spectrum and is completely insensitive to phase variations.

Bandwidth Efficient Schemes

- Here speech sounds (known as *phonemes*) are identified and encoded. By transmitting a description of the speech and emulating the mechanism of the human vocal tract in the receiver, it is possible to produce artificial speech at greatly reduced bit rates.
- The basic idea behind analysis/synthesis encoders is to produce a signal that "sounds" like the original signal, rather than "looks" like it.
- Two basic techniques used are:
  - Vocoding
  - Linear Predictive Coding (LPC)

Bandwidth Efficient Schemes

- Vocoding extracts the significant components in a speech waveform in an efficient way and can achieve bit rates as low as 2.4 kbits/sec.
- The are two basic types of vocoder used:
  - Channel vocoder
  - Formant vocoder

Bandwidth Efficient Schemes

- The *channel vocoder* models the speech generation mechanism.
- Speech is produced by forcing air past the larynx to produce sound that is then shaped by the oral and nasal cavities.
- Speech consists of periodic emissions called *voiced sounds* and turbulent noise called *unvoiced sounds.*
- The frequency of the voiced sounds is called *pitch*.
- The oral and nasal cavities can be modelled as a time-varying filter that spectrally shape the signal.

Bandwidth Efficient Schemes

- In the *channel vocoder*, the speech signal is divided into several separate frequency bands and the magnitude of the energy in each band is transmitted together with information on the presence or absence of pitch.
- At the receiver, voiced or unvoiced energy is applied to a set of filters that produce recognisable speech.

Bandwidth Efficient Schemes

- The *formant vocoder* analyses the speech waveform to identify the dominant spectral resonances of the oral cavity, known as *formants*.
- The *formant vocoder* transmits to the receiver the amplitude and spectral position of 3 or 4 of the speech formants.

Bandwidth Efficient Schemes

- *Linear Predictive Coders (LPCs)* are a natural extension of *N*-tap predictive coders.
- Adaptive predictors are used to form estimates to an input speech signal.
- However, instead of using the signal estimate to generate a prediction error, a set of linear prediction coefficients is produced.
- The linear prediction coefficients are transmitted to the receiver where they are used to regenerate the speech sound using an artificial vocal tract.
- LPCs are very effective at realising low bit rate speech encoding .
- NATO LPC vocoder standard (LPC-10) produces a bit rate at 2.4 kbits/sec, however with a low MOS = 2.

Bandwidth Efficient Schemes

- GSM mobile phone system uses LPC to encode speech at a rate of 13 kbits/sec called Regular Pulse Excitation with Long Term Predictive Coding (RPE-LPC).
- The speech waveform is bandlimited to 3.4 KHz and sampled at 8 kHz using a 13-bit linear quantizer producing a 104 kbits/sec bit stream.
- This is reduced to 13 kbits/sec using RPE-LPC where every 20 msec (speech frame), 160 speech sample values are encoded into a block of 260 bits.

Bandwidth Efficient Schemes

- Considerable research is in progress to improve the quality of vocoders.
- However the price to be paid for this reduction in bandwidth is:
  - Increased complexity and hence cost.
  - Increased processing delay.
- For example, consider the *processing delay* for the following:
  - For standard 64 kbits/sec PCM the delay is 125 μsec.
  - For ADPCM the delay is 250 μsec
  - For SBC the delay is 20 msec
  - For ATC the delay is 100 msec
  - For vocoders the delay is 500 msec