RTT TECHNOLOGY TOPIC
April 2002

Channel coding – the 2G to 3G transition

Channel coding has been used in digital cellular handsets (and base stations) for the past ten years as a mechanism for improving transmission quality in a band limited noise limited Rayleigh faded channel.

Channel encoding adds bits **to** the source coded data (see last month's HOT TOPIC) calculated **from** the source coded data. The decoder uses these extra bits to detect/correct errors. Errors are detected when the actual transmitted redundancy value fails to match the redundancy value calculated from the transmitted data.

Two code types are used; **block codes** which segment the message into blocks adding a parity check number which is a product of the information bits contained in the block and **convolutional codes** (also known as 'tree' codes) where the encoder has memory and the output code words depend on the current bit value and adjacent bits held within the register.

Block codes are good for detecting bursty errors, convolutional codes work best with evenly distributed errors. Interleaving is used to help randomise error distribution to ensure convolutional encoders/decoders deliver coding gain. If an error burst lasts longer than the interleaving depth, the convolutional decoder will suffer from 'error extension' - it will make matters worse. This will (hopefully) be detected by the block code parity check. The (voice, image or video) sample can be discarded and the prior sample re-used.

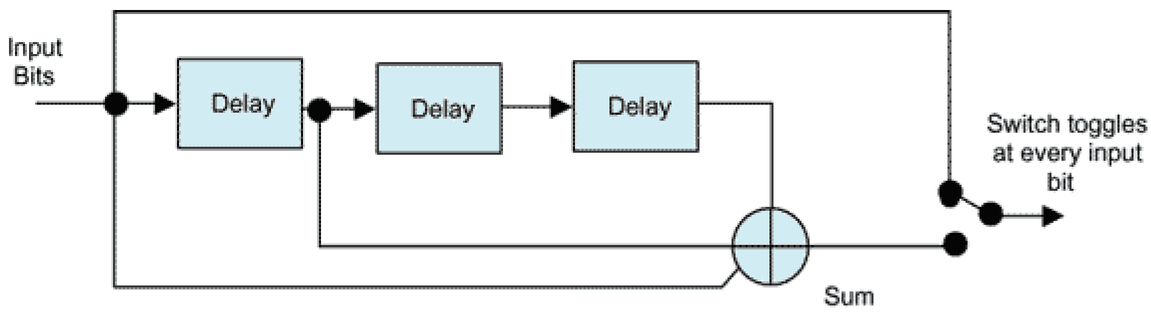

Figure 1 - Simple Convolutional Encoder

Figure 1 shows a simple convolutional encoder. Each time an information bit arrives at the front end of the encoder a branch code word is produced. As the bit moves through the code register it influences other bits entering the decoder.

The objective is to increase the distance between '0's' and '1's'. In the decoder, the information bits are effectively 'weighted' with the code word bits to provide coding gain. These decoders are commonly described as maximum likelihood decoders.

The diagram shows how coding gain is achieved in a GSM vocoder (encoder/decoder).

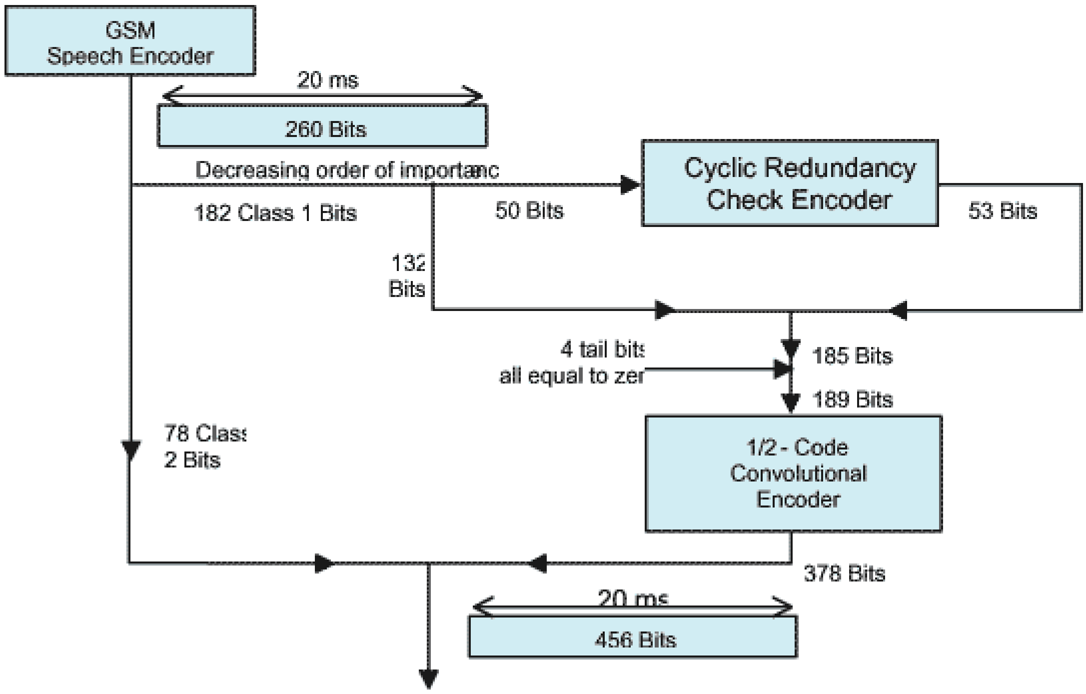

Figure 2 - Encoding and Decoding of a Speech Burst

A 20 millisecond speech sample is described (see last month's HOT TOPIC!) as a 260 bit word (which effectively contains the speech sample frequency coefficients). The 260 bits are split into Class 1 bits which have parity bits added and are then convolutionally encoded. Note the 1/2 encoder doubles the coder bit rate (2 bits out for every 1 bit in). Class 2 bits are uncoded. In the decoder, coded bits pass through the convolutional decoder. If the burst errors are longer than the interleaving depth (40 milliseconds in GSM), the block coded parity check detects a parity error, the speech sample is discarded and the prior sample re-used.

Increasing 'k' (the length of the convolutional encoder) increases resilience against burst errors and delivers additional coding gain ('k' = 7 typically delivers 5.2 dB gain, 'k' = 9 delivers 6 dB of gain) but requires an exponential increase in decoder complexity (trading MIPs against receive sensitivity).

The 3GPP1 AMR (**A**daptive **M**ulti **R**ate) vocoder moves the technique along a little bit further. There are 8 switchable source rates (4.75, 5.15, 5.9, 6.7 (PDC compatible), 7.4 (US TDMA compatible), 7.95, 10.2 and 12.2 kbps (GSM EFR compatible). Source rates can be chosen to give better quality (higher source rates) or more capacity (lower source rates). Note that lower source rates may be less error resilient (and may perform less well in fringe coverage areas).

The AMR codec is an adaptive codebook excitation linear prediction codec. Effectively a match is done between the frequency content of a speech sample and speech waveforms stored in a codebook. This increases memory bandwidth but decreases processor bandwidth, ie the power budget of the device is decreased. The speech sample bits are block and convolutionally decoded as Class A, Class B and

Class C bits. The additional coding overhead is justified on the basis that the vocoder delivers about a 1 dB gain in Eb/No over an existing enhanced full rate GSM (EFR) vocoder.

This coding gain depends on having sufficient interleaving depth available on the air interface. Interleaving depth in 3GPP1 (IMT2000DS/W-CDMA) is a variable, a minimum 10 milliseconds, a maximum of 80 milliseconds.

Increasing the interleaving depth (from 10 to 80 milliseconds) increases coding gain, by just under 1 dB for slow mobility users (3 km/h), by just over 1 dB for medium mobility users (20 km/h).

However, increasing interleaving depth increases delay.

An alternative is that careful implementation of fast power control (using the 1500 Hz power control loop specified in 3GPP1), makes it possible to 'follow' the fast fading envelope (already partly 'tamed' by the coherence bandwidth of the 5 MHz channel).

If the 'fast fading' can be counteracted by the power control loop, the Rayleigh channel becomes a Gaussian channel in which burst errors no longer occur.

Fast power control in a 5 MHz channel can therefore (theoretically) deliver additional coding gain at least for 'medium mobility' users (up to 20 km/h) **without** the need for deep interleaving.

This shows the intricate (and rather complex) relationship between source rate, convolutional encoding (the choice of 1/2 or 2/3 encoders for example), interleaving depth and coding gain (which in turn determines uplink and downlink sensitivity). Note also all the above parameters can be dynamically 'tuned' to optimise network performance.

The trade-off between voice quality, vocoder complexity and network density is shown in the diagram below.

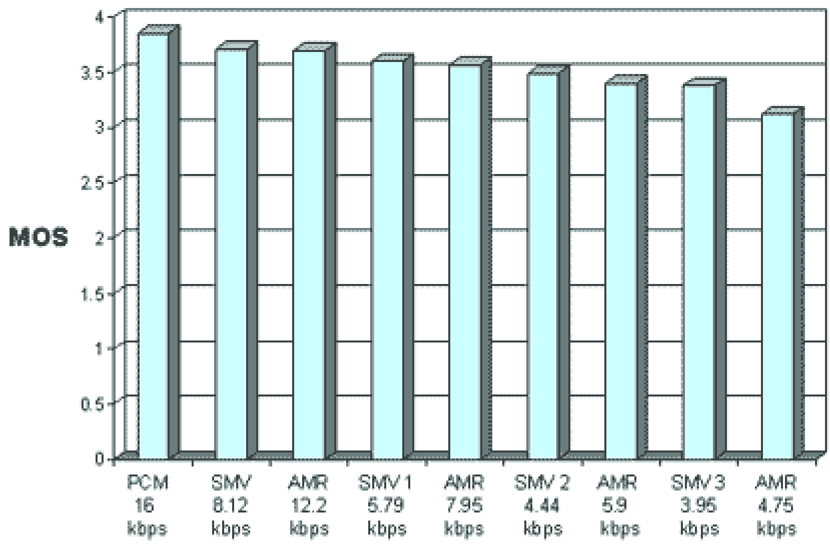

Figure 3 - SMV Modes vs AMR Rates
(Listening Tests at Dynastat Labs 10/00)

This shows a comparison between the vocoder adopted by 3GPP2 (the SMV selective multi-rate vocoder) and the AMR codec adopted by 3GPP1. The SMV vocoder, confusingly, given its description, adapts dynamically to the audio waveform presented to the decoder and provides a better voice quality/bandwidth quality trade off.

The 'cost' is additional processor complexity (and codec memory footprint). The benefit is a better perceived voice quality for a given network density. Quality is measured in terms of a mean opinion score (an objective way of collating subjective measurements).

CONVOLUTION AND CORRELATION

We have described convolutional encoding as a key mechanism for delivering coding gain (sensitivity) in 2G and 3G cellular networks. A further development of convolutional encoding is proposed for 3G handsets called turbo coding in which 2 (or more) convolutional encoders are used in parallel to increase 'coding distance' - is it a '0', is it a '1'.

Convolutional encoders are effectively implemented as shift registers. They are similar in terms of implementation to the PN code generators used to create long codes for IMT2000DS and IMT2000MC.

Long codes are used to provide (in IMT2000DS), base station to base station selectivity on the downlink and user to user selectively on the uplink. (See HOT TOPIC November 2001 for a more detailed explanation).

Both techniques exploit digital domain processes to deliver distance. Convolutional encoders deliver distance between '0' and '1's' (sensitivity). PN code generation delivers distance between parallel code streams (selectivity).

Baseband processing gain is a summation of source coding gain (last month's HOT TOPIC), channel coding gain (block codes, convolutional coding and long code correlation) and modulation and multiplexing gain (which will be covered next month).

Handset resident processor and memory budgets become directly tradeable against radio bandwidth efficiency and are a key ingredient for 3G bandwidth quality.

About RTT Technology Topics

RTT Technology Topics reflect areas of research that we are presently working on.

We aim to introduce new terminology and new ideas to clarify present and future technology and business issues.

Do pass these Technology Topics on to your colleagues, encourage them to join our Push List and respond with comments.

Contact RTT

RTT, the Shosteck Group and The Mobile World are presently working on a number of research and forecasting projects in the cellular, two way radio, satellite and broadcasting industry.

If you would like more information on this work then please contact

geoff@rttonline.com

00 44 208 744 3163