RTT TECHNOLOGY TOPIC
October 2005

Matrix Maths in Mobiles
Maths in Mobile Phones Part 2

The role of Hadamard-Rademacher-Walsh transforms in present and future radio systems.

In last month's Hot Topic we studied the **Fourier transform**, it's role in waveform geometry and **the role of waveform geometry in OFDM** (orthogonal frequency division multiplexing).

In this month's Hot Topic, we look at the role of **matrix maths** in mobiles and specifically **Hadamard-Rademacher-Walsh Transforms**, the contribution these transforms make to reducing costs and improving performance in present radio systems and their role in future radio systems.

The useful property of the **Fourier Transform** was defined as it's ability to represent any complex signal viewed in the frequency domain as a composite of simple sinusoidal signals.

In source coding, we use Fourier Transforms, or rather **Fast Fourier Transforms**, to analyse and synthesise the composite analogue waveforms encountered in real life. We do this in order to make the content (voice, audio, image, video) easier to compress.

In OFDM we user **Fourier transforms** to translate a symbol stream on to discrete frequency sub carriers. The effect is to provide **additional frequency diversity and time diversity** in that the symbols are further apart in time than the original bit stream. We do this so that we can support higher data rates in OFDM radio systems.

However present cellular systems make wide use of the **Hadamard Transform** both in the channel encoding/decoding process(convolutional and block coding)and code division multiple access (CDMA). **Channel coding** provides **time diversity**. **CDMA** provides **frequency diversity** (by spreading the original data signal across a wider bandwidth).

Future radio systems combine the properties of channel coding, CDMA **and** OFDM to deliver performance advantage.

Channel coding and CDMA together exploit the properties of matrix maths.

We exploit the properties of the Fourier Transform to provide performance gain in source coding and to implement OFDM.

We exploit the properties of matrix maths in channel coding and CDMA to achieve a system performance gain.

The origins of matrix maths

Matrix maths is the science of putting numbers in boxes or rather, arranging numbers in rows and columns (a horizontal and vertical matrix).The numbers can be **binary or non binary**.

The earliest documentary evidence of applied maths using matrices are clay tablets used by the Babylonians in the third or fourth century BC, the forerunner of the counting tables used by the Romans. Counting tables were simply tables with a ridged edge. The tables contained sand and the sand would be divided up into squares to help in counting and calculation.

Early matrix concepts were developed by the Chinese in the **Han Dynasty between 200 and 100 BC** to solve linear equations. This implied an understanding that **matrices exhibit certain properties**, later described as '**determinants**' which are revealed when **the numbers in the matrix are added or multiplied either vertically by row, horizontally or diagonally.**

This understanding was documented in **'Nine Chapters on the Mathematic Art'** put together around **100 AD** and was the product of a period now often described as '**The First Golden Age of Chinese Mathematics**' (the second golden age was in the 13th and 14th century).

Historically it is important to realise that the development of **matrix theory in China** was contemporaneous with the work of **Archimedes** on **curves and circles and Pi** between 287 and 212 BC.

Thus **the origins of the Fourier Transform and the Hadamard Transform can both be traced back to the pre Christian era.**

As with the science of geometry, it took over 1000 years before anything else meaningful happened in matrix theory.

In **1683**, the Japanese mathemetician **Takakazu Seki** wrote the 'Method of solving dissimulated problems' which precisely described how the earlier Chinese matrix methods had been constructed.

At the same time, in Europe, **Gottfried Leibniz** was producing work on determinants (which he called 'resultants'). This became the basis for a body of work to which mathematicians like **Cramer, Maclaurin, Bezout and Laplace** all contributed throughout the **18th century.**

The term 'determinant' was used by **Gauss** in **1801** in his 'Disquisitiones arithmeticae' side by side with his work on the coefficients of quadratic forms in **rectangular arrays** and **matrix multiplication.** This became the basis for a body of work to which other mathematicians such as **Cauchy** (who proved that **every real symmetric matrix is diagonisable**), **Sturm, Cayley and Eisenstein** contributed in the **first half of the 19th century.**

However it was **JJ Sylvester** who is first credited with using the term 'matrix' in **1850.** Sylvester defined a matrix to be 'an oblong arrangement of terms which could be used to discover various 'determinants' from the square arrays contained within it'

This work was developed in his grandly titled **1867** paper 'Thoughts on **Inverse Orthogonal Matrices**,**Simultaneous Sign-successions** and **Tesselated Pavements** in two or more colours, with applications to **Newton's Rule**, **Ornamental Tile Work**, and the **Theory of Numbers**'.

This paper established a new level of understanding about pattern behaviour in matrices (using ornamental tiles as an example). In between times, Sylvester gave maths tuition to **Florence Nightingale.**

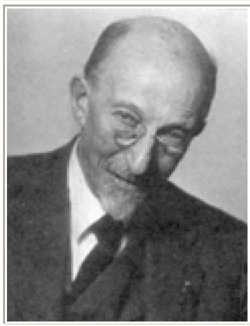
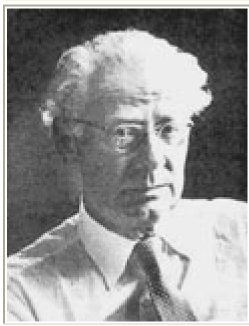
At which point we can introduce **Jacques Hadamard**.

Jacques Hadamard and his family survived the Prussian siege of Paris in 1870 by eating elephant meat.

Hadamard obtained his doctorate in 1892 with a thesis on analytic theory and related work on **determinant equality**, the property that **allows matrices to be used as a reversible transform.** Hadamard also produced pioneering work on **boundary value problems** and **functional analysis** and is generally seen as 'the founding father' of **modern coding theory**.

Hadamard's work was developed by **Hans Rademacher**, particularly in the area of **orthogonal functions** now known as **Rademacher functions** that appeared in a paper published in **1922** and was the forerunner of pioneering work in **analytic number theory.**

Hans Rademacher's work was contemporaneous with the work of **Joseph Leonard Walsh** (known as 'Joe' to his friends). This included a publication in 1923 on orthogonal expansions, later called **'Walsh Functions'.** Joe Walsh became a full professor at Harvard in 1935 and produced pioneering work on the relationship of **maths and discrete harmonic analyis**.

(Matrix) Mathematicians of the Month		
Jacques Hadamard 1865-1963	**Hans Rademacher 1892-1969**	**Joseph Leonard Walsh 1895-1972**
		

The Hadamard Matrix and the Hadamard Transform

In last month's Hot Topic we said that a **transform changes something into something else**. The process is **most useful when** it is **reversible/bi-directional** and the **purpose** is generally to **make a particular process easier to achieve**.

In our context of interest, we want to take a string of numbers and rearrange or redistribute the number string in rows and columns so that they are more easily processed. In other words, **a Hadamard Transform is a transform that exploits the properties of a Hadamard matrix** in the same way that a Fourier transform exploits the properties of the Fourier number series (the ability to describe waveforms as summations of sines and cosines).

The properties of a Hadamard Matrix

Hadamard matrices possess a number of useful properties.

Hadamard matrices are **symmetric** which means that **specific rows can be matched to specific columns.**

Hadamard matrices are **orthogonal** which means that **the binary product between any two rows equals zero**.(the binary product is simply the result of multiplying all the components of two vectors, in this case rows, together and adding the results). We will see why this is useful later.

This means that if you compare **any two rows**, they **match in exactly N/2 places and differ in exactly N/2 places** so **the 'distance' between them is exactly N/2**. We explain why 'distance' is useful later.

Exactly half of the places that match are +1's and the other half are -1's. Exactly half of the places that differ are (-1+1) pairs and exactly half are (+1-1) pairs (the symmetric properties of the matrix).

You can turn a Hadamard matrix upside down (reverse the +1's and -1's) and it will still work.

The matrix has the property of **sequency**. The sequence number of each row is the product of the number of transitions from +1 to -1 in that row. A row's sequence number is called it's sequency because it measures the number of zero crossings in a given interval. Each row has it's own unique **sequency value** which is **separate from its natural order (the row number)**.

The **Hadamard Transform** can therefore be correctly described as a **Sequency Transform** which is directly analogous to describing the **Fourier Transform** as a **Frequency Transform**.

In other words, given that the rows in a Hadamard matrix are orthogonal, the Hadamard Transform can be used to decompose any signal into it's constituent Hadamard components. In this sense it works just like a Fourier Transform but with the components based on **sequency rather than frequency.**

As with the Fourier Transform, a number of these components can be discarded without destroying the original data so the **Hadamard transform** can be used for **compression** . We address this in detail in next month's Hot Topic

The **Hadamard transform** can also be used for **error correction.** A much quoted example is the use of Hadamard Transforms to code the pictures coming back from the visits to the moon in the 1960's and the Mariner and Voyager missions to Mars. The pictures were produced by taking three black and white pictures in turn through red, green and blue filters. Each picture was considered as a thousand by thousand matrix of black and white pixels and graded on a scale of 1-16 according to it's greyness (white is 1, black is 16). These grades were then used to choose a codeword in a eight error correction code based on a Hadamard matrix of order 32. The codeword was transmitted to earth and then error corrected.

It was this practical experience with applied Hadamard Transforms that led on to the use of Hadamard Transforms in all present generation cellular systems (including GSM and CDMA).

These use convolutional and block coding to increase the distance between a 0 and a 1 or rather a -1 and a +1. The process is described in more detail below. Note that **channel coding** is a distinct and separate though related process to **code division multiple access.** Both processes exploit the properties of the Hadamard matrix.

Channel coding produces 'coding gain' and code division multiple access produces 'spreading gain'. Coding gain can be in the order of 10 dB or so and spreading gain in the order of 20 dB for lower user data rates. Together they show the extent to which the Hadamard transform contributes to the link budget of present cellular radio systems.

Differences between the FFT and the FHT

The FFT is best at modelling curves and sinusoidal waveforms. The hardest curve to model with a Fourier Transform is a step function, also known as a square wave, where the edges of the waveform exhibit a theoretically infinite number of sinusoids. In practice these can be approximated but it is **the 'Achilles heel' of the Fourier transform** (often described as **'the Gibbs Effect'**).

The **FHT** is **best at capturing square waves**. The hardest curve to model with a Hadamard Transform is a basic Sine/Cosine curve. This is intuitively consistent with matrix theory - describing square waveforms by putting numbers into squares.

Hadamard Transforms when implemented as Fast Walsh Hadamard Transforms use only **additions and subtractions** and are therefore **computationally efficient.**

Fourier Transforms require **many multiplications** and are slow and expensive to execute. Fast Fourier Transforms employ imperfect 'twiddle factors' so trade accuracy against complexity and 'convergence delay'.

The magnitude of an FFT is invariant to phase shifts in the signal. This is not true in the case of the FHT because a circular shift in one row of the Hadamard matrix does not leave it orthogonal in other rows. This is the Achilles heal of the FHT and is a weakness that underpins the ultimate limitations of CDMA in terms of error performance and susceptibility to AM/PM distortion.

However with this proviso, the Hadamard Transform has been and remains a fundamental part of the signal processing chain in a mobile phone both in terms of it's application in discrete processes such as channel coding and code division multiplexing but also in a support role to other processes.

The fact that it is simpler to execute and has different but complementary properties makes it a useful companion to the FFT and the two processes **working together** provide the basis for future performance gain.

Next month's Hot Topic looks specifically at the role of the FHT, the FFT, wavelet transforms and AM/FM transforms (a sense of the wheel going full circle) in source coding applications and the integration of these techniques with other parts of the signal processing chain.

But for the moment let's concentrate specifically on the Hadamard Transform.

Having mastered the theory, let's examine how the Hadamard Transform is applied in present radio systems.

Some naming issues

For brevity and in due deference to Hans Rademacher and Joe Walsh, we shall describe these codes as **Hadamard codes** used as a **Hadamard Transform**.

As with the Fourier Transform, the Hadamard Transform computation can be decimated to speed up the computation process in which case it is known as a **Fast Hadamard Transform (FHT)**.

In some ways, the **FHT** is **easier to implement computationally** as it does not require the 'twiddle factors' implicit in the FFT. This is because the discrete F**F**T is approximating and then describing a composite sinusoidal waveform (hence the twiddle factors). THe F**H**T is describing square waves and therefore does not need additional approximation correction factors The **Fast Hadamard Transform (FHT)** is used on the **receive path.** The **Inverse Fast Hadamard Transform (IFHT)** is used on the **transmit path.**

As we have said, the FHT has properties that are distinct and different from the FFT (the Fast Fourier Transform) but are also complementary.

The combination of the two techniques (the FHT and FFT together) deliver a number of specific performance advantages which should be realisable in next generation radio systems including wide area (**HSOPA and 1XEVDO/FLO**)and local and personal area networks.

These benefits should include cost reduction, improved coverage, improved capacity, and more consistent and flexible radio access connectivity.

Cost reduction

In the 1970's there was a consensus that it was going to be easier(cheaper) to filter in the time domain rather than the frequency domain and by implication, to process and filter in the digital domain rather than the analogue domain

This started the process whereby channel spacing in cellular systems has relaxed from the 25 or 30 KHz used in first generation systems to the 200 KHz used in GSM to the 1.25 MHz or 5 MHz systems used in present 3G networks. The process is taken further, for example in WiFi systems (20 MHz) and ultra wide band radio systems (>500 MHz).

The objective is to reduce the cost of RF(radio frequency) filtering both in the handset and the base station

The need to deliver cost reduction AND better performance

However user expectations of performance increase over time. User data rates in first generation analogue systems were typically 1200 or 2400 bits per second, GSM data rates are tens of kilobits, 3G data rates are (supposed to be) hundreds of kilobits and have to compete in the longer term with local area WiFi systems delivering tens of megabits and personal area systems delivering hundreds of megabits (wireless UWB USB at 460 mbits/s as an example).

The performance of a radio system can be measured in terms of the radio system's sensitivity, selectivity and stability.

Sensitivity is the ability of the radio system to extract a wanted signal from the noise floor. Improved sensitivity translates into improved range and/or an ability to support higher data rates. In terms of the user experience, **sensitivity equals coverage and capacity.**

Selectivity is the ability of the radio system to extract a wanted signal in the presence of unwanted signals from other users. As with sensitivity, improved selectivity translates into improved **range and capacity.** However, by relaxing the RF channel spacing over time, we have thrown away some of the selectivity inherent in narrow band radio systems so have the need to replicate this in some other way. In parallel, users expect to receive voice and non voice services in parallel so we have **the need to support multiple data streams per user** which implies a need to provide additional per user channel to channel selectivity.

Stability is the ability of the radio system to perform consistently over temperature over time which in turn is dependent on the short and long term accuracy of the frequency and time reference used in the transceiver. The move to higher frequencies in the microwave band has increased the need for a more accurate **frequency reference** but this has been off set by the relaxation in RF channel spacing. The combination of higher data rates and the need to deliver improved sensitivity and selectivity in the baseband processing sections of the transceiver has increased the need for a more accurate (and potentially expensive) **time reference**. As we shall see later, this in a sense is the Achilles heal of present CDMA systems (their inability to scale to much higher data rates without an inconveniently accurate time reference). In last month's Hot Topic we showed how OFDM shifts some of the hard work involved here back into the frequency domain. In terms of the user experience, stability therefore translates directly into **user data rates** AND the **consistency of the user experience**.

The role of binary arithmetic in achieving sensitivity, selectivity and stability

In 1937, **Claude Shannon**'s MIT thesis 'A symbolic analysis of relay and switching circuits' helped to establish the modern science of using binary arithmetic in wireless(and wireline) communications. The science was consolidated by **Richard Hamming** in his work on error detection and correction codes (1950), digital filtering (1977), coding and information theory (1980), numerical analysis (1989) and probability (1991). Hamming formalised the concept of **distance** between binary numbers and binary number strings which is in practice the foundation of modern radio system design.

In digital radio systems, we take real world analogue signals (voice, audio, video, image) and turn the analogue signals into a digital bit stream which is then mathematically manipulated to achieve the three 'wanted properties' - sensitivity, selectivity, stability.

Note that analogue comes from the Greek word meaning proportionate. Analogue implies that the output of a system should be directly proportionate (i.e. linear) to the input of the system and is continuously varying. To represent these waveforms digitally requires a sampling process which has to be sufficiently robust to ensure that analogue waveforms can be reconstructed accurately in the receiver. (**Harry Nyquist** 'Certain Factors affecting telegraph speed 1924).

Anyway, taking this small but significant proviso into account binary numbers can be used to deliver sensitivity.

Coding distance - sensitivity
0 - 1

For example, moving a 1 further away from a 0 implies an increase in **distance** which implies an increase in **sensitivity**.

Coding distance - selectivity
0110101101001010 0 10011011101100010

The greater the **distance** between two strings of numbers (code streams in CDMA), the better the **selectivity** between users. The above two codes differ in 10 places which describes their 'hamming distance' from each other.

Coding distance - stability (code correlation)
01101011010010100 01101011010010100

If two code streams are identical (no distance between them) they can be used to lock on to each other, for example to provide a **time reference** from a base station to a handset or a handset to a base station. Longer strings of 0s and 1s will produce distinct spectral components in the frequency domain which can be used to provide a **frequency** reference.

Counting in binary						
1	1	0	1	0	0	1
64	32	16	8	4	2	1

We can also use binary numbers as a counting system. Interestingly, as we shall see later, if we start arranging 0's and 1's in a **symmetric matrix** of rows and columns, the binary product (sometimes known as the dot product) of the numbers in a column or row can be used to uniquely identify the position of that column or row. This is a property(described earlier) known as **sequency** and is the basis of many of the error **correction** schemes used in present radio systems.

Coding distance and bandwidth gain

A first step to increasing the distance between a 0 and a 1 is to change a 0 into a -1 and a 1 into a +1. If we take either a -1 or a +1 and multiply by a series of -1's and +1s running at a faster rate then the bandwidth of the composite signal is expanded. The converse process applied in the receiver will take the composite 'wide band' signal and collapse the signal back to its original (data) bandwidth. This is the principle of spreading gain used in CDMA systems and is in many ways analogous to the bandwidth gain achieved in a wideband FM radio system.

An example - The Barker code used in 802.11 b WiFi systems

Basic 802.11 b WiFi systems provide an example. The original 802.11 b standard supports data rates of 1 M/bit/s and 2 M/bits/s using either BPSK modulation (1 M/bit/s) or QPSK (2 M/bit/s). The data bits are multiplied with an 11 bit Barker sequence at a 1 MHz data rate which expands the data bandwidth of 2 MHz to an occupied channel bandwidth of 22 MHz giving just over 10 dB of coding gain.

Barker sequences are named after **RH Barker** from his 1953 paper on ' Group Synchronisation of Binary Digital Systems read at the **IEE in London**. They were/ are widely used in radar systems to help in distance estimation and were first used in low cost commercial two way radio systems in the first generation of digital cordless phones developed for the 902-908 MHz US ISM band.

The 11 bit Barker code used in 802.11 b is as follows

11 bit Barker sequence
+1 -1 +1 +1 -1 +1 +1 +1-1-1-1-1

If we take this spreading sequence and multiply it with an input data bit -1 and apply the rule that **if the signs are different, the result is a -1**, **if the signs are the same the result is a + 1** then we get the following

Input data bit	-1										
Spreading code	+1	-1	+1	+1	-1	+1	+1	+1	-1	-1	-1
Composite code	-1	+1	-1	-1	+1	-1	-1	-1	+1	+1	+1
Despreading code	+1	-1	+1	+1	-1	+1	+1	+1	-1	-1	-1
Output data bit	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1

As you can see, the despreading code is the same as the spreading code.

Effectively we have answered the question 'is at a -1 or + 1?' eleven times over and it is this which gives us the **spreading gain**.

Complementary Code Keying used in 802.11 b

However, if the WiFi data rate is increased to 11 M/bits per second, the spreading gain disappears.

In this case, the Barker code is replaced with 64 different 8 bit codes. The data bits are grouped into 6 bit symbols and each 6 bit symbol is mapped to one of the 64 codes. When the receiver demodulates the symbol stream/bit stream, the 8 bits received should be one of the 64 eight 8 bit code sequences which correspond to one of the 6 bit input data symbols. This is described as **complementary code keying and is a good example of the use of sequency in the encode decode process.**

There is technically no spreading gain with this arrangement though there is some (modest) coding gain due to the equality of distance between each of the 64 codes. The occupied bandwidth remains at 22 MHz.

The difficulty then arises as to how to manage higher user data rates. The answer with 802.11 b is to use an OFDM multiplex as described in previous Hot Topics.

Walsh Codes used in IS95 CDMA/1EX EV

Present generation **wide area cellular** CDMA systems have to date not needed to support the higher data rates expected in local area systems and for that reason have not to date needed to use an OFDM multiplex.

The code multiplexing and channel coding were chosen to provide a good compromise between implementation complexity and performance.

IS95 CDMA, the pre cursor of the CDMA2000 and 1XEV/DO system in use today, uses a **64 by 64 Hadamard matrix.** This consists of 64 codes of length 64 of which code 0 is made up of all 1's and is used as a pilot and code 32 is made up of alternating 1's and 0's and is used for synchronisation. The other codes have their 0's and 1's, or rather -1's and +1's, arranged so that each of the codes is orthogonal to each other. Orthogonal in this context means that the codes are equally distinct from one another or in other words do not interfere with each other as a product of the (FHT) transformation process. These codes are often referred to as **Walsh codes** (named after Joseph Walsh) but are in practice based on the Hadamard matrix. Each code has 32 places where it is different from other codes. In other words each code has a **Hamming distance** of 32 from other codes in the matrix.

In the **uplink,** every six information bits are mapped to one of the 64 bit rows of the Hadamard matrix. The 64 bits in the row are substituted for the original 6 bits and the 64 bits are modulated on to the radio carrier using QPSK modulation. This is an **Inverse Fast Hadamard transform.**

The base station applies a **Fast Hadamard Transform** on every 64 received bits. Ideally only one of the resultant FHT coefficients will be non zero. The non zero value determines the row number which in turn determines the 6 bits originally sent. In other words, the process exploits the property of **'sequency'** implicit in the Hadamard Matrix.

Elegantly, the IFHT/FHT delivers some useful spreading gain (64/6=10.75 dB). It is also error tolerant. Given that the Hamming distance between each Hadamard code is 32, up to 15 bits can be errored per block of 64 without corrupting the 64 bits actually sent.

The only slight snag is that all users are co sharing the 64 codes and have to be separated from each other by unique scrambling codes (-1's and + 1's running at the same rate as the data) . The **spreading codes deliver sensitivity**, the **scrambling codes deliver selectivity**, **the pilot and synchronisation codes deliver stability.**

In the downlink, each row in the Hadamard matrix can be used to carry a unique channel to a unique user. Theoretically this means 62 channels per 1.25 MHz of channel bandwidth (taking out the pilot and synchronisation channel). Every single information bit is replaced with the entire 64 bits of the users code (a 64/1 expansion). A data rate of 19.2 kbps therefore is spread to an outbound data rate of 1.2888Mbps occupying a 1.25 MHz channel. As with the uplink, a scrambling code that is unique to the base station is also applied to provide base station to base station selectivity (actually a single code 'pseudo noise' sequence off set in time for each base station).

Later evolutions of IS95 have increased the matrix to 128 rather than 64 but the same principles apply. Either way, the CDMA multiplexing and channel coding have proved to be an effective format for exploiting the properties of the Hadamard matrix to deliver beneficial performance gains over simpler radio systems.

OVSF Codes in W-CDMA

The orthogonal variable spreading factor codes used in W CDMA were originally conceived as a re-ordering of the Walsh codes used in IS95 CDMA with the added twist that user data rates could be changed every 10 milliseconds with users being moved between different lengths of spreading code (hence the 'variable'description used).

SF4	SF8	SF16
		+1+1+1+1+1+1+1+1+1+1+1+1+1+1+1+1
	+1+1+1+1+1+1+1+1	
		+1+1+1+1+1+1+1+1-1-1-1-1-1-1-1-1
+1+1+1+1		
		+1+1+1+1-1-1-1-1+1+1+1+1-1-1-1-1
	+1+1+1+1-1-1-1-1	
		+1+1+1+1-1-1-1-1-1-1-1-1+1+1+1+1
		+1+1-1-1+1+1-1-1+1+1-1-1+1+1-1-1
	+1+1-1-1+1+1-1-1	
		+1+1-1-1+1+1-1-1-1-1+1+1-1-1+1+1
+1+1-1-1		
		+1+1-1-1-1-1+1+1+1+1-1-1-1-1+1+1
	+1+1-1-1-1-1+1+1	
		+1+1-1-1-1-1+1+1-1-1+1+1+1+1-1-1
		+1-1+1-1+1-1+1-1+1-1+1-1+1-1+1-1.
	+1-1+1-1+1-1+1-1	+1-1+1-1+1-1+1-1-1+1-1+1-1+1-1+1
+1-1+1-1		+1-1+1-1-1+1-1+1+1-1+1-1-1+1-1+1
	+1-1+1-1-1+1-1+1	+1-1+1-1-1+1-1+1-1+1-1+1+1-1+1-1
		+1-1-1+1+1-1-1+1+1-1-1+1+1-1-1+1
+1-1-1+1	+1-1-1+1+1-1-1+1	+1-1-1+1+1-1-1+1-1+1+1-1-1+1+1-1
		+1-1-1+1-1+1+1-1+1-1-1+1-1+1+1-1
	+1-1-1+1-1+1+1-1	+1-1-1+1-1+1+1-1-1+1+1-1+1-1-1+1

The section in bold indicates that this is a tree structured code. The codes to the right are longer copies of the codes to the left. The bold segment denotes a branch of the tree stretching from left to right. At SF4 (which means spreading factor 4) four users can be supported on each of 4 codes at, say a theoretical data rate of 960 Kbits. The code tree then extends rightwards to SF256 (not shown for reasons of space and k/bits/s.

As the data rate changes, potentially every frame (every 10 milliseconds), users can be moved to the left or right of the code tree. However if a user is at SF4, no users can be on codes to the right on the same branch. Similarly if you have two users at SF8 or 4 users at SF16 on the same branch no users to the right on the same branch can be supported and so on rightwards across the branch.

A user at SF4 will have minimal spreading gain. A user at SF256 will have maximum spreading gain with a difference of just over 20 dB between the two extremes. As you would expect this means that as a user's data rate increases, the spreading gain decreases. The occupied bandwidth (5 MHz in this case) remains the same.

The spreading codes are used with **scrambling codes** (long codes) with the scambling codes providing user to user/ channel to channel **selectivity** on the uplink and base station to base station selectivity on the downlink. Additional **short codes** are used for uplink and downlink **synchronisation.**

This is the **Release 99** WCDMA radio layer code scheme. It provides a significant amount of flexibility both in terms of being able to support a wide range of variable (and potentially fast changing) data rates per user and a significant amount of flexibility in being able to support multiple data streams per user.

It does however require careful implementation both in terms of code planning and power planning. Although the variable spreading factor codes are orthogonal (hence their name orthogonal spreading factor codes), this orthogonality can be lost if code power is not carefully balanced or the non linearities inherent both in the radio system and the channel are not managed aggressively.

If orthogonality is compromised, unwanted error energy is projected across code channels which will then suffer from an unacceptably high error vector magnitude which in turn compromises sensitivity and selectivity which in turn compromises coverage (range) and capacity.

The mechanism for power management is an outer and inner control loop. The inner control loop also known as fast power control runs at 1500 Hz and if correctly implemented can be effective but in its own right absorbs a significant percentage of the available signal energy (about 20%).

HSDPA aims to simplify this process and reduce some of the power control signalling and energy overhead by only using the SF16 part of the code tree and dispensing with fast power control. However, as you might have noticed, this takes away one of the desired properties of the OVSF code tree which is the ability to support lots of users each with multiple simultaneous channels each at a variable data rate. In other words, much of the multiplexing capability of the OVSF code tree disappears if only SF16 is used.

The answer used in HSDPA is to have a **high speed data shared channel (HS-DSCH)** which can be shared by multiple users with a MAC driven access control based on 2 millisecond (and later .5 millisecond frames) that is not dissimilar to the contention based MAC used in present WiFi systems.

The challenge here is that the shared channel requires a new **high speed shared physical control channel (HSDPCCH).** This control channel has to carry the channel quality indication (**CQI)** messages and acknowledgement/negative acknowledgement (**ACK/NACK**) messages that the MAC needs to decide on admission control and other factors such as coding overhead, choice of modulation and transmission time interval.

This signalling is discontinuous but when present increases the peak to average ratio of the transmitted (composite) signal and can also encounter relative timing issues with the other (dedicated)control channels.

If the high speed control channel is not correctly detected, no communication takes place which is potentially a bit hazardous. The peak to average ratio can be accommodated by backing off the PA but this has an impact on coverage(range).

In a sense, **HSDPA has exchanged the code planning and power planning challenges inherent in Release 99 WCDMA with code sharing and power sharing issues.** This means that the RF performance of the handset and base station remains as a critical component of overall system performance.

Although some of the functional complexity at the PHY (physical) radio level has been moved to the MAC (medium access control) level, the effectiveness and efficiency of the MAC is dependent on the careful measurement and interpretation of the CQI and ACK/NACK responses.

The **7dB step change in power** that occurs when the **CQI and/or ACK/NACK signalling** is transmitted can trigger **AM/PM distortion.** This may cause phase errors which in turn will compromise CQI measurements or disrupt the ACK/NACK signalling.

This will probably be the determining factor limiting coverage and will probably require some conservative cell geometry factors (signal versus noise rise across the cell) in order to maintain the signalling path (without which nothing else happens).

The requirement for a more complex and flexible multiplex can be met either by having a rather over complex code structure and/or an over complex MAC. Either or both can be problematic both from a radio planning and/or a handset/base station design perspective.

CDMA/OFDM hybrids as a solution

This seems to imply that something else will need to be done to allow these wide area radio systems to deliver data rates that meet user's likely future data rate expectations.

The options are either to increase cell density and/or to increase the sensitivity, selectivity and stability of the handsets and base stations- preferably both.

In 1XEV, (including the most recent 1XEV-DO Revision A) handset enhancements are based on implementing receive diversity and advanced equalisation. Base station enhancements include implementing 4 branch receive diversity (two pairs of cross polarised spatially separated antennas) and pilot interference cancellation - a mechanism for getting unwanted signal energy out of the receive path.

Qualcomm's **MediaFlo** uses these techniques but adds an **OFDM multiplex.** This is specifically applied to optimise a broadcast downlink (hence the name MediaFlo). A time division multiplex slot is allocated to all cells in a region as a dedicated broadcast downlink. The slot carries one or more dedicated broadcast packets. Handsets receive the same broadcast packet from multiple cells and then soft combine to improve reception. The OFDM multiplex helps the soft combining process (by slowing the symbol rate).

MediaFlo is however an early example of a more general application of **combined CDMA/OFDM radio systems** sometimes described as **'scaleable bandwidth'** systems where the occupied bandwidth can be anything from a few tens of KHz to hundreds of MHz. Practically in the 1XEV and MediaFlo road map this seems to suggest scaleability from the existing 1.25 MHz channels to **20 MHz channels** (with compatibility to existing WiFi and proposed WiMax system options).

A similar evolution path exists for HSDPA with advanced receiver techniques being specified in Release 6 together with an enhanced uplink (**HSUPA**). **Release 7** is likely to include standardised multiple antenna/**MIMO** proposals and **HSOPA** - the addition of an OFDM multiplex in the downlink to provide data rates approaching **40 Mbps in a 20 MHz or 40 MHz channel.**There is no present proposal to implement OFDM in the uplink either in 1X EV-DO or HSOPA.

The downlink evolution road map for 1XEV and HSDPA does however mean that we will have a combination of CDMA and OFDM in at least two mainstream wide area cellular standards, sometimes generically described as **'Super 3G'.**

It is therefore useful to have an understanding of how a combination of **CDMA and OFDM** will work in terms of **signal processing task partitioning**.

FHT/ FFT task partitioning in future radio systems

Transmit path			
Source coding	Channel coding	Channel Multiplexing, Orthogonal spreading codes and scrambling codes	Frequency multiplexing orthogonal frequency division multiple access
Voice, audio, image,video	Convolutional and block coding	CDMA	OFDM
Inverse FFT	Inverse FHT	Inverse FHT	Inverse FFT

The above shows the transforms, or rather inverse transforms used in the transmit path of a hybrid CDMA/OFDM transceiver.

The job of **the inverse FFT in source coding** is to take the composite time domain waveform from the quantised voice, audio, image and video samples and to transform them to the frequency domain. The transform makes it easier to separate out the entropy (information) and redundancy in the source signal. **The bandwidth of the signal is compressed**. We cover this process in more detail in next month's Hot Topic.

The bit streams representing the voice, audio, image or video samples are then channel coded using **convolutional and block encoding.** This **expands the occupied bandwidth** but increases the distance between the information bits (-1's and +1's). This is an **Inverse Hadamard Transform**.

The bit stream is then 'covered' with a **spreading code and scrambling code**. This is another **Inverse Hadamard Transform**. This further **expands the occupied bandwidth.**

The bit stream is then transformed again using an **IFFT** to distribute the data stream across discrete frequency sub carriers. Note that the IFFT is imposing a set of time domain waveforms on a series of frequency sub carriers (lots of sine/cosine calculations).The number of points used in the I FFT/FFT and the characterisation of the IFFT/FFT determines the number of sub carriers and their spacing and is the basis of the **'scaleable bandwidth'** proposition implicit both in MediaFlo and present HSOPA proposals.

In the receiver, the **OFDM demultiplex** (an **FFT**), recovers the wanted symbol energy from the discrete frequency sub carriers. The benefit here, as explained last month, is that the symbol rate per sub carrier will be divided down by the number of sub carriers. The more sub carriers, the slower the symbol rate. This should make the next stage of the process easier which is;

If this is a combined OFDM/CDMA system, the **combining** of the symbol stream (assuming this is a diversity or MIMO system) and the despreading and descrambling of the signal using an **FHT.** The result should be some useful **diversity gain** (courtesy of the multiple receive paths), some **spreading gain** (courtesy of the **spreading/despreading** codes) and **additional selectivity** (courtesy of the **scrambling codes)**. This should make the next stage of the process easier which is:

Channel decoding, usually or often implemented as a turbo coder (two convolutional encode/decode paths running in parallel). The **FHT** produces some additional 'distance' which translates into **coding gain**.

And finally, the bit stream now finally recovered from the symbol stream is delivered to the source decoder where an FFT is applied (frequency domain to time domain transform) to recover or rather reconstruct or synthesise the original analogue waveform.

Summary

In this month's Hot Topic, we have studied the properties of the Hadamard Transform and it's practical application in present CDMA cellular systems and early generation WiFi systems.

We have showed how the IFHT/ FHT is used in the code division multiplex/demultiplex and in channel encoding/decoding to deliver 'distance' which can be translated into coverage and capacity gain.

We have reviewed how the IFFT/FFT is used to add in an OFDM multiplex to slow down the channel symbol rate as a (potentially) power efficient mechanism for accommodating higher user data rates.

Note that all these processes are absorbing processor clock cycles with the express purpose of achieving a nett gain in system performance which can be translated into supporting higher data rates at lower power (the end user benefit).

The challenge is to deliver these benefits consistently given that user expectations increase over time and that physics tends to dictate the limits of any process that we are using to realise performance gain.

In the context of HSDPA and HSOPA, it will for example be a real challenge to implement the PHY and MAC sufficiently robustly to provide a realistic chance of achieving the theoretical gains. The same implementation issues will apply to future iterations of 1XEV/DO.

In next month's Hot Topic we study the role of the FFT in voice, audio, image and video source coding and the integration of the FHT, FHT and other transforms (including AM/FM and wavelet transforms) to deliver step function improvements in image and voice processing efficiency.

In practice, it always tends to be a combination of techniques that together deliver performance improvements which are translatable into a better more consistent user experience.

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