TERMINAL EQUIPMENTS ANALYSIS FOR IMT 2000 NETWORK PL

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I would like to dedicate this book to my lovely family for the support and encouragement that always give me, and also to the Universidad Tecnológica Empresarial de Guayaquil – UTEG for the support on this work.

Mariajose Vaca R.

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Prologue

This book presents a compilation of important information in the technological and communication areas, its publishment holds a significative input with celular phones. The author shows experience in the drafting and presentation of tables and graphics validated in her experience on different areas.

The book presents a good structure in which the steps done by the study are detailed to compile the data shown on this paper, it describes the actual situation of communication from an infrastructure point of view.

The process of measuring performance of networks in production and compare them against the focus infraestructure is known as “network validation”. Because efficient use of resources is a key design problem for any cellular network, in general, and code division multiple access (CDMA) networks, in particular, where the number of
simultaneous calls that can be admitted in one cell depends on the number of simultaneous calls in many cells in the network, network validation becomes a must perform process.

The focus infraestructure is usually a new or recently implemented network. Network validation is necessary to match network planning and configuration to “sturdy” (or immune) networks. These networks are grade of service proven and in production. The know-how transferred in this process is one of the singlemost important factors for network fine tune-up and used as a parameter to validate the network design and implementation based on the comparison of performance against other networks.

Chapter I

Project Overview

The IMT-2000 is the term used by the International Telecommunications Union (ITU) for a set of globally harmonized standards for third generation (3G) mobile telecoms services and equipment. 3G services are designed to offer broadband cellular access at speeds of 2Mbps, which will allow mobile multimedia services to become possible. In 1998, the ITU called for proposals for IMT-2000 from different interested parties and it received many different ideas based on time division multiple access (TDMA) and code division multiple access (CDMA) technology. The European Telecommunications Standards Institute (ETSI) and Global System for Mobile Communications (GSMC) companies, such as the infrastructure vendors Nokia and Ericsson, are backing wideband code division multiple access (W-CDMA), whilst the US vendors, including Qualcomm and Lucent Technologies, are backing CDMA2000.

The acceptance of CDMA2000 solutions in Asia, North America, and Latin America, which accounted for over 19 million 3G CDMA2000 users as of September 2002 and adding more than 2 million new subscribers per month, provides vivid proof of the demand for 3G services and applications. A broad range of color LCD phones and wireless devices are entering

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the market, including: camera phones, Video on Demand phones, Global Positioning System (GPS), and multimedia-enabled devices. These new devices, combined with high-speed packet data, are opening the door for the creation of a broad array of services that deliver greater choice for consumers and enhance operators' abilities to differentiate and drive new sources of revenue. The Japanese and Korean experiences demonstrate that the increased capabilities of CDMA2000 networks can drive a fundamentally different consumer experience and operator business model.

High-speed wireless data, phones with color displays and camera attachments, built-in
MP3 and video players, downloadable games and graphics, new location-based safety
and security services—3G delivers all of these, and more.

CDMA2000 is a cost-effective evolutionary technology for CDMA and TDMA operators. CDMA2000 evolved from cdmaOne (IS-95) technology. CDMA2000's development benefited from extensive experience acquired through the operation of cdmaOne systems, as well as the requirements of third generation wireless systems.

The result is an efficient and versatile technology. CDMA2000 1X supports both voice and data, while CDMA2000 1xEV-DO (Data Optimized) supports data only in 1.25 MHz

of spectrum.

The first commercial CDMA2000 network was launched in Korea in October 2000.More than 20 operators worldwide will be launching commercial 3G networks in 2002, and 30 more are expected in 2003. Today, there are more than 33 vendors with over 170 different commercial CDMA2000 handsets. cdmaOne has been deployed in over 50 countries on five continents, and there are over 120 million subscribers worldwide.

Major new enhancements that CDMA2000 1X is offering include:

- ncreasing voice capacity from 22 TCH per sector per carrier in cdmaOne to 35 TCH per sector per carrier (current phase), increasing to 49 TCH per sector per carrier in 2003/2004 with SMV vocoder (mode2)
- "Always On" peak packet data rate of 153 kbps (current phase), increasing to 307 kbps in 2003/2004 (1xEV-DO delivers a peak data rate of 2.4 Mbps)
- Connectivity to ANSI-41, GSM-MAP, and All-IP networks
- Various bands and bandwidths of operation in support of different operator needs:
- 450, 800, 900, 1700, 1800, 1900, and 2100 MHz (there are no commercial networks in 900 and 1800 Mhz band today)
- Fully backward compatible with cdmaOne systems

- Improved service multiplexing and QoS management
- Flexible channel structure in support of multiple services with various QoS and variable transmission rates

The CDMA2000 standard is being supported by “The Third Generation Partnership Project 2,” or 3GPP2 organization. It is benefiting from the expertise of specialists from Korean, Japanese, Chinese, and North American standards bodies. The ITU approved the CDMA2000 1X, 1xEV-DO, and 1xEV-DV radio access systems as members of the IMT-2000 family of standards.

Using CDMA 2000 benefits Increasing voice capacity and higher data throughput provide a strong incentive for wireless operators to deploy CDMA2000 1X technology. These benefits will make the technology very attractive to potential consumers.

Voice continues to be a major source of traffic and revenue for wireless operators today, while packet data traffic is beginning to emerge as a major source of revenue in upcoming years. CDMA2000 1X is the most spectrally-efficient technology, delivering the highest voice and packet data throughput using the least amount of spectrum for the lowest cost.

CDMA2000 1X supports 35 traffic channels per sector per RF (26 Erlangs/sector/RF) using the EVRC vocoder. The

EVRC vocoder first became commercial in 1999. The next generation SMV vocoder is expected to become commercial in 2003 or 2004. The

SMV vocoder (SMV2 mode) will increase the voice capacity to 49 traffic channels per sector per RF (39 Erlangs/sector/RF), while maintaining similar voice quality to EVRC.

Voice capacity improvement in the forward link is contributed to fast power control, lower code rates (1/4 rate), and transmit diversity (for single path Rayleigh fading). In reverse link, capacity improvement is primarily due to coherent reverse link.

Today’s commercial CDMA20001X networks (phase 1) support a peak data rate of 153.6 kbp. The theoretical maximum data rate supported by the standard is 628 kbps, and it is achieved by having 2 SCH at 307 kbps while FCH supports an additional 14.4 kbps of data traffic. While some of the CDMA2000 ASICs support 307 kbps, handsets delivering this data rate have not become commercial yet. Some of today’s commercial

handsets deliver 153 kbps with an average dat arate of 60 to 90 kbps on 1X and 2.4 Mbps on 1xEV-DO with an average data rate of 650 to 1400 kbps.

Data Gathering

Data gathering or mining from network measurements was performed as follows:

Country	Locations	Operator
Panama	Panama City	Cables & Wireless
China	AnHui Beijing Henan Henan Jiangsu Liaoning Shanghai Sichuan Yunnan	China Unicom
Sweeden		Telia Asa
U.S.A.	Denver, CO. New York City, NY. New York City, NY. St. Louis, MO. Atlanta, GA. Atlanta, GA. Los Angeles, CA. Los Angeles, CA. Stillwater, OK. Miami, FL. Miami, FL. Fort Lauderdale, FL. Port Charlotte - Fort Myers, FL. West Palm Beach, FL. Athens, GA Monterrey-Salinas, CA.	Leap Verizon Sprint PCS Verizon Metro PCS Sprint PCS Sprint PCS Western Cellular US Cellular Sprint PCS Verizon Metro PCS Metro PCS Metro PCS Metro PCS Metro PCS
Norway	Oslo	Telenor ASA
Finland	Helsinky	Telecom
Korea	Seoul	Korea Telecom

Terminal Equipments

The following terminal equipments were used to perform our measurements and testing.

Client	Model	Notes
Kyocera	2255	Data Capable, SMS
Kyocera	KE413/KX414 Phamton	Data Capable, BREW 2.0, SMS
Kyocera	2235	Data Capable, SMS
Kyocera	S14	Data Capable, SMS
Audiovox	CDM 8300	Data Capable, SMS, WAP
Audiovox	CDM 9155	Data Capable, SMS, WAP
Sanyo	PM-8200	Data Capable, SMS, Push To Talk, MMS, WAP
Sanyo	SCP-8100	Data Capable, SMS, WAP
Motorola	V720	Data Capable, JAVA enabled
Motorola	T730	Data Capable, BREW, gpsOne
Samsung	SCH-N255	Data Capable, SMS
Samsung	SCH-A610	Data Capable, BREW, gpsOne, SMS, MMS
Samsung	SCH-A310	Data Capable, gpsOne, SMS
LG	VX-4400	Data Capable, BREW, gpsOne, WAP 2.0, SMS
LG	VX-6000	Data Capable, BREW, gpsOne, WAP 2.0, MMS, SMS
Nokia	2270	SMS

Platform Manufacturer

Manufacturer	Platform	Notes
Ericsson	CDMA 2000 1X EV-DO	1, 2A, 6, G
Motorola	CDMA 2000 1X EV-DO	B, 2A, C
Huawei	CDMA 2000 1X EV-DO	3, XE
Lucent	CDMA 2000 1X EV-DO	A, D, 8
Nortel Networks	CDMA 2000 1X EV-DO	9, K, L
Samsung	CDMA 2000 1X EV-DO	1, 1, 0
ZTE	CDMA 2000 1X EV-DO	A, B, 2A
Airvana	CDMA 2000 1X EV-DO	3, XE, 0, 1
Dalian	CDMA 2000 1X EV-DO	1B, 9, RX
Huanvu	CDMA 2000 1X EV-DO	SS, 1, 2, 6, 7, G
LG	CDMA 2000 1X EV-DO	1B, 9, K, L, 3

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Technology

Protocol	Networks	Notes
IS-95A	None tested	-
IS-95B	None tested	-
IMT-2000	None alone	All networks had developed into EV-DO. Some were developing into EV-DV
IMT-2000 1X EV-DO	Some	Most networks were EV-DO enabled
IMT-2000 1X EV-DV	None tested	Although there were EV-DV networks in operation in the US, Lenswood did not test these.

Key Features

CDMA2000 has unique features leading to increased capabilities of CDMA networks. The increased voice and data capabilities will deliver a wider choice of services for service providers and consumers.

The CDMA2000 physical layer includes several modes of operation. This enables the operators to deploy and configure their networks based on their own needs and local regulations. Several options exist for the band of operation and system bandwidth.

CDMA2000 networks have already been deployed in the 450 MHz, 800 MHz, 1700 MHz, and 1900 MHz bands, and deployments in the 2.1 GHz band are expected in 2003.

CDMA2000 is also standardized to operate in several additional frequency bands, including the 900 and 1800 MHz bands. Spectral efficiency of CDMA2000 permits high traffic deployments in any 1.25 MHz of spectrum.

Initially in the CDMA2000 standards, two spreading rates were specified:
1.25 MHz full duplex bandwidth referred to as “Spreading Rate 1” (SR1), or “IX”
3.75 MHz full duplex bandwidth referred to as “Spreading Rate 3” (SR3), or “3X”
Today the 1.25 MHz configuration, SR1, is being deployed.

The fundamental spreading rate is 1.2288 Mcps for SR1. It requires 1.25 MHz of bandwidth (625 kHz from the center frequency) when the adjacent RF carriers are other CDMA carriers. However, it requires 1.8 MHz of bandwidth (900 kHz from the center frequency) when both adjacent RF carriers are narrow band GSM or TDMA carriers

As mobile phones take the place of wire-line phones, the standby times of phones have become an essential customer satisfaction parameter. CDMA2000 doubled the tandaby

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battery life compared to cdmaOne and its major battery life enhancing features include:

- Quick paging channel operation
- Improved reverse link performance
- New common channel structure and operation
- Reverse link gated transmission
- New MAC states for efficient and ubiquitous idle time operation

CDMA2000 is synchronized with Universal Coordinated Time (UTC). The forward link transmission timings of all CDMA2000 base stations in the world are synchronized within a few microseconds. The base station synchronization can be achieved through several techniques, including self-synchronization, radio beep, or through satellite-based systems such as GPS, Galileo, or GLONASS. The reverse link timing is based on the received timing derived from the first multipath component used by the terminal.

There are several benefits to synchronization of all base stations in a network. The common time reference improves acquisition of channels and handoff procedures since there is no time ambiguity when looking for and adding a new cell in the active set. It also enables the system to operate some

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of the common channels in soft handoff, which improves the efficiency of the common channel operation. In addition, the common network time reference allows implementation of a very efficient "position location" technique such as gpsOne.

The basic frame length is 20 ms divided into 16 equal power control groups. In addition, CDMA2000 defines a 5 ms frame structure, essentially to support signaling bursts, as well as 40 and 80 ms frames, which offer additional interleaving depth and diversity gains for data services. Unlike IS-95 where Fast Closed Loop Power Control was applied only to the reverse link, CDMA2000 channels can be power controlled at up to 800 Hz in both reverse and forward links. The reverse link power control command bits are punctured into the F-FCH or the F-DCCH (explained in later sections), depending on the service configuration. The forward link power control command bits are punctured in the last quarter of the R-PICH power control slot.

In the reverse link, during gated transmission, the power control rate is reduced to 400 or 200 Hz on both links. The reverse link power control sub-channel may also be divided into two independent power control streams, either both at 400 bps, or one at 200 bps and the other at 600 bps. This allows for independent power control of forward link channels.

In addition to the closed loop power control, the power on the

reverse link of CDMA2000 is also controlled through an Open Loop Power Control mechanism. This mechanism inverses the slow fading effect due to path loss and shadowing. It also acts as a safety fuse when the fast power control fails. When the forward link is lost, the closed loop reverse link power control is "freewheeling" and the terminal disruptively interferes with neighboring. In such a case, the open loop reduces the terminal output power and limits the impact to the system.

Finally, the Outer Loop Power drives the closed loop power control to the desired set point based on error statistics that it collects from the forward link or reverse link. Due to the expanded data rate range and various QoS requirements, different users will have different outer loop thresholds; thus, different users will receive different power levels at the base station. In the reverse link, CDMA2000 defines some nominal gain offsets based on various channel frame format and coding schemes. The remaining differenes will be corrected by the outer loop itself.

In parallel to dedicated channel operation, the terminal keeps searching for new cells as it moves across the network. In addition to the active set, neighbor set, and remaining set, the terminal also maintains a candidate set.

When a terminal is traveling in a network, the pilot from a new BTS (P2) strength exceeds the minimum threshold TADD for addition in the active set. However, initially its relative contribution to the total received signal strength is not sufficient and the terminal moves P2 to the candidate set. The decision threshold for adding a new pilot to the active set is defined by a linear function of signal strength of the total active set.

The network defines the slope and cross point of the function. When strength of P2 is detected to be above the dynamic threshold, the terminal signals this event to the network. The terminal then receives a handoff direction message from the network requesting the addition of P2 in the active set. The terminal now operates in soft handoff.

The strength of serving BTS (P1) drops below the active set threshold, meaning P1 contribution to the total received signal strength does not justify the cost of transmitting P1. The terminal starts a handoff drop timer. The timer expires and the terminal notifies the network that P1 dropped below the threshold. The terminal receives a handoff message from the network moving P1 from the active set to the candidate set. Then P1
strength drops below TDROP and the terminal starts a handoff drop timer, which expires after a set time. P1 is then

moved from candidate set to neighbor set. This stepby-step procedure with multiple thresholds and timers ensures that the resource is only used when beneficial to the link and pilots are not constantly added and removed from the various lists, therefore limiting the associated signaling.

In addition to intrasystem, intrafrequency monitoring, the network may direct the terminal to look for base stations on a different frequency or a different system.

CDMA2000 provides a framework to the terminal in support of the inter-frequency handover measurements. This consists of identity and system parameters to be measured. The terminal performs required measurements as allowed by its hardware capability.

In case of a terminal with dual receiver structure, the measurement can be done in parallel. In case of a terminal with a single receiver, the channel reception will be interrupted when performing the measurement. In this case, during the measurement, a certain portion of a frame will be lost. To improve the chance of successful decoding, the terminal is allowed to bias the FL power control loop and boost the RL transmit power before performing the measurement. This method increases the energy per information bit and reduces the risk of losing the link in the interval. Based on measurement reports provided by the terminal, the network then decides whether or not to handoff a given terminal to a different frequency system. It does not release the resource until it receives confirmation that handoff was successful or the timer expires.

This enables the terminal to come back in case it could not acquire the new frequency or the new system.

Transmit diversity consists of de-multiplexing and modulating data into two orthogonal signals, each of them transmitted from a different antenna at the same frequency. The two orthogonal signals are generated using either Orthogonal Transmit Diversity (OTD) or Space-Time Spreading (STS). The receiver reconstructs the original signal using the diversity signals, thus taking advantage of the additional space and/or frequency diversity.

Another transmission option is directive transmission. In this case, the base station directs a beam towards a single user or a group of users in a specific location, thus providing space separation in addition to code separation. Depending on the radio environment, transmit diversity techniques may improve the link performance by up to 5 dB.

The CDMA2000 forward traffic channel structure may include several physical channels:

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- The Fundamental Channel (F-FCH) is equivalent to functionality Traffic Channel (TCH) for IS-95. It can support data, voice, or signaling multiplexed with one another at any rate from 750 bps to 14.4 kbps.
- The Supplemental Channel (F-SCH) supports high rate data services. The network may schedule transmission on the F-SCH on a frame-by-frame basis, if desired.
- The Dedicated Control Channel (F-DCCH) is used for signaling or bursty data sessions. This channel allows for sending the signaling information without any impact on the parallel data stream.
- The reverse traffic channel structure is similar to that of the forward traffic channel. It may include R-PICH, a Fundamental Channel (R-FCH), and/or a Dedicated Control Channel (R-DCCH), and one or several Supplemental Channels (R-SCH). Their functionality and encoding structure is the same as for the forward link with data rates ranging from 1 kbps to 1 Mbps (It is important to note that while the standard supports a maximum data rate of 1 Mbps, existing products are supporting a peak data rate of 307 kbps.)

The traffic channel structure and frame format are very flexible. In order to limit the signaling load that would be associated with a full frame format parameter negotiation, CDMA2000 specifies a set of channel configurations. It defines a spreading rate and an associated set of frames for each configuration.

The forward traffic channel always includes either a fundamental channel or a dedicated control channel. The main benefit of this multichannel forward traffic structure is the flexibility to independently set up and tear down new services without any complicated multiplexing reconfiguration or code channel juggling. The structure also allows different handoff configurations for different channels. For example, the F-DCCH, which carries critical signaling information, may be in soft handoff, while the associated F-SCH operation could be based on a best cell strategy.

One key CDMA2000 1X feature is the ability to support both voice and data services on
the same carrier. CDMA2000 operates at up to 16 or 32 times the FCH rate, also referred to as 16x or 32x in Release 0 and A respectively. In contrast to voice calls, the traffic generated by packet data calls is bursty, with small durations of high traffic separated by larger durations of no traffic. It is very inefficient to dedicate a permanent traffic channel to a packet data call. This burstiness impacts the amount of available power to the voice calls, possibly degrading their quality if the system is not engineered correctly. Hence, a key CDMA2000 design issue is assuring that a CDMA channel carrying voice and data calls simultaneously does so with negligible impact to the QoS of both.

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Supplemental Channels (SCHs) can be assigned and deassigned at any time by the base station. The SCH has the additional benefit of improved modulation, coding, and power control schemes. This allows a single SCH to provide a data rate of up to 16 FCH in CDMA2000 Release 0 (or 153.6 kbps), and up to 32 FCH in CDMA2000 Release A (or 307.2 kbps). Note that each sector of a base station may transmit multiple SCHs simultaneously if it has sufficient transmit power and Walsh codes. The CDMA2000 standard limits the number of SCHs a mobile station can support simultaneously to two.

This is in addition to the FCH or DCCH, which are set up for the entire duration of the call since they are used to carry signaling and control frames as well as data. Two approaches are possible, individually assigned SCHs, with either finite or infinite assignments, or shared SCHs with infinite assignments.

For bursty and delay-tolerant traffic, assigning a few scheduled fat pipes is preferable to dedicating many thin or slow pipes. The fat-pipe approach exploits variations in the channel conditions of different users to maximize sector throughput. The more sensitive the traffic becomes to delay, such as voice, the more appropriate the dedicated traffic channel approach becomes.

In CDMA2000 the most common way to allocate a high-speed link to a user is to assign an SCH to the user. During the assignment, the data being transmitted on the SCH is directed to only that particular user. This type of approach requires signaling, which introduces delays. Current implementations require approximately five frames, that is, 100 msec of latency. If the user is moving at a relatively slow speed, it would be possible to change the rate from one SCH assignment to the other in a manner so as to adapt to a changing path loss or shadowing. For example, if the user is moving away from the cell, the assigned SCH rate may be gradually reduced with time as the link quality deteriorates. The delay introduced by signaling, however, is intolerable if one wants to schedule SCH transmissions based on faster changing characteristics of the channel such as Rayleigh fading.

Assign a user an SCH with an infinite duration. This will ensure that the user will not need to go through the signaling process every time it needs to transmit data using the SCH, and no interference will be generated to other users when no data is transmitted because the F-SCH may have discontinuous transmission at any time, for any duration.

However, it is a very inefficient approach because the Walsh space is reserved to a user who might be idle most of the time.

Assign a new SCH for a very large duration to users that need

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to download a large amount of data, and release that SCH as soon as the user becomes idle. This approach is more efficient than Option A because idle users do not occupy Walsh space for the entire duration of the call. However, every download must bear with an additional delay due to signaling. Another advantage of this approach is that the base station does not need to guess how much data the user has to download; it just de-assigns the SCH when the user is idle. The associated drawback is that the base station does not know when the assignment will end, which makes it hard for the base station to plan the sharing of the sector capacity. There is additional signaling necessary to de-assign SCHs when users become idle.

Same as Option B, but in order to let the base station conduct some scheduling, the assignments are stopped before the user is done downloading; it can resume later when the base station gives it a new assignment. This approach allows the base station to do some scheduling as with Option B, with the advantage that the duration is not predetermined. There is again additional signaling involved to de-assign an SCH before it becomes idle; the base station scheduling chops down the previous transmission in pieces.

Assign a new SCH for a predetermined duration to users that need to download a large amount of data. This approach is similar to Option C with the difference that since assignments have a pre-determined duration, no signaling is needed to end the SCH assignment. However, the associated limitation is that the scheduling is less flexible (the duration is fixed).

The Shared SCH approach enjoys the benefits of both methods C and D without their drawbacks.

In this approach, a particular SCH with one fixed rate is assigned to a group of users. This means that all users in the group keep processing the same SCH indefinitely. The key is that the data transmitted on the SCH may be directed to only one of them. When
the data is directed to a particular mobile station, it should be able to decode a good frame. At the same time, all other users in the same group will decode an erasure and nothing will be passed to RLP, which will behave as if the frame was a discontinuous transmission.

This method enjoys the benefits of both previous methods because as long as a user can demodulate the same rate, no signaling is involved. At the same time, the same Walsh space can be shared among an arbitrarily large number of users. Moreover, with this type of SCH assignment being virtually signaling-free, a scheduling algorithm can change the scheduled user every frame (20 ms). This allows channel-sensitive scheduling. However, with the frame size being 20ms, the fading process must stay reasonably slow, so

Rayleigh fading-sensitive scheduling is only possible for very low speeds.

The signaling-free assumption is true as long as the user supports the rate of the SCH. If channel conditions change so that the rate cannot be supported anymore, the user would have to be moved to a different SCH group with a lower rate, in which case signaling would be involved. As long as a user can support the SCH rate, no signaling is necessary.

CDMA2000 provides the option of using either turbo coding or convolutional coding on the forward and reverse SCHs. Both coding schemes are optional for both the base station and the mobile station, and the capability of each is communicated through signaling messages prior to the setup of the call. In addition to peak rate increase and improved rate granularity, the major improvement to the traffic channel coding in CDMA2000 is the support of turbo coding at rate 1/2, 1/3, or 1/4. The turbo code is based on 1/8 state parallel structure and can only be used for supplemental channels and frames with more than 360 bits. Turbo coding provides a very efficient scheme for data transmission and leads to better link performance and system capacity improvements. In general, turbo coding provides a performance gain in terms of power savings over convolutional coding. This gain is a function of the data rate, with higher data rates generally providing more turbo coding gain.

Frequency Bands

The frequencies for IMT-2000 were allocated in two phases, the first made in 1992 when IMT-2000 began development, and the second set at the recent conference. The bands that had initially been identified in 1992, on the basis of which licensing has already been made or is under way in many parts of the world, remained unchanged. Around 100 licenses are expected to be awarded worldwide by the year 2002. These bands are 1885-2025MHz and 2110-2200MHz. The additional bands identified for the terrestrial component of IMT-2000 are: 806-960MHz, 1710-1885MHz and 2500-2690MHz. All bands globally identified for IMT-2000 have equal status.

The only IMT-2000 system that was fully operational in the first quarter of 2002 was NTT DoCoMo's 3G service which went online in October 2001. This runs to the ITU' standards (W-CDMA) and spectrum frequency (1.9-2GHz). Other operators plan to start commercial services by the end of 2002 when handsets become available. These include Telenor Mobile and Netcom in Norway; Sonera and Radiolinja in Finland; and Orange Sverige, Europolitan, Hi3g, and Tele2 in Sweden.

In other areas operators have reported field trials and experiments in W-CDMA across these frequencies. These include: Orange SA in France; Telecom Italia Mobile in Italy; Manx Telecom on the Isle of Man; and Monaco Telecoms in Monaco. Operators in Japan and the Republic of Korea have begun to implement systems on other spectra for a similar time-scale.

Frequency Channel Allocation

Frequency	Carrier
800	Verizon, SPRINT PCS, China Unicom, Smartcom. AllTel. Commnet
1800	Verizon, Western Wireless, Commnet
1900	SPRINT PCS, China Unicom, SmartCom PCS, Verizon, US Cellular, Metro PCS, Leap, MTS Mobility, AllTel
2100	China Unicom

Protocol Anlaysis in Networks

The analysis of protocol recordings in UMTS is much more complex than in GSM-/GPRS-networks for a number of reasons:

1. No Pre-configured Timeslots or Channels for Signaling Messages and User Data.

First of all, UMTS networks use the packet-switched ATM protocol to provide for the highest possible flexibility in resource allocation. Using so called cells on a serial link with a payload of just 48 octets, ATM is capable to share an E1, T1 or STM-1 link among a literally unlimited number of users or, at the other extreme, to provide all resources to only a single user.

With respect to ATM, the term “user” refers to virtual paths and channels, which in turn, are only bearers for the higher UMTS-layers. This and other advantages of using ATM are good for UMTS but unfortunately, ATM does not come with pre-configured timeslots or dedicated channels. This issue appears to be trivial, but finding signaling messages on the Iub- or Iu-interfaces in a UMTS-network without knowing

where to look for them is impossible.

As a matter of fact, ATM is not providing dedicated timeslots or channels for neither signaling messages nor user data. Rather, ATM is providing virtual channels and virtual paths that need to be configured by the higher protocol layers of UMTS upon putting an interface and a network node (e.g. RNC, NodeB or cell) into service.

Another way is to use an Iub Automatic Configuration application; the configuration task becomes completely automated. Expert software can automatically configure all the logical links required to monitor NBAP, ALCAP for each Node B under observation and RACH, FACH, PCH for each cell under observation. The ability to perform automatic analysis on the Iub-interface will give users a large advantage. It will automatically track the configured channels and display the configuration of each channel in plain text. The protocol tester locks to these channels and allows tracing of the upcoming signaling messages.

2. Limited Possibility of HEX-Trace Analysis

Another challenge for the experienced GSM-protocol expert is the fact that some higher layer UMTS-protocols like RANAP, NBAP or RRC do not only use ASN.1 encoding rules but for optimization, apply the so called Packed Encoding Rules

(PER).

Each parameter is encoded using a unique tag, followed by a length indication and finally the parameter value. Of course, ASN.1 provides many more capabilities, but what is important to consider is the fact this type of encoding is a waste of radio resources.

Using the example parameter “MyInteger”, this parameter can take on only two values: ‘0’ and ‘1.’ Besides, the presence of the parameter “MyInteger” shall be mandatory in the message to be encoded. Just applying ASN.1 basic encoding leaves us with three octets to be transmitted. But after applying PER, only a single bit (‘0’ or ‘1’) needs be sent. The benefits of PER are obvious; however, one may still miss the possibility to easily match hexadecimal values to protocol tester mnemonics.

In UMTS, it is therefore no longer trivial to translate hexadecimal recordings into mnemonics just by using the respective specifications. One needs a PER decoding and therefore, a protocol tester for almost any kind of protocol analysis. The hexadecimal debugging alternative is cumbersome for all protocols that apply the PER.

3. Following a Single Call Flow

Terminal equipments analysis for IMT 2000 Network platform

Protocol analysis in UMTS, compared to GSM, has an essential impact on the very basic task to follow a single call setup or registration or PDP context activation. To recall some of the very basic questions:

- Which parameters tie the various messages to each other?
- Which call was successful and which one wasn’t?
- Did this transaction fail because of errors in the previous parameterization?

Terminal Equipment Test – High Throughput and Accuracy.

Most of the manufacturers that we visit are struggling to keep up with demand for the cellular phones and for the infrastructure equipment needed to cover more regions and handle increased cellular phone traffic. Furthermore, new technology is being integrated in the cellular phones to increase their functionality. Phones are or will be integrated with computers and personal digital assistant capabilities. They will link to the Internet and will link to computers. The constant influx of new technology and increasing demand for the new products creates a tremendous challenge in terms of manufacturing capacity.

Often, one of the bottlenecks in a telecommunication equipment plant is found in the production test area. One way to break the bottleneck is to replace or upgrade test equipment so that more tests can be done in less space at higher speeds. Cellular phones are a case in point.

Production tests on cellular phones fall into three broad categories: board level tests, operational adjustments, and final tests. In the first category, some of specific tests and adjustments include internal voltage levels, current drains, audio functionality, and power consumption.

Adjustments include bandwidth, carrier frequency, RF output, receiver sensitivity, and audio input/output levels. Typical tests in the third category are simulated calls and overall RF performance.

Usually, these measurements and adjustments require the integration of several pieces of test equipment, along with a PC and associated test software. Typically, all the equipment is installed in a standard 19” wide rack mount.

As production and test needs expand, the ideal situation is to add another test system to the rack, which could handle a second production line or increased throughput on the existing line. Depending on the size of existing test units, this may or may not be possible. So, a major consideration when specifying a new piece of equipment is its footprint and/or rack mount vertical height. By carefully selecting instruments

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for size as well as functionality, the test engineer can leave enough rack space for future expansion and higher throughput.

When considering new or replacement test equipment, look for instruments that have multiple functions in a single package. Some instrument manufacturers are combining functions formerly supplied by several pieces of equipment in a test system. They are also providing interfaces that are compatible with component handlers and other commonly used test fixtures.

Manufacturers and users alike are recognizing the benefits of such packages, including reduced rack space, simplified setup, and less programming time.

One example is the Keithley Model 2400 SourceMeter instrument, a new class of instrument introduced as a high-speed test solution for large volume component and equipment manufacturers. This unit combines precision voltage and current sources with a high resolution digital multimeter and measurement firmware to provide high throughput as well as a simplified package.

Throughput affects an instrument’s ROI, as does the number of test functions supplied at a given price. But as alluded to earlier, simplified packaging can reduce system integration costs, the expense of configuring equipment, and costs associated with the development of test programs. All of these can reduce the cost of ownership. Where a new, more productive instrument breaks a testing bottleneck, overall production capacity is increased and ROI enhanced for the entire line.

Still, a complete COST/ROI picture involves ongoing operation and maintenance costs and even training expense. Thoughtful instrument designs aimed at production test environments that make it easy for operators to learn and use the equipment also increase productivity and drives down the costs of testing and ATE ownership. So, look for ease-of-use features, such as front panel programming, a display with DMM simplicity, and built-in functions like pass/fail and comparative measurements.

In telecommunications equipment testing, the switching system often is the key to measurement integrity and high productivity. Achieving high throughput in a small package requires the careful selection of a switching system with appropriate I/O cards. Fortunately, the shrinking size and increasing functionality of ICs and other electronic modules are giving designers the opportunity to reduce package size.

This is not to say that a new instrument design automatically results in a smaller footprint with better performance. The instrument designer also has to consider the tradeoffs

between speed, accuracy and cost. When making rapid multiple measurements with a switching matrix, it takes time for a scanning relay’s contacts to settle (to achieve required accuracy), with contact bounce lasting about 3 to 6 milliseconds for dry reed relays. However, instrument manufacturers can reduce this time and increase switch density by using electronic relays that are faster and smaller, albeit more costly.

For example, Keithley’s Model 7001 High Density Switch System takes up only a half rack space and has two slots that will accept a wide variety of switching cards for signals up to 1.3GHz. This package supports twice as many channels (80) in half the space compared to earlier switching systems.

To reduce the time it takes to get a production test system up and running, instrument builders are giving heavy consideration to how easily instruments and switching devices can be integrated. One goal is to make it easier for the user to set up and program instruments and switching systems so they work together more effectively. Instrument design considerations include such things as flexible trigger I/O, ability to handle a wide range of signal types and levels, multiple electronic connectors, and compatible data communication protocols.

Being able to control measurements with a variety of trigger inputs provides a great deal of flexibility in setting up the test configuration. In this regard, the specifier should consider how easily the various system components can be integrated, including compatibility of physical connections between the trigger source and the triggered instruments. Also, the test engineer should be able to program the instruments and switching system easily to work together with appropriate trigger levels, signal types, delays, etc.

Making it easy to modify or expand test programs as requirements change is another feature needed in telecommunication test equipment. Many of today's instruments offer the user two programming options: front panel programming with a graphic status display or programming from a PC controller over an IEEE-488 bus using the SCPI (Standard Commands for Programmable Instruments) command set. In simpler applications, an RS-232 connection maybe sufficient, but it's better when an instrument offers both types of connectivity and commonly used protocols to go with them.

Although a larger set of functions provides greater instrument flexibility, it may also mean higher cost if functions are added indiscriminately. Again, tradeoffs must be managed. On one hand, having an instrument with features for a complete set of functional tests, and the capability to meet changing requirements, is highly desirable. While more features may mean fewer instruments for a given set of tests, fewer features may mean lower cost. Understanding the test environment

allows an instrument designer to build in the right set of features at the lowest possible cost.

For example, designing a programmable voltage supply for computer-controlled testing can eliminate the need for front panel controls and readouts. These functions are supplied by the computer and its display. The result is a substantially lower power source cost compared to those with comparable precision and all the usual manual controls and digital displays.

Often, production test equipment operates around the clock, seven days a week, under harsh manufacturing conditions. To provide reliable test capacity in this environment, instrument designers are improving the physical packaging of instrument circuits in key areas. For example, surface mount technology and circuit boards with no more than two layers can hold down manufacturing costs and improve reliability without sacrificing accuracy or other performance areas. Typically, instrument manufacturers are shooting for mean time between failures (MTBF) measured in thousands of hours.

Although measurement integrity and reproducibility are important in a production testing environment, instrument accuracy is almost taken as an article of faith. When considering new test equipment purchases, production and test engineers quickly turn their attention to productivity, space, and cost issues. Instrument manufacturers, such as

Keithley, are addressing these issues by introducing new instruments that are easier to use, with smaller footprints, higher throughput, and better reliability than ever before.

Terminal Configuration

Software defined radio technology is one of the key technologies for systems beyond IMT-2000, promising the flexibility required for total mobility and, from the user's perspective, pervasive access to services. Such technology will herald a new generation of flexible and reconfigurable user terminals, able to detect their operating environments and adapt to make optimum use of the available resources, downloading appropriate software to the terminal when necessary to achieve interoperability, increased performance and security or improving spectrum utilisation efficiency. It is clear that there are many similarities between SDR technology and personal computers, both providing flexible platforms for execution of software that is potentially sourced from multiple vendors. The difference lies in the fact that SDR technology applies software implementation to functions normally performed in hardware (or embedded into operating system kernels), to increase the level of flexibility. This will enable lower terminal manufacturing costs through economies of scale and will open up new revenue generating possibilities (via core software download) an of course fulfill the user

deman for higher services availability and innovative new services.

Similarities between the telecommunications and the computing, or IT, domains can be seen elsewhere. Much has been written about the convergence of these domains and perhaps the most obvious sign of this is the adoption of the Internet Protocol (IP) in core telecommunications networks. One of the key challenges that faces the telecommunications domain as we look beyond IMT-2000 is how to integrate all of the existing technologies, and provide support for future technologies, without restricting innovation. The IT domain can provide a solution here, through integration of heterogeneous technologies, especially when IP is the network protocol. This is leading researchers to consider other IT-oriented solutions to telecommunications domain problems.

Due to the limitations of the terminal resources, it is assumed that the reconfiguration functions are distributed in the terminal, access and core network. For instance, the detection and monitoring of access networks for the SDR terminal may be assisted by the network.

Service discovery protocols will be investigated as the means to provide network availability information to the terminals. Mode selection (i.e. selection of appropriate access network) and software download requires complex interactions between the distributed functions involved in the reconfiguration process. These functions can be deployed at different locations and can be supported by different types of hardware platforms. Therefore, wireless middleware will be proposed in section 4 to manage the interactions between the reconfiguration functions.

Terminal Architecture and Flexible Protocol Stack

The protocol stack of a SDR terminal device is the system component where multiple system domains meet. Coming top down from an application's point of view, the upper layers of a protocol stack are implemented in software (mostly object-oriented designs, executed on embedded microprocessors). Further down in the ISO OSI layer model, around layer two or three, we will find digital signal processors, executing parts of the protocol stack software, which has been implemented either in C or in highly optimized assembler code. Deep down on layer 1 of the protocol stack, custom hardware (ASICs, FPGAs) is used to execute physical layer processing on digitized data streams. At the very bottom of the physical layer of a SDR protocol stack, the analogue hardware can also be considered to be reconfigurable or at least configurable by parameters.

The goal of work on flexible protocol stacks is to provide an open protocol stack architecture framework that is extensible

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and can be used for supporting different radio access network technologies with different protocols and protocol features. In particular the architecture is proposed to allow the support for a range of reconfigurable (SDR) terminals with different capabilities that can support only one mode in some cases, or more likely many modes simultaneously. This requires that the architecture can support dynamic insertion and configuration of protocol modules from different vendors in a common manner taking into account the resources and capabilities of the target devices.

Previous work on flexible and adaptable protocol stacks has not considered the requirements from all scenarios. One approach that has been proposed allows reconfiguration by configuration parameters providing limited flexibility but good performance. Alternatively, reconfiguration of generic protocol elements can enable a larger degree of flexibility using a standardized software engineering methodology.

Other approaches provide a high degree of flexibility by using run-time software module extensibility (such as provided by the Java language), but compromise resource requirements and performance as a result. Open platforms can provide more flexible solutions as described, and the aim of work on a flexible protocol stack framework is to provide an open architecture for protocol stack implementation.

However, none of these previous approaches address flexible fine grain reconfiguration of protocol stacks that could potentially utilize a mixture of native code, Java or indeed programmable logic implementations. This requires an overall conceptual architecture that can allow protocol stacks to be extended and modified to best suit the performance requirements and underlying execution environments available supporting:

• Multiple CPU design: It must be possible to use more than one CPU in protocol stack's data processing. Even heterogeneous processing units (e.g.microcontroller, DSPs, etc.) should be supported.

• Dynamic reconfiguration of protocol stacks during active communication sessions: It must be possible to introduce new protocol stack modules or remove no longer used modules during stack run-time.

• Upgrading of protocol behavior to support optional features,

• Adding extra functionality into existing protocol stacks or replacing complete stacks,

• Download of third party protocol stacks or protocol stack modules,

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• Optimization of protocol stack behavior depending on the context of the terminal,

• Operation of multiple stacks simultaneously.

To support the above requirements mechanisms and features are required such as:

• Hardware description language: To enable the download of protocol stack configurations that are optimized to the capabilities of the underlying hardware and software execution environments.

• Validation: Because of high security requirements placed on mobile terminals, it is necessary to validate the actual protocol stack configuration as well as the implementation of the protocol stack software components. Proper validation can help to prevent the execution of malicious code, i.e. turning the device into a so called rogue terminal, harming equipment and / or network.

• Validation can be carried out in a three stage process, composed of off-line validation (virtual terminal simulation in network infrastructure), on-line validation (e.g. simple, rule based validation of protocol stack configuration), and run-time validation (e.g. introduction of hardwired assertions into protocol stack execution environment)

• Inter-CPU communication: An efficient mechanism must be provided for the exchange of data between heterogeneous processing domains.

• Reliability and integrity protection: New software modules that get introduced into the protocol stack must be reliable and must not harm the system.

• Battery power efficiency: Because SDR terminals are battery-operated, an optimized protocol stack architecture is needed, which consumes as little energy as possible.

The requirements for protocol stack reconfiguration have been derived from key scenarios, which were identified:

1. Session Handover Scenarios:
- Maintenance of security context during mode changes

2. Inactive Session Scenarios:
- Deletion of software
- Software upgrade

3. Active Session Scenarios:
- Optimization of protocol stacks to context of terminal
- Ad-hoc networking
- Addition of mode detection software

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4. Initiating Session Scenarios:
- Incoming session on different mode
- Reconfiguration during session initiation

Initiating session scenarios were identified as being particularly important. They stem from the requirement for a terminal to be highly available and allow sessions to be initiated regardless of the terminal state and mode of operation. This includes the interruption of other active sessions and even termination of active sessions to support new sessions (depending on the user preferences).

These scenarios present some important implications on protocol stack flexibility concerning handling of session initiation, particularly when incoming sessions result in a need to reconfigure to a new mode in order to support that session. For example, a terminal in a GPRS mode of operation could have an incoming session request that is best supported on a WLAN mode.

The session could be set-up first in GPRS mode and then handover could take place to WLAN mode or alternatively, the terminal could reconfigure to WLAN mode prior to or during the session set-up process.

Other implications of the requirements for protocol stack

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flexibility are that there are occasions in which protocol stacks need to be reconfigured during active sessions. This is not only to provide optimization of performance, but also to allow addition of new functionality such as enhanced mode identification and monitoring features. Updating of protocol stacks is necessary, but the frequency of occurrence of this process depends on the detailed user scenarios and whether the optional features are standardized across multiple operators and networks or whether new optional features are generated by each operator. The additional of protocol features need to be supported by mechanisms to allow the inserting and testing (either in-situ or in a virtual environment) of new modules.

The requirement to download third party protocol stacks is not apparent from the scenarios considered. However, there is obvious need to ensure that the protocols are compatible. There is also a requirement that the security protocols provided in the terminal are compatible with those in the corporate infrastructure to allow roaming from private to public domains and that the protocol stack flexibility does not compromise the security.

The performance enhancement of protocols by an adaptation layer or other adaptation concepts is attractive and can have large performance benefits. Therefore, this implies that the protocol stack should at least be flexible enough to allow the

insertion of protocol layers that can be configured to optimize the performance obtained to that required by the services depending on what air interface technologies are available.

The operation of multiple protocol stacks on the same terminal is required when there are multiple radio modules or a reconfigurable radio module that can support more than one mode of operation at the same time. There may also be a need if multi-homing is being emulated over the same air interface, or when the air interface is being switched between radio access network technologies. The security issues associated with simultaneously operating different stacks have been highlighted and this is an important implication of this requirement.

A framework has been implemented in order to examine the performance and potential benefits of the protocol stack reconfiguration supporting the above requirements. The framework utilizes Generic Protocol Interfaces (GPI) and Generic Service Access Points (GSAP) in order to allow the composition of different protocol stack configurations. Each GSAP instance operates in a separate thread or process of execution to allow protection between different stack instances and also permits the reconfiguration of stack instances during run-time. The framework also supports the ability to utilize proxy GSAP instances which enables virtual devices or protocols to be realized, which is particularly important

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for session initiation scenarios, as described in the following example.

Consider an example implementation of the protocol stack framework which supports two radio access technologies (Bluetooth and 802.11). In the current implementation the 802,11and Bluetooth radio modules are commercial off-the-shelf devices and so some of the protocol related functionality is implemented within the devices and cannot be reconfigured. However, in a future SDR implementation all of these protocols can be dynamically configured. However, in spite of this the framework operation can still allow some of the key scenarios, identified above, to be investigated.

In the scenario there are two applications providing services to the user. Application 2 is initially inactive and application 1 is being supported over the 802.11 mode. In this implementation the Intelligent Routing Layer (IRL) determines which modes are used by each service. Bluetooth mode monitoring is performed using a proxy GSAP instance to reduce the power consumption of the terminal device. The peer proxy GSAP entity interaction is either supported over IP based transport or over link layer encapsulation depending on the location of the peer proxy entity. At a later point in time Application 2 becomes active and the best mode to support the service is selected as Bluetooth, but initially the connection is established using the proxy GSAP over the 802.11 mode to prevent disruption

of the existing active application service (assuming both 802.11 and Bluetooth modes are mutually exclusive). When the application 1 terminates the mode supporting application 2 is seamlessly switched over to Bluetooth without a need to establish a connection as it already established using the proxy GSAP.

The scenario described above has been extended with the incorporation of a Generic Virtual MAC layer beneath the IRL. This has allowed the exploration of performance optimization during active sessions scenarios. The performance optimization that can be performed is selection of different MAC and also ARQ schemes to best suit the context of the terminal device, both in terms of resources available (processing and available battery power) and the other devices sharing the same radio resources.

Services Techniques

Service discovery protocols reflect the propagation of networking, and connectivity in general, from the traditional realms of the office and campus into homes and the personal area. With this change in environment comes a change in the number and diversity of users and a corresponding multiplicity of devices and services. The burden of administration increases too and passing this on to the user is neither desirable nor realistic. Many service discovery protocols and frameworks are therefore emerging that enable the automatic advertisement and discovery of and, in some cases, interaction with, services in the IT domain.

Taking an abstract interpretation of the term ‘service’, it is proposed that service discovery protocols could be used as an assisted method for network detection and monitoring for reconfigurable terminals. The main advantage of this approach is the potential of significantly more efficient network detection and monitoring procedure. Conventional methods utilized by current terminals involve scanning in known frequency ranges for recognized energy patterns and performing the appropriate procedure for camping on to the network.

While performance is considered acceptable for current systems (e.g. a maximum of 30s is specified by the GSM standard), the requirements on future terminals will be significantly stricter. Flexible multi-mode terminals will be able to operate on a wide variety of networks and network technologies, which will have a direct impact on the time taken to detect and monitor available networks.

Moreover, the spectrum for current systems is statically assigned; consequently, terminals are aware of which network technologies can be found in which frequency ranges. However, in the future it has been proposed to adopt

a more efficient approach of allocating spectrum to systems on a dynamic basis. This adds a considerable degree of complexity to the network detection and monitoring procedure for reconfigurable terminals; they will be required to detect and monitor a variety of networks that could be deployed almost anywhere in the spectrum, hence the requirement for a more efficient solution.

It is proposed to create a service for each available network at a particular point of measurement. The service contains identification and monitoring information which can be queried and obtained by terminals. Interactions with the services are realized using service discovery protocols. There are a number of implementation choices in realizing such a service. Perhaps the most fundamental choice is whether to adopt a client-server or peer-to-peer model. The former involves a centralized service that pools network availability and monitoring information from subscribed terminals and provides collated reports on request to client terminals. The latter involves terminals obtaining network availability and monitoring information directly from other donor terminals. The peer-to-peer model can either be physically distributed, with communications via network infrastructure, or on a localized basis utilizing short-range, ad hoc links.

The service model and associated operating environment will have an effect on the requirements for the protocols utilized. Service discovery protocols have been developed and optimized for wired local area deployment; typical usage scenarios include the advertisement and discovery of printer services. Such operating environments are characterized by high bandwidths, low propagation delays, low bit/packet error rates and reliable connectivity. Conversely, the service models utilize wireless links, with comparatively low bandwidth, high propagation delays (especially in the client-server case), high error rates and unreliable connectivity. It is therefore required that the service discovery protocols:

• Are sufficiently lightweight so as not to cause undesirable network loading;

• Are scalable to avoid causing network instability;

• Are resilient to delays inherent in wide area networks;

• Are resilient to errors introduced during transmission over wireless links and through mobility (handoff, fades etc.);

• Are resilient to discontinuities of service inherent in the use of wireless connectivity;

• Support event-based operation, enabling terminals to be efficiently notified of service changes, rather than insisting on the terminals frequently polling to check for changes.

Security must also be addressed. This is not so important for blind methods of network detection and monitoring, as these rely on the terminal undertaking to obtain network information by itself. Assisted methods, however, involve the terminal obtaining network information from another entity. In this case, it is important to ensure that the information can be verified as being correct (operators will be particularly keen to ensure that their network coverage and performance is not being misrepresented).

Therefore, service discovery protocols have the following security-related requirements:

• The protocols used to deliver network availability or monitoring information to terminals should provide security mechanisms to prevent malicious alteration or interception of reports;

• The protocols should also provide mechanisms to prevent the malicious advertisement of a bogus service purporting to offer network availability reports.

Finally, by effectively moving the network monitoring process up the protocol stack, we can conceive of highly detailed network availability reports. Conventional, lower-layer mechanisms for network monitoring have the distinct disadvantage that they yield minimal information.

The GSM BCCH, for example, supplies data such as the operator identifier, location area, frequencies of the neighboring cells and the channel control parameters.

Network reports delivered via higher-layer service discovery protocols can be significantly more detailed, including not only quantitative parameters such as bit error rates but qualitative parameters such as level of QoS or costs. Therefore, service discovery protocols are required to support extensible format for reports or service templates.

Wireless middleware for SDR Terminals

Middleware is widely used in the Information Technology domain and is best defined as the connectivity software that integrates distributed and legacy systems. Middleware has evolved to fulfill new requirements based on the increasing demand for mobile computing and new kinds of applications such as multimedia or mobile ecommerce.

Before the actual terminal reconfiguration, several procedures between the terminal and the network have to be performed. In the case of mode change (e.g. the user moves from the white cell to the grey cell), the TRUST/SCOUT project has defined three procedures: mode detection and monitoring

(see previous section), distributed mode negotiation and mode selection and software download. The two latter procedures involve interactions among distributed reconfiguration entities and resources (i.e. SPRE/SDRC servers, profiles repository, distributed SW repositories, etc.). Therefore, it is proposed to apply a middleware layer on the SPRE/SDRC servers to facilitate the interactions between the reconfiguration components and resources. CORBA or EJB are possible component-based midddleware candidates for the implementation, providing the bridge between client terminals and repositories. For these technologies, containers are used to manage the execution of the reconfiguration components on the platforms (i.e. SPRE/SDRC servers) and provide integrated services to these components such as transaction or security.

Reliable reconfiguration of SDR Terminals

The middleware transaction service has been widely used for distributed database management, e-commerce and workflow applications where the participating components are at fixed locations and are connected via high bandwidth. Transactions are useful to build reliable distributed applications: If one component fails to perform a task or operation, the transaction is rolled back and other participating components roll back their changes, especially in the case of the change of persistent data in databases.

In the context of SDR terminals, transactions can be used to provide reliable terminal reconfiguration. It can be identified following reasons for reconfiguration failures:

• The terminal may be disconnected or no more reachable during the reconfiguration.

• No appropriate radio access network is available according to application requirements, user preferences, terminal and network resources.

• Inappropriate or incompatible software is downloaded to the terminal. This can be verified using the certification information provided by a certification authority.

• Wrong validation of the new configuration of the terminal.

The transaction will commit only after the successful completion of the terminal reconfiguration. In such a case, profile information data are updated in the repository and the user might be charged for the reconfiguration service. In the case of reconfiguration failures, the terminal will have to keep the old configuration.

Security Consideration

Security for reconfigurable terminals is quite a challenging research problem. The acceptance of SDR technology for the commercial industry will greatly depend on the availability of security technologies and standards for reconfigurable terminals. The regulatory bodies are also aware of the importance of security on different areas: Control of interference level, equipment approval, integrity and security of the equipments.

The OMG has defined the CORBA security service that provides key features such as authentication, authorization and access control, definition of security policies to application developers and security administrators. It can be identified the following security objectives for reconfigurable terminals that can be handheld by the OMG security services:

• Management of access control to resources such as software/profiles repositories based on specific security policies.

• Management of access control to reconfiguration and software downloading functions based on specific security policies.

• Secure (integrity and confidentiality protection) transmission of information between reconfigurable components.

A deeper study of middleware security for reconfigurable terminals is required. However, it must be first clarified these open questions:

• Which entity can take the role of software provider? Will and how terminal manufacturers, regulatory bodies and network operators trust third party software providers?
• Which entity is responsible for the reconfiguration control on the network and terminal side? The answers to these questions have a great impact on the security architecture and policies required to support terminal reconfiguration.

We have identified and described some of the main research issues in the area of software defined reconfigurable terminals. The specific issues covered highlighted the convergence of the IT and telecommunications domains and the use of technologies previously specific to the IT domain to solve challenges arising from reconfigurability. Within the SCOUT project we are investigating the roles that middleware, flexible protocol stacks and service discovery frameworks can play in supporting reconfigurable terminals. The first step in these investigations is to identify the specific requirements for these technologies, some of which have been outlined.

Drive Testing

The growth and expansion of cellular and PCS networks continues at a rapid pace throughout the world. To retain existing customers and attract new customers, wireless service providers must maintain the highest quality of service throughout their networks. Drive-testing remains an essential part of the network life cycle, as an effective means for continually optimizing network performance to maintain customer satisfaction and reduce subscriber churn.

This application note provides an overview of how drive-test tools can help optimize your TDMA-based cellular and PCS networks. These tools allow you to turn-up networks faster, reduce optimization time, and improve network quality of service.

Drive-test solutions are used for collecting measurements over a TDMA air interface. The optimum solution combines network-independent RF measurements using a digital receiver with traditional phone-based measurements. A typical collection system includes a digital RF receiver, phone, PC, GPS receiver and antennas. Optimum drive-test solution consists of an integrated digital receiver and phone. A GPS receiver is required for location information.

Optimization Process

Optimization is an important step in the life cycle of a wireless network. Drive-testing is the first step in the process, with the goal of collecting measurement data as it relates to the user's location. Once the data has been collected over the desired RF coverage area, the data is output to a post-processing software tool or mapping software such as MapInfo.

Engineers can use these tools to identify the causes of potential RF coverage or interference problems and analyze how these problems can be solved. Once the problems, causes, and solutions are identified, steps are performed to solve the problem.

The goal is to improve quality of service, retain existing subscribers, and attract new ones while continually expanding the network. The optimization process begins with drive-testing, moves to post-processing, then requires data analysis, and finally action needs to be taken correct the problems. Drive-testing is performed again to verify that the actions were effective.

In recent years the number of wireless networks based on the IS-136 standard has grown considerably throughout both

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North America and Latin America. Many of these networks evolved from the Advanced Mobile Phone System (AMPS) standard. In anticipation of addressing capacity concerns, the IS-54 standard was written, enabling TDMA systems in the cellular frequency range (850 MHz). IS-54 based networks were implemented, offering capacity relief to crowded AMPS networks. As PCS frequencies became available, the IS-136 standard emerged, providing the same TDMA operation as IS-54 in both the cellular (850 MHz) and PCS (1900 MHz) bands. It also and provided the mechanism for additional services. Today, mobile handsets are readily available that can operate in TDMA and AMPS modes, and in both the cellular and PCS bands.

For channel assignments, the original AMPS wireless networks reliedon Frequency Domain Multiple Access (FDMA). To transmit and receive in FDMA systems, each user was assigned a dedicated frequency. In the case of AMPS, each channel is 30 kHz wide and uses frequency modulation (FM) to transport conversations. Since the amount of spectrum owned by wireless service providers is limited to a fixed number of 30 kHz channels, each channel must be reused many times throughout a network in order to provide enough channel capacity to satisfy customer demand. Two base stations assigned to use the same channel must be located far enough apart so that the channel users do not interfere with each other. To provide satisfactory voice quality in a wireless network, it is important to detect and fix situations in which base stations are interfering with each other.

Wireless networks based on the IS-136 standard operate using two methods of access FDMA and Time Domain Multiple Access (TDMA). The same 30 kHz channel bandwidth used in AMPS systems is used here. Also as with AMPS networks, each channel must be used by multiple base stations, so networks must be engineered to minimize interference between base stations. In addition, three users share each 30 kHz channel. This is accomplished by dividing the channel into time slots so each user of the channel has full use of the channel only onethird of the time. Thus the TDMA network capacity is tripled as compared to an AMPS network.

In AMPS systems, each conversation is assigned a 30 kHz channel. In TDMA systems three conversations are assigned to each 30 kHz channel. Each conversation uses the channel1/3 of the time. Interference between base stations can exist since each channel is used in more than one base station. If two base stations are using the same channel simultaneously it can cause co-channel interference or if they are using an adjacent channel simultaneously it can cause adjacent channel interference. To maximize voice quality, both types of interference must be identified and eliminated.

It is not difficult to identify adjacent channel and co-channel

interference when optimizing IS-136 TDMA wireless networks. This is typically done using a phone-based drive-test system, which can identify areas of high bit error. However, identifying the base station transmitting the interfering signal can be a challenge. Using a drive-test tool that has both phone- and receiver-based measurements allows the user to not only identify that interference exist, but also which base station is the source of the interference.

TDMA wireless service providers need to assign channels to each base station. The channel assignments must be made in a way that minimizes interference. A channel table is used to assign channels for each of the service provider’s geographical markets, based on the market needs.

Channel tables are commonly used to group channels into channel sets. Channel sets are based on a channel reuse number (the number of cells in which all channels will be used once, before they are reused in additional cells). The channel reuse number determines the number of columns in the channel table. Typically the number of columns is 3 times the channel reuse number. For example, for a network using 7-cell reuse (channel reuse number = 7) there are 21 columns in the channel table (7 cells x 3 sectors per cell = 21 channel groups). A network with 6-cell reuse would have 18 columns. A network using 12-cell reuse would have 36 columns. The smaller the channel reuse number, the larger the number of channels per channel set. This is true since each of the channels owned by an operator must occupy one spot in the channel table. If the number of columns in the table is decreased then the number of rows will increase. Conversely, if the number of columns increases then the number of rows decreases. Using a smaller channel reuse number requires a shorter distance between sectors that reuse the same channels, since the number of channel sets (columns in the table) is decreased. Tables 6 and 7 contain typical channel tables for both the cellular and PCS bands.

A1	B1	C1	D 1	E 1	F1	G1	A2	B2	C2	D2	E2	F2	G2	A3	B3	C3	D3	E3	F3	G3
333	332	331	330	329	328	327	326	325	324	323	322	321	320	319	318	317	316	315	314	313
312	311	310	309	308	307	306	305	304	303	302	301	300	299	298	297	296	295	294	293	292
291	290	289	288	287	286	285	284	283	282	281	280	279	278	277	276	275	274	273	272	271
270	269	268	267	266	265	264	263	262	261	260	259	258	257	256	255	254	253	252	251	250
249	248	247	246	245	244	243	242	241	240	239	238	237	236	235	234	233	232	231	230	229
228	227	226	225	224	223	222	221	220	219	218	217	216	215	214	213	212	211	210	209	208
207	206	205	204	203	202	201	200	199	198	197	196	195	194	193	192	191	190	189	188	187
186	185	184	183	182	181	180	179	178	177	176	175	174	173	172	171	170	169	168	167	166
165	164	163	162	161	160	159	158	157	156	155	154	153	152	151	150	149	148	147	146	145
144	143	142	141	140	139	138	137	136	135	134	133	132	131	130	129	128	127	126	125	124
123	122	121	120	119	118	117	116	115	114	113	112	111	110	109	108	107	106	105	104	103
102	101	100	99	98	97	96	95	94	93	92	91	90	89	88	87	86	85	84	83	82
81	80	79	78	77	76	75	74	73	72	71	70	69	68	67	66	65	64	63	62	61
60	59	58	57	56	55	54	53	52	51	50	49	48	47	46	45	44	43	42	41	40
39	38	37	36	35	34	33	32	31	30	29	28	27	26	25	24	23	22	21	20	19
18	17	16	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	-	-	-
716	715	714	713	712	711	710	709	708	707	706	705	704	703	702	701	700	699	698	697	696
695	694	693	692	691	690	689	688	687	686	685	684	683	682	681	680	679	678	677	676	675
674	673	672	671	670	669	668	667	-	-	-	-	-	-	-	-	-	-	-	-	-
1023	1022	1021	1020	1019	1018	1017	1016	1015	1014	1013	1012	1011	1010	1009	1008	1007	1006	1005	1004	1003
1002	1001	1000	999	998	97	996	995	994	993	992	991	-	-	-	-	-	-	-	-	-

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Each column of a channel table is a channel set. Channel sets can be named in a variety of ways. Two typical ways are to number the channel sets (1,2,3,…21) and to use the following convention (A1,B1,C1,…A2,B2,C2,…A3,B3,C3…). Refer to Tables 8 and 9 for channel set examples.

Table 8. Channel set A1 taken from A-band cellular table, 7-cell reuse

A1	B1	C1	D1	E1	F1	G1	H1	I1	J1	K1	L1	A2	B2	C2	D2	E2	F2	G2	H2	I2	J2	K2	L2	A3	B3	C3	D3	E3	F3	G3	H3	I3	J3	K3	L3
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	32	33	34	35	36
37	38	39	40	41	42	43	44	45	46	47	48	49	50	51	52	53	54	55	56	57	58	59	60	61	62	63	64	65	66	67	68	69	70	71	72
73	74	75	76	77	78	79	80	81	82	83	84	85	86	87	88	89	90	91	92	93	94	95	96	97	98	99	100	101	102	103	104	105	106	107	108
109	110	111	112	113	114	115	116	117	118	119	120	121	122	123	124	125	126	127	128	129	130	131	132	133	134	135	136	137	138	139	140	141	142	143	144
145	146	147	148	149	150	151	152	153	154	155	156	157	158	159	160	161	162	163	164	165	166	167	168	169	170	171	172	173	174	175	176	177	178	179	180
181	182	183	184	185	186	187	188	189	190	191	192	193	194	195	196	197	198	199	200	201	202	203	204	205	206	207	208	209	210	211	212	213	214	215	216
217	218	219	220	221	222	223	224	225	226	227	228	229	230	231	232	233	234	235	236	237	238	239	240	241	242	243	244	245	246	247	248	249	250	251	252
253	254	255	256	257	258	259	260	261	262	263	264	265	266	267	268	269	270	271	272	273	274	275	276	277	278	279	280	281	282	283	284	285	286	287	288
289	290	291	292	293	294	295	296	297	298	299	300	301	302	303	304	305	306	307	308	309	310	311	312	313	314	315	316	317	318	319	320	321	322	323	324
325	326	327	328	329	330	331	332	333	334	335	336	337	338	339	340	341	342	343	344	345	346	347	348	349	350	351	352	353	354	355	356	357	358	359	360
361	362	363	364	365	366	367	368	369	370	371	372	373	374	375	376	377	378	379	380	381	382	383	384	385	386	387	388	389	390	391	392	393	394	395	396
397	398	399	400	401	402	403	404	405	406	407	408	409	410	411	412	413	414	415	416	417	418	419	420	421	422	423	424	425	426	427	428	429	430	431	432
433	434	435	436	437	438	439	440	441	442	443	444	445	446	447	448	449	450	451	452	453	454	455	456	457	458	459	460	461	462	463	464	465	466	467	468
469	470	471	472	473	474	475	476	477	478	479	480	481	482	483	484	485	486	487	488	489	490	491	492	493	494	495	496	497	498	499	-	-	-	-	-

Each row in a channel table can be set aside for a particular use. For example, a row is dedicated to analog control channels in mixed AMPS/TDMA systems. In addition, rows may be set aside for AMPS channels, Digital Traffic Channels (DTCs) and Digital Control Channels (DCCHs). Please refer to Table 10 for an example of assigning dedicated uses for each row in a channel table.

Designated use	A1	B1	C1	D1	E1	G3
Analog control channel	333	332	331	330	329	313
Digital traffic channel	312	311	310	309	308	292
Digital traffic channel	291	290	289	288	287	271
Digital traffic channel	270	269	268	267	266	250
Digital traffic channel	249	248	247	246	245	229
Digital traffic channel	228	227	226	225	224	208
Digital traffic channel	207	206	205	204	203	187
Digital traffic channel	186	185	184	183	182	166
Digital traffic channel	165	164	163	162	161	145
Digital traffic channel	144	143	142	141	140	124
Digital traffic channel	123	122	121	120	119	103
Digital traffic channel	102	101	100	99	98	82
Digital traffic channel	81	80	79	78	77	61
Digital traffic channel	60	59	58	57	56	40
Digital traffic channel	39	38	37	36	35	19
Digital traffic channel	18	17	16	15	14	–
Digital traffic channel	716	715	714	713	712	696
Digital traffic channel	695	694	693	692	691	675
Digital control channel	674	673	672	671	670	–
AMPS	1023	1022	1021	1020	1019	1003
AMPS	1002	1001	1000	999	998	–

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For example, a row is dedicated to analog control channels in mixed AMPS/TDMA systems. In addition, rows may be set aside for AMPS channels, Digital Traffic Channels (DTCs) and Digital Control Channels (DCCHs). Please refer to Table 10 for an example of assigning dedicated uses for each row in a channel table.

Channel Planning Techniques

In general, one channel set is assigned to each base station sector when designing a channel plan for a TDMA/AMPS network. A channel set designation is given to each sector (such as A2 or E3). Just because a channel set is assigned to a sector, it doesn’t mean that all the channels in that channel set are used at that sector. The number of channels used depends on the sector’s voice traffic demand. Only a few channels are used in areas of low mobile phone density, while all the channels in the channel set are used in areas of high mobile phone density. Some sectors with very high usage may require assignment of additional channel sets in order to have enough channels to handle customer demand.

Channel plans can be designed so channel sets are assigned once in each reuse cluster (once every 7 cells in a system using a reuse number of 7). Each cell cluster is shaded the same way. Cells that use the same channel are labeled with the same number. However, cells aren't always built on a grid with an equal distance of separation. Terrain and traffic loading pose challenges to a standard reuse plan. Channel sets are not always assigned in reuse clusters, but are instead assigned where they will cause the least amount of interference.

Most TDMA infrastructure allows the channels in a base station to be utilized in order according to an assignment list. This means that the first channel in the assignment list is the first channel to be assigned a call. If another call is set up it will be assigned to the second channel in the assignment list and so on. Using this method, the first channel in the assignment list will have the highest amount of usage and the last channel in the assignment list will have the least amount of usage. The last channels in the assignment list will only be used during the periods of time when the sector is busiest.

When assigning multiple sectors of a network to use the same channel set, alternating channel assignment can be used to minimize the probability of co-channel interference. Alternating channel assignment takes two of the closest sectors and assigns them the same channel set, but their channels are assigned in alternate directions. One sector will use the lowest channel number in the channel set then gradually increase to higher channel numbers, until there are enough channels assigned to meet the traffic demand. The

other sector (assigned with the same channel set) uses the highest channel number in the channel set, then gradually decreases to lower channel numbers until there are enough channels assigned to meet traffic demand. If the two sectors don't require all the channels in the channel set be used, then some channels used at one sector will not be used at the other sector. There will be no co-channel interference between the sectors for these channels -- consisting of the highest and lowest channel numbers in the channel set. This method is not as beneficial if either of the sectors require all channels in the channel set.

The previous example of two of the closest sectors being assigned to the same channel set, helps illustrate the power of using alternating channel assignment and assignment lists in conjunction. One sector is given an assignment list, which starts with the lowest channel number then ends with the highest used channel number in the channel set. The other sector's assignment list is just the opposite, starting with the highest number in the set and ending with the lowest assigned channel.

Using this method minimizes the probability of co-channel interference between the two sectors. Table 11 illustrates the assignment list and alternating channel assignment techniques.

Chapter II

Interference Guidline

This section gives quantitative measures for describing interference. Two types of self-imposed interference are present in TDMA systems -- adjacent channel and co-channel interference. When channel planning, it is important not to use adjacent channel groups too close together because the two channels can interfere with each other. There is no guard band between 30 kHz TDMA channels, so the network designer must implement a guard band by properly assigning channels. How close together can sectors be which use the same channel set? When frequency planning, the carrier channel must always be stronger than any adjacent channel. If one of the two adjacent channels is ever stronger than the serving channel, voice quality will begin to degrade (bit errors will be induced). To quantify this condition the carrier to adjacent ratio can be used (C/A). The C/A is found by subtracting the dBm value of the adjacent channel from the dBm value of the serving channel. If the C/A is less than or equal to zero, voice quality will begin to degrade.

Interference on the same channel as the serving channel must also be considered. In this case the signal coming from any base station other than the serving base station (the one communicating with the phone) must be 17 dB lower than the serving signal. If it is higher, voice quality will begin to degrade. Planning adequate isolation between sectors that use the same channel set will ensure interfering signals are 17 dB lower than the serving signal. The carrier to interference ratio is used to quantify co-channel interference. The C/I is found by subtracting the dBm value of the interfering signal from the dBm value of the serving signal. If the C/I is less than 17 dB then voice quality will begin to degrade.

Digital Verification Color Code

The Digital Verification Color Code (DVCC) is a signal sent from the base station to the phone, then from the phone back to the base station. When the phone transmits the DVCC on the uplink it must send the DVCC that it received from the base station on the downlink. The phone cannot use prior knowledge of the DVCC. It is not allowed to send the correct DVCC on the uplink if it received the wrong DVCC on the downlink.

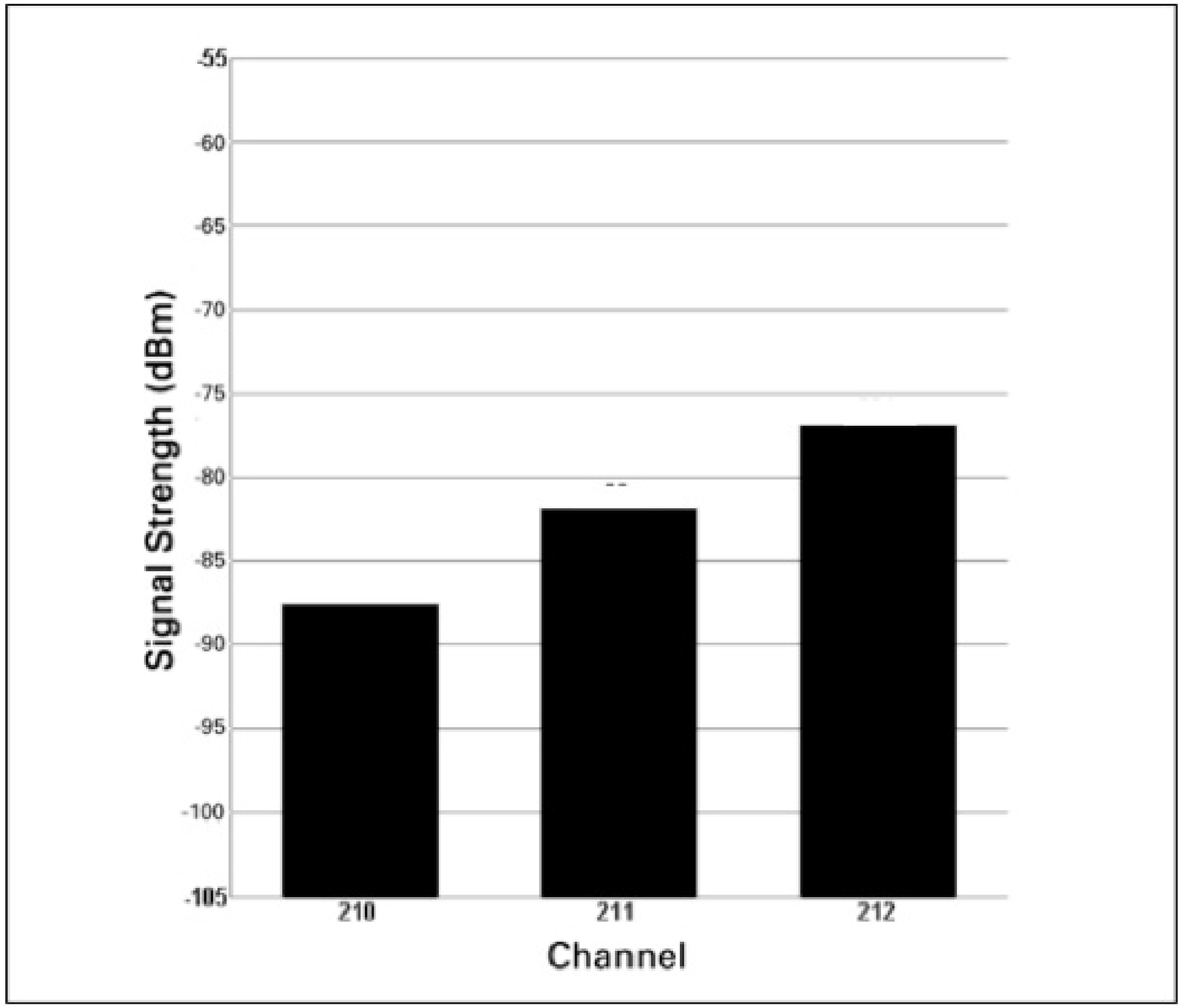
If the correct DVCC is not received by the base station on the uplink then the phone call may be handed off or dropped after a period of time specified by the network operator. The DVCC is heavily error coded before it is transmitted so that the DVCC can be correctly decoded despite some amount of bit error. In cases when the heavily protected DVCC code is not making it through correctly on both the downlink and uplink, then some impairment must be present which is causing bit

errors.

Typically the same DVCC is assigned to all sectors of a base station. The DVCC is transmitted on both Digital Control Channels (DCCH) and on Digital Traffic Channels (DTC). Since there are 255 unique DVCCs, there is usually great distance between base stations that use the same DVCC. Each of the 255 DVCCs are assigned to a base station before having to reuse the DVCC for an additional base station. Since the same DVCC is typically used for all sectors of a base station, the DVCC reuse will occur only in systems having more than 255 base stations. Once a DVCC must be used in more than one base station, the base stations can be isolated by a large distance, thus having little chance to interfere with each other.

Method For Adjacent Channel Interference Identification

When there is adjacent channel interference, bit errors occur and voice quality degrades. Many receiver-based drive-test tools are available to help identify that the adjacent channel C/A guideline (C/A >0) has been violated. In Figure 6 the horizontal axis is used for channel number and the vertical axis is used for signal strength (in dBm). An adjacent channel interference problem since channel 212 is stronger than the serving signal, channel 211. The channel number of the serving channel can be determined using a phone-based drive-test tool.



Bit errors will occur on channel 211 due to the strong signal present on channel 212. Knowing that the interfering signal is on channel 212 is useful. A channel plan can now be used to determine the source of the strong signal on channel 212. First, examine the channel table for channel set D3, of which channel 211 is a member. For this example assume that the serving sector is 5. Notice that the techniques of alternating

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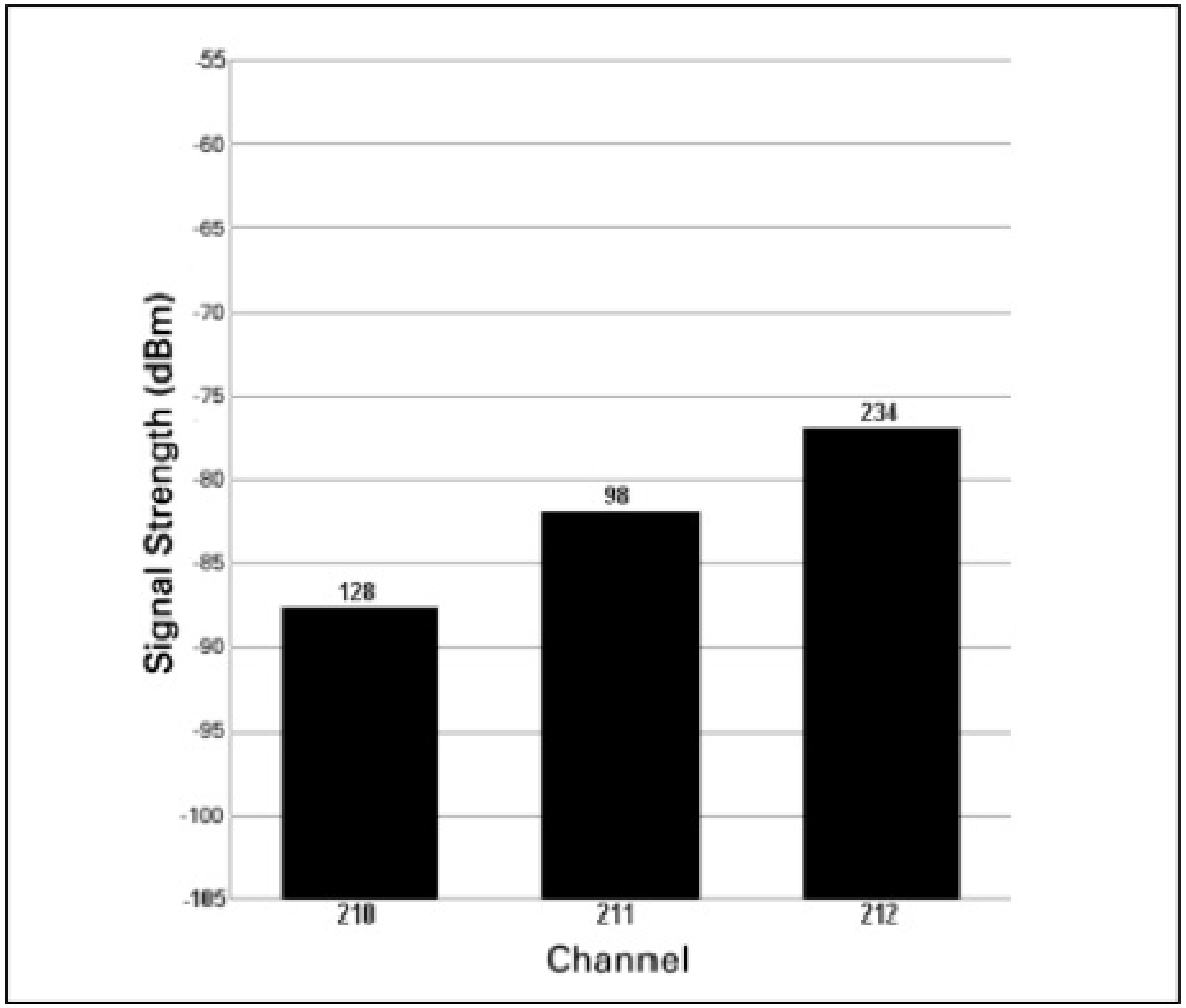
channel assignment and assignment list and channel assignment.

A strong signal on channel 212 is causing interference on channel 211. The channel plan for the E3 channel set can be used to determine which sectors are assigned to use channel 212.

Using knowledge of the base station locations, terrain, antenna heights and orientations, each potential interferer can be evaluated. The results will be as illustrated.

Potential interferers on channel 212	**Interference likely?**
17α	Yes
26β	Maybe
38β	No
49γ	No
109γ	Maybe
234γ	Yes



The process of identifying the interfering base station is simplified if a drive-test tool is able to provide the channel number, power level of the interfering signal and the DVCC of the interfering signal. In addition, the DVCC of the signal on channel 212 is shown on top of the bar. In this case the DVCC is 234.

Knowing the DVCC, a new table can now be constructed to

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help identify the interferer. All the sectors using channel 212 are listed, along with the DVCC assigned to each of those sectors. The DVCC of the interferer was identified as 234. Using the following table it is clear to see that 234 is the signal transmitting the interfering sector since it is the only sector assigned both channel 212 and DVCC 234. No guesswork or additional investigation is required since the interferer has been uniquely identified. Using this method reduces drive-test and investigation time. Now that the offending base station has been identified, steps can be taken to solve the interference problems. These steps may include channel plan changes, antenna height or orientation changes, and others.

Force idle channel method

Finally, force idle channel is the last method. Force idle channel applies to potentially interfering base stations with channels that completely overlap with the serving base station, as in the wait for idle channel method. The force idle channel method is the same as wait for idle channel method except that it does not require that the user wait for channels to go idle at the serving base station and hope that the same channel will be active at the interfering base station. Instead a channel is temporarily forced to be idle at the serving base station and the same channel is temporarily forced to be active at the potentially interfering base station. Force idle channel is quicker than the wait for idle method, but requires the user to send control commands to the serving base station and to the potentially interfering base stations in order to force the channels idle and active respectively.

Force idle channel method can be used if there is high channel usage at the serving base station, making the wait for idle method ineffective. All three methods discussed can be effective in determining the DVCC of the interfering base station. Using these methods saves time over methods that don't involve the use of decoded DVCC because they provide the user with the ability to definitely determine the base station transmitting the interfering signals. Once the interfering base station has been identified, steps can be taken to solve the interference problems. As with adjacent channel interference, these steps may include channel plan changes, antenna height or orientation changes, and others.

Adjacent and co-channel interference problems in IS-136 TDMA wireless networks are not difficult to identify using a phone-based drive-test tool. However, identifying the base station which is the source of the interference can be difficult and time consuming without the use of identification methods that rely on a receiver-based drive-test tool capable DVCC decode. The methods which have been presented for interferer identification can save time and lead to significant voice quality enhancement in IS-136 TDMA wireless networks.

Effects of Additive Noise on the Throughput of CDMA Data Communications

We analyze the optimum transmitter power levels and the optimum number of active terminals sending data to a CDMA base station. The objective is to maximize the aggregate throughput of the base station. We find that in the presence of additive noise, received power balancing is suboptimal mathematically. We consider N terminals transmitting at the same data rate, with the power of the most distant terminal (terminal N), fixed at its maximum value, and the power of the other N-1 terminals varying. We conclude that the aggregate throughput at the base station is maximized when the receiver powers for the N-1 terminals are equal and larger than the receiver power of the N-th terminal. This finding reduces the complexity of the analysis to a unvaried optimization problem.

A numerical analysis indicates the extent to which additive noise reduces the optimum number of active terminals and the maximum base station throughput. Keywords- power control; radio resources management; power balancing.

We analyze the throughput of a CDMA base station receiving data from N transmitters, all operating at the same constant bit rate. We consider two resource management issues: transmitter power control and the number of terminals that should be admitted to the system in order to maximize base station throughput.

Early work on uplink CDMA power control focused on telephone communications and determined that to maximize the number of voice communications, all signals should arrive at a base station with equal power. Initial studies of power control for data communications focused on maximizing the utility of each terminal, with utility measured as bits delivered per Joule of radiated energy.

By contrast, the results pertain to the case when noise and interference from other cells are not negligible. We show that with additive noise power balancing leads to sub-optimal performance and that when one terminal has a maximum power constraint, the optimum set of transmitter power levels depends on the maximum received SNR of the constrained terminal. Furthermore, we demonstrate that when one terminal has a maximum power constraint, the other terminals should aim for the same received power, which depends on the maximum SNR of the constrained terminal.

A data source generates packets of length L bits at each terminal of a CDMA system. A forward error correction encoder, if present and a cyclic redundancy check (CRC) encoder together expand the packet size to M bits. The data rate of the coded packets is Rs b/s. The digital modulator spreads the signal to produce Rc chips/s. The CDMA processing gain is G=W/Rs, where W Hz, the system bandwidth, is proportional to Rc.

Terminal i also contains a radio modulator and a transmitter radiating Pi watts. The path gain from transmitter i to the base station is hi and the signal from terminal i arrives at the base station at a received power level of Qi=Pihi watts. The base station has N receivers, each containing a demodulator, a correlator for despreading the received signal, and a cyclic redundancy check decoder. Each receiver also contains a channel decoder if the transmitter includes forward error correction.

The noise appears at the receiver as an additional signal that does not contribute to the overall throughput. The system has to use some of its ower and bandwidth resources to overcome the effects of the noise.

The effects of noise depend on the power limits of practical terminals. With unlimited power, we would increase all the

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received powers Qi indefinitely until the effect of the noise is negligible. To account for the power limits, let Pi,max denote the power of the trongest possible signal transmitted by terminal i and Qi,max=Pi,maxhi, the power of the corresponding received signal.

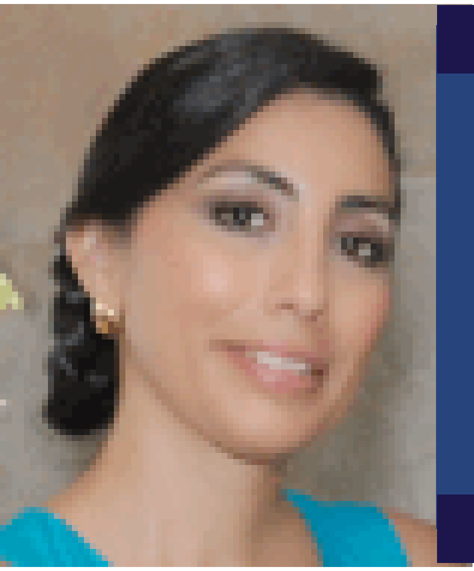
The research we made considers how to maximize the throughput of a CDMA base station receiving data from N transmitters, all operating at the same constant bit rate. The principal conclusions are that the aggregate throughput at the base station is maximized if N-1 transmitters aim for a target SINR that is greater than the maximum SINR of the weakest terminal. Further, the number of active terminals should not exceed N*, where N* is the maximum number of terminals that can simultaneously transmit whilst ensuring an interior solution of the first and second order optimality conditions.

The development of CDMA2000 benefited from the extensive experience acquired through several years of operation of cdmaOne systems. Explicit steps were taken to increase voice capacity and data throughput. As a result, CDMA2000 is a very efficient and robust technology. Supporting both voice and data, the standard was devised and tested in various spectrum bands, including the new IMT-2000 allocations.

In many countries, there is tremendous demand for new services and the operators are looking to provide new services to many more subscribers at reasonable prices. Being fully backward compatible with cdmaOne systems and operating in a variety of bandwidths, CDMA2000 can support a wide variety of operator needs. Connectivity to existing ANSI-41, GSM-MAP, and IP networks further demonstrates its versatility.

The unique features, benefits, and performance of CDMA2000 makes it an excellent technology of choice for high voice capacity and high-speed packet data. The fact that CDMA2000 1X has the ability to support both voice and data services on the same carrier, makes it cost effective for wireless operators.

Due to its optimized radio technology, CDMA2000 allows operators to invest in a fewer number of cell sites with faster deployment, ultimately enabling the service providers to increase their revenues with faster Return On Investment (ROI). Increased revenues, along with a wider array of services, make this the technology of choice for service providers.



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