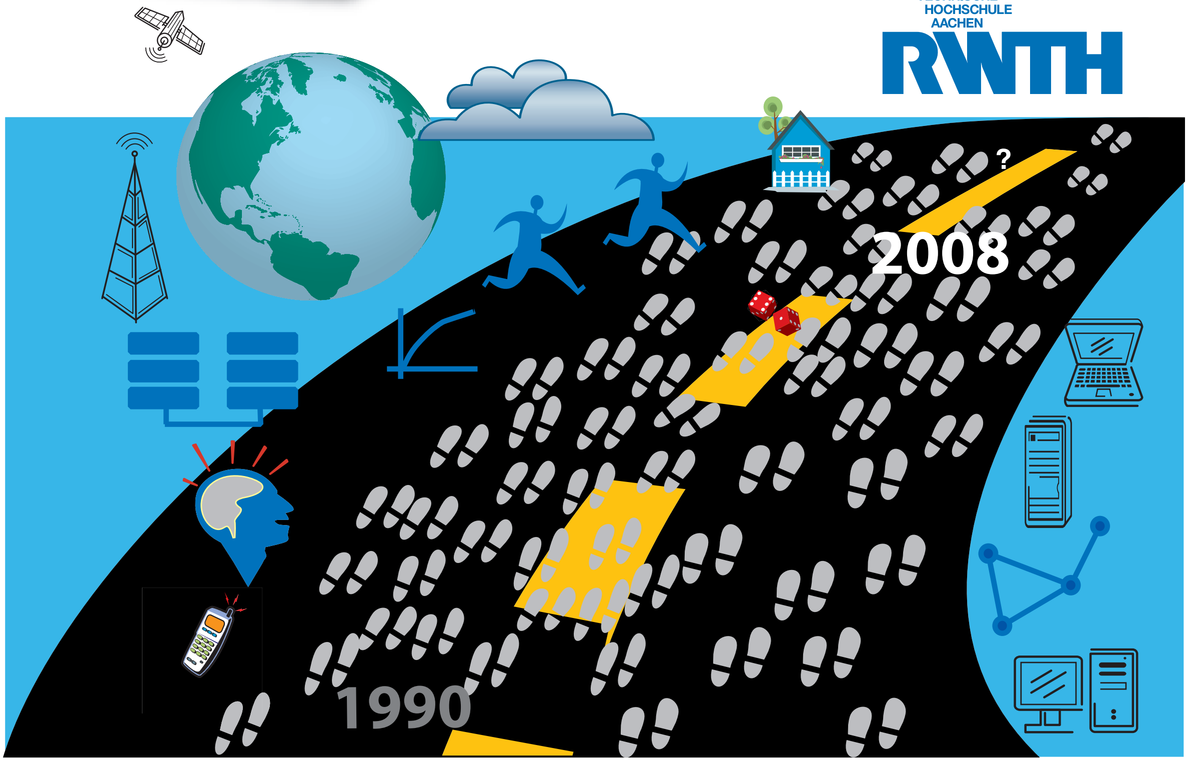
ComNets

Lehrstuhl für Kommunikationsnetze
Prof. Dr.-Ing. Bernhard Walke

RHEINISCH-WESTFÄLISCHE TECHNISCHE HOCHSCHULE AACHEN
RWTH



Sonderband zur Einweihung des
ComNets - Gebäudes

Aachen, 21. November 2008

ABMT SONDERBAND 1000

VORWORT

Liebe Freunde des Lehrstuhls Kommunikationsnetze der RWTH Aachen,

trotz seiner Emeritierung ist Professor Walke nach wie vor so aktiv wie eh und je und als Forscher „immer im Dienst“. Seine außerordentlichen Leistungen und Erfolge wurden bereits mehrfach gewürdigt. Dennoch wollten wir vom Freundeskreis des Lehrstuhls für Professor Walke als herausragenden Wissenschaftler und Mentor noch etwas Besonderes gestalten, welches die Historie und wissenschaftliche Arbeit des Lehrstuhls reflektiert und gleichzeitig einen persönlichen Charakter aufweist. In gemeinsamen Diskussionen wurde die Idee für einen Sonderband der „Aachener Beiträge zur Mobil- und Telekommunikation (ABMT)“ geboren.

Die anstehende Einweihung des neuen ComNets-Gebäudes erschien uns als *der* geeignete Zeitpunkt zur Übergabe und so wurden wir aktiv. Hinter „verschlossenen Türen“ fanden geheim und nicht wirklich verwunderlich doch irgendwie für Professor Walke bekannt deutschlandweit mehrere Treffen und Diskussionen am Rande von Tagungen und Projekttreffen statt. Reger Datenaustausch begann und unter dem Arbeitstitel „ABMT 1000“ – um genügend Platz für viele weitere Dissertationen zu lassen – gewann der Band langsam aber stetig an Form. Einen kleinen Einblick in die Vernetzung von ComNets mit Politik, Hochschule und Industrie geben die Grußworte von einigen ausgewählten Stellvertretern aus diesen Bereichen. Stellvertretend für alle ehemaligen und derzeitigen ComNets‘ler beinhaltet der Band im Anschluss Schnappschüsse zurückliegender Arbeiten und aktueller Forschung, die auf Arbeiten bei ComNets aufbauen.

So ist dieser Band auch ein Produkt des aufgebauten Freundeskreises, der der Beweis dafür ist, dass ComNets für die Vernetzung auf allen Ebenen und Schichten steht!

Viel Spaß beim Schmökern wünscht das Programmkomitee

Götz Brasche

als Vorsitzender des Freundeskreises

und in alphabetischer Reihenfolge:

Lars Berlemann, Stephan Böhmer, Reinhold Gebhardt, Carmelita Görg, Michael Gude, Stefan Mangold, Maciej Mühleisen, Norbert Niebert, Carl Herbert Rokitansky, Peter Stuckmann, Thomas und Heide Walke, Christian Wietfeld, Bangnang Xu

Sonderausgabe anlässlich der Eröffnung des ComNets-Gebäudes

RWTH Aachen

November 2008

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Ministerium für Innovation,
Wissenschaft, Forschung und Technologie
des Landes Nordrhein-Westfalen

GRUSSWORT DES MINISTERS FÜR INNOVATION, WISSENSCHAFT, FORSCHUNG UND TECHNOLOGIE DES LANDES NORDRHEIN-WESTFALEN

für die Sonderausgabe der Aachener Beiträge zur Mobil- und Telekommunikation aus Anlass der Verabschiedung von Professor Dr.-Ing. Bernhard H. Walke, Lehrstuhl für Kommunikationsnetze an der RWTH Aachen

Prof. Dr. Andreas Pinkwart

Es gibt nur wenige Wissenschaftler, die am Ende ihrer Dienstzeit auf eine solche Erfolgsbilanz zurückblicken können, wie Prof. Walke.

Prof. Walke war 25 Jahre im Land Nordrhein-Westfalen als Hochschullehrer tätig, davon 15 Jahre an der RWTH Aachen, an „unserer" Elite-Universität, wie wir neuerdings in Nordrhein-Westfalen mit Stolz sagen dürfen, denn sie hat in der Exzellenzinitiative von Bund und Ländern ihren Ruf als eine der besten deutschen Universitäten nachdrücklich untermauert.

Professor Walke hat in seiner Zeit in Aachen rund 1.000 Diplom-, Master- und Doktorarbeiten betreut, die inhaltlich nahezu alle Aspekte drahtgebundener und mobiler Kommunikationsnetze behandeln. Was seine Forschung betrifft, so belegen über 150 rezensierte Konferenzbeiträge, mehr als 30 Veröffentlichungen in Zeitschriften und 15 Fachbücher seine Schaffenskraft.

Solche Zahlen deuten schon an, welche Bedeutung Prof. Walkes Arbeit für den Bereich der Kommunikationsnetze hat. Um seine Leistung angemessen zu würdigen, muss man jedoch hinzufügen, dass seine Arbeit von der akademischen Welt wie von der Industrie hochgeschätzt wird. Ohne seine Arbeit wären die vielen Möglichkeiten und Vorteile der heutigen

Kommunikationsstandards nicht denkbar. Das Forschungsgebiet „Algorithmen und Protokolle für zukünftige Mobilfunknetze“ des Lehrstuhls ist praktisch konkurrenzlos in Europa.

Damit nicht genug: Prof. Walke darf auch deshalb als Vorbild gelten, weil er seine Forschung nie im luftleeren Raum, sondern immer als Beitrag zum Innovationsprozess betrieben hat. Er genießt hohes Vertrauen der industriellen Partner und leitete als wissenschaftlicher Koordinator ebenso große deutsche Forschungsprogramme wie europäische Konsortien. Sie alle aufzuzählen, ist hier nicht genügend Raum.

Den ganz überwiegenden Teil seiner Arbeit hat Professor Walke dabei kontinuierlich aus Drittmitteln finanzieren können, dazu gehörte auch die Finanzierung von jährlich um die 35 wissenschaftlichen Angestellten. Vor allem: Welcher Lehrstuhl erwirtschaftet schon so viel zusätzliche Mittel, um den Neubau des Instituts auf dem Campus aus eigenen Mitteln finanzieren zu können?

Wer immer über Prof. Walke spricht, kann es nur mit Hochachtung tun. Er zeigt uns seit 25 Jahren, wie Wissenschaft an den Hochschulen als Schrittmacher für Innovation agieren kann. Persönlichkeiten wie Prof. Walke brauchen wir, um auf unserem Weg zum Innovationsland Nummer 1 voranzukommen.

Für sein Engagement für die Wissenschaft in Nordrhein-Westfalen sage ich Prof. Walke persönlich und im Namen der Landesregierung ganz herzlichen Dank.

Professor Dr. Andreas Pinkwart
Minister für Innovation,
Wissenschaft, Forschung und
Technologie des Landes
Nordrhein-Westfalen

GRUSSWORT DES REKTORS ZUR SONDERAUSGABE ABMT BAND 1000

Univ.-Prof. Dr. Burkhard Rauhut

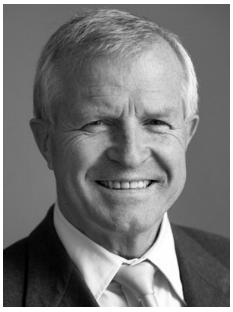
Sehr geehrte Leserinnen und Leser,

das Innovationstempo im Bereich Mobil- und Telekommunikation ist rasant. Durch neue technische Möglichkeiten und wissenschaftliche Visionen sind wir gewohnt, den Blick gespannt in die Zukunft zu richten. Diese Sonderausgabe der Aachener Beiträge zur Mobil- und Telekommunikation gewährt eine kleine Atempause und schaut einmal zurück. Der Band liefert eine Übersicht über die Themen und Inhalte der am Lehrstuhl für Kommunikationsnetze angefertigten Dissertationen und ordnet sie aus heutiger Sicht in die Entwicklung der letzten Jahre ein.

Gleichzeitig ist diese Sonderausgabe eine ideale Gelegenheit, Prof. Dr.-Ing. Bernhard H. Walke für seine ambitionierte und erfolgreiche Arbeit als Inhaber des Lehrstuhls für Kommunikationsnetze (ComNets) zu danken. Unter seiner Leitung wurde ComNets eines der größten universitären Forschungseinrichtungen Deutschlands auf dem Gebiet der mobilen Kommunikation. Die Ergebnisse der hier geleisteten Arbeit sind in verschiedene Standards eingeflossen. Im Rahmen der von Prof. Walke auch international wahrgenommenen Aktivitäten ist sicherlich seine Rolle als einer der Initiatoren des Exzellenzclusters „Ultra High-Speed Mobile Information and Communication“ (UMIC) hervorzuheben, in dem er eines von vier Forschungsgebieten koordiniert.

Prof. Walke ist es gelungen, gleichzeitig Visionär und praxisorientierter Wissenschaftler zu sein, der technologische Zukunftsthemen entsprechend kommunizieren und anhand von Modellen und Simulationen fassbar machen kann. Das schlägt sich in einer engen Zusammenarbeit mit der Industrie und einem entsprechenden Drittmittelbudget nieder. Wie sehr sich viele Absolventen persönlich mit dem Lehrstuhl verbunden fühlen, dokumentiert die Existenz des ComNets-Freundeskreises. Es ist eine besondere Freude, dass das diesjährige Treffen mit der Einweihung des neuen Institutsgebäudes zusammenfällt.

Ich wünsche allen Teilnehmern an dieser Zusammenkunft einen informativen und kurzweiligen Aufenthalt!



Univ.-Prof. Dr. Burkhard Rauhut
Rektor der RWTH Aachen
(September 1999 bis Juli 2008)

GRUSSWORT

Prof. Dr.-Ing. Ralf Lehnert

Aus Sicht eines ehemaligen Doktoranden des Vorgängers von Professor Bernhard Walke gratuliere ich ganz herzlich zu seinem überaus erfolgreichen Wirken, das zu einem der größten Lehrstühle der Informationstechnik mit dem Schwerpunkt Mobilkommunikation in Deutschland geführt hat.

Als Ehemaliger habe ich die Aktivitäten dieses Lehrstuhls immer mit besonderem Interesse verfolgt. In der Zeit seit 1990, dem Jahr der Berufung auf den Lehrstuhl Kommunikationsnetze, hat es viele technologische Fortschritte in der Kommunikationstechnik, speziell im Mobilfunk gegeben. In vielen Fällen ist der Lehrstuhl ComNets an diesen Entwicklungen im Vorfeld entscheidend tätig gewesen. Hier sei nur erwähnt die Einführung der GSM-Technik, die heute als Weltstandard in der Teilnehmerzahl das Festnetz überflügelt hat.

Das zweibändige Lehrbuch Walke, „Mobilfunknetze und ihre Protokolle“, erschienen 1998, hat vielen vielen Studenten und Ingenieuren diese damals neue Technik nahe gebracht. Dieses Buch wurde mehrfach aktualisiert und auch in Englisch veröffentlicht. Auch die Weiterentwicklung der Mobilfunksysteme zu 2.5G (GPRS, EDGE), 3G (UMTS) und 3.5G (HSPA) wurden jeweils durch wissenschaftliche Arbeiten am Lehrstuhl konzipiert, vorangetrieben und im Vorfeld unterstützt. Hier ist mir besonders ein Video in Erinnerung, welches den Prototyp einer Mobilstation mit ATM over Wireless beim Filmen zeigt – damals zur Expo in Lissabon 1998 – ein Rucksack-großes Gerät von beträchtlichem Gewicht.

In den nun fast 18 Jahren unter der Führung von Herrn Walke hat der Lehrstuhl eine sehr große wissenschaftliche Produktivität bewiesen; dies wird durch eine lange Liste an Veröffentlichungen und über 50 Dissertationen seit 1995 eindrucksvoll unter Beweis gestellt. Die Arbeiten am Lehrstuhl waren und sind immer so hochaktuell, dass Herr Walke zu Recht als Instanz in allen Fragen der Mobilfunktechnik oft zu Rate gezogen wird. Er hat durch sein mit seiner schlagkräftigen Mannschaft erworbenes und ständig gepflegtes Überblickwissen sehr viele Forschungsarbeiten auf nationaler und internationaler Ebene stimuliert und beeinflusst.

Ich wünsche dem Lehrstuhl Kommunikationsnetze und ganz besonders Herrn Kollegen Walke in den neuen Räumlichkeiten weiterhin viel Erfolg!

Ralf Lehnert

Prof. Dr.-Ing. Ralf Lehnert
Lehrstuhl Telekommunikation
Technische Universität Dresden

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Prof. Dr.-Ing. Dr. h. c. mult. Paul J. Kühn

Lieber Herr Walke,

mit großer Freude richte ich diese Grußworte an Sie, gespeist aus gemeinsamen Erfahrungen über mehr als vier Jahrzehnte hinweg, ausgehend von einem Studium der Elektrotechnik an der Universität Stuttgart. Das Studium und Ihre Forschungstätigkeit in der exzellenten Umgebung des früheren Forschungsinstituts der Fa. AEG Telefunken in Ulm (dem Vorläufer der Forschungsinstitute von EADS und Daimler AG), aus dem ganze Generationen von Wissenschaftlern und Professoren der Informations- und Kommunikationstechnik in Deutschland hervorgingen, hat Sie zeitlebens geprägt, dort wurden die Fundamente gelegt zur Architektur und Systemanalyse von Rechnersystemen und drahtlosen Netzen. Mit Ihren Berufungen an die Fernuniversität Hagen und später die RWTH Aachen haben Sie diese Kenntnisse in die Hochschullehre eingebracht und innovative Forschungsprojekte begründet, mit denen eine große Anzahl von Talenten geformt wurde, die ihrerseits heute in Forschung, Lehre und in verantwortlichen Positionen der Industrie und bei Netzbetreiber-gesellschaften stehen.

Ihr Lehrstuhl hat in den letzten zwanzig Jahren die Entwicklung der Mobilfunknetze und der drahtlosen Kommunikation geprägt wie kein anderer. Sie haben es verstanden, die Entwicklungen theoretisch zu fundieren und über Projekte mit der Industrie auf nationaler und internationaler Ebene mitzugestalten und über die kontinuierliche Mitarbeit in der Standardisierung zu verankern. Damit konnten Sie eindrucksvoll belegen, daß auch in Deutschland Technologietransfer möglich ist und zu wirtschaftlichen Erfolgen führt. Ein umfangreiches Schrifttum zeugt von äußerst fruchtbaren Forschungsergebnissen, Ihre Bücher zur Mobil- und drahtlosen Kommunikation sind weltweit als Standardwerke anerkannt. Das erfolgreiche Abschneiden der RWTH Aachen bei der Exzellenzinitiative geht nicht unwesentlich auf Ihre Visionen und Mitgestaltung zurück.

Mit dem Bezug des neuen Institutsgebäudes für ComNets wird ein weithin sichtbares Zeichen gesetzt für eine erfolgreiche Hochschulforschung; die neue Umgebung und der Institutsgeist wird auch zukünftig fähige Talente anziehen, die ihre Arbeit fortsetzen. Die informationstechnische Community dankt Ihnen für Ihr Wirken, mit dem Sie unser Ansehen weltweit erhöht haben. Ihnen wünsche ich auch zukünftig eine fruchtbare Zeit; wir werden alle Kräfte aufwenden müssen, um uns weiterhin im globalisierten Umfeld behaupten zu können.

Ihr

Paul J. Kühn

Prof. Dr.-Ing. Dr. h. c. mult. Paul J. Kühn
Institut für Kommunikationsnetze und
Rechnersysteme (IKR)
Universität Stuttgart

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Dr. Fiona Williams

Der von Prof. Walke seit 1990 geführte *Lehrstuhl für Kommunikationsnetze* hat seine Arbeit zeitgleich mit der Gründung des Ericsson F&E-Standortes in Herzogenrath/Aachen und eine ähnliche Erfolgsgeschichte wie die von dem Lehrstuhl maßgeblich begleitete GSM/Technologie vorzuweisen. Prof. Walke machte den Lehrstuhl zu einem der weltweit führenden Forschungsplätze der Mobilkommunikation. Gleichzeitig haben sich von Beginn an viele Berührungspunkte zwischen Lehrstuhl und Industrie ergeben, und wir blicken auf eine für beide Seiten erfolgreiche Zeit der Kooperation zurück. Die so oft geforderte Nähe von Hochschule und Industrie ohne eine unabhängige Stellung der Hochschule aufzugeben ist durch die Persönlichkeit und das Engagement von Prof. Walke beispielhaft und zu beiderseitigem Vorteil realisiert worden.

Einige Höhepunkte dieser Kooperation waren sicher die Arbeiten Anfang der 90er Jahre in Richtung einer weitergefassten Mobilität in Netzen und einer Kommunikationssteuerung, die auf individuelle Bedürfnisse zugeschnitten ist (z.B. im RACE Mobilise Projekt). Schon Anfang der 90er Jahre wurde die Bedeutung von paketvermittelten Diensten auch im Mobilfunk von Prof. Walke erkannt und Arbeiten in Richtung GPRS gestartet. Die Leistungsoptimierung dieser Dienste im Mobilfunk ist bis zum heutigen Tag ein Gebiet der Zusammenarbeit gewesen, mittlerweile mit einem Fokus auf den Einsatz in Fahrzeugen. Ein weiteres wichtiges Arbeitsgebiet ist die optimale Nutzung des für Kommunikationsdienste geeigneten Frequenzspektrums. Hier wurden in dem BMBF-geförderten Projekt COMCAR und später in den EU-Projekten Drive, OverDrive und WINNER für den Fortschritt des Mobilfunks entscheidende Ergebnisse erarbeitet.

Gerade hierzu möchte ich die durch das persönliche Engagement von Prof. Walke gekennzeichneten Aktivitäten im Zusammenhang mit dem Entstehen des Wireless World Research Forums (WWRF) hervorheben. Die innovativen Ideen und die Bereitschaft, komplexe und langfristige Themen engagiert zu verfechten, sind mir besonders eindrucksvoll in Erinnerung.

Prof. Walke hat auch in anderer Hinsicht zur Vertiefung der Beziehungen von Hochschule und Industrie beigetragen. So war er maßgeblich bei der Errichtung des von Ericsson geförderten Stiftungslehrstuhls „Mobilkommunikation" beteiligt. Ich möchte mich auch persönlich für die langjährige, vertrauensvolle und fruchtbare Zusammenarbeit bedanken.

Fiona Williams

Dr. Fiona Williams
Head of Research & IPR
Ericsson GmbH

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Dr. Werner Mohr

Sehr geehrter Herr Prof. Walke,

Ihr *Lehrstuhl für Kommunikationsnetze* nimmt eine besondere Stellung in der deutschen Forschungslandschaft im Bereich der Kommunikationstechnik ein, da Sie Protokollarchitekturen und die Protokollentwicklung für Mobilfunk- und drahtlose Kommunikationssysteme zu einem Forschungsschwerpunkt gewählt haben. In diesem Bereich haben Sie auch mit Ihren Mitarbeitern Themen zu Relay-basierten Systemen mit den entsprechenden Protokollen (Multihop-Systeme) und zur Funknetzplanung aus Systemsicht erfolgreich bearbeitet.

Seit mehr als 10 Jahren arbeiten Ihr Lehrstuhl und unser Haus in gemeinsamen Forschungsprojekten zusammen. Beispielhaft sind hier BMBF-Projekte wie ATMmobil, Coverage, Scalenet und Wigwam zu nennen. ATMmobil hat wesentlich zur Entwicklung des ETSI Standards HIPERLAN/2 beigetragen sowie auch den IEEE802.11 Standard für WLAN-Systeme durch Beiträge Ihres Lehrstuhls beeinflusst. Wir haben auch in EU-Projekten insbesondere im 6. Rahmenprogramm der EU kooperiert wie in den WWI-Projekten (Wireless World Initiative) WINNER und Ambient Networks zu Systems beyond 3G oder IMT-Advanced.

Von 2001 bis 2003 waren Sie im WWRF (Wireless World Research Forum) Chairman der Working Group 4 *New Radio Interfaces, Relay-based Systems & Smart Antennas* und haben durch Ihre erfolgreiche Arbeit zur internationalen Konsensbildung für zukünftige Systemkonzepte beigetragen.

Sie und Ihre Mitarbeiter haben Ihre Forschungsergebnisse durch viele internationale Konferenz- und Zeitschriftenbeiträge sowie in Büchern insbesondere aus dem Bereich mobiler Funknetze veröffentlicht.

Absolventen und Mitarbeiter Ihres Lehrstuhls sind in unserem Haus und auch in vielen anderen Unternehmen und Organisationen erfolgreich tätig.

Ich bedanke mich für Ihre Beiträge zur Kommunikationstechnik, die die Entwicklung neuer Systeme wesentlich beeinflusst haben.

Ihr

Werner Mohr

Dr. Werner Mohr
Head of Research Alliances
Nokia Siemens Networks GmbH & Co. KG
München

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Dr. Michael Gude

Sehr geehrter Herr Prof. Walke,

zu dem durch diesen Band an Aufsätzen dokumentierten Erfolgen in der Forschung im Bereich Wireless Communication möchte ich herzlich gratulieren. Ihr Institut hat als ComNets über viele Jahre national und international große Bedeutung erlangt. Dabei sind die Beiträge in der Forschung, die Ausbildungsleistungen in der Lehre, die vielen Diplom- und Promotionsarbeiten und die Anregung und Begleitung von Industrieprojekten gleichsam von hoher Bedeutung. Als Firmengründer, Ingenieur und Unternehmer aus Leidenschaft bin ich auch über die Anregungen zur Gründung von Spin-Offs dankbar. Gerade in einer Zeit, in der bereits Diplomanden mit hohen Gehältern von Unternehmen abgeworben werden, ist die Gründung von erfolgreichen Unternehmen nicht leicht. Im Gegensatz zu den meisten Lehrstühlen, die die Anmeldung von gewerblichen Schutzrechten, also vorwiegend Patenten, nicht betreiben, können Sie auch hier auf eine erfolgreiche Arbeit als Patentanmelder zurückblicken.

Ich möchte in diesem Zusammenhang auch betonen, dass dem Standort Deutschland nach wie vor die Umsetzung von Inventionen in Innovationen sehr schwer fällt. Diverse Nobelpreisträger, auch in den letzten Jahren, zeigen große Erfolge in der Grundlagenforschung. Die Umsetzung dieser Erkenntnisse und Inventionen in marktreife Produkte und Dienstleistungen, also in Innovationen, gelingt in Deutschland nur sehr vereinzelt. Dieses liegt nach meinen Beobachtungen überwiegend an den Barrieren zwischen Hochschullehrern, Unternehmern und Unternehmen. Trotz erheblicher Fördermittel überlegt nur ein verschwindender Teil von Naturwissenschaftlern und Ingenieuren die berufliche Tätigkeit mit der Gründung eines eigenen Unternehmens zu verbinden. Hier fehlen nicht zuletzt die Vorbilder in der Gesellschaft und der Hochschule. Kaum ein Hochschullehrer hat - wie Sie - die Gründung neuer Unternehmen oder die Förderung von Spin-Offs aktiv unterstützt.

Durch Ihre Arbeit in Wissenschaft und Lehre haben Sie dafür gesorgt, dass die durch Ihre Weitsicht begonnenen Arbeiten weltweit fortgesetzt werden.

Ihnen persönlich möchte ich von Herzen viel Befriedigung in Ihrer Arbeit im „Unruhestand“ und im neuen ComNets-Gebäude wünschen.

Herzlichst

Ihr Michael Gude

Dr. Michael Gude
Cologne Chip AG
Gude A&D Systeme GmbH
Gude Stiftung

GRUSSWORT

H.-Prof. Dr. Carl-Herbert Rokitansky

Lieber Professor Walke,

Vor fast genau 20 Jahren wurde in Deutschland das nationale Forschungsprojekt PROMETHEUS ins Leben gerufen, mit den Teilprojekten PRO-GEN, PRO-NET und PRO-COM für die Fahrzeugkommunikation.
Die Welt war damals eine andere als heute, der "Eiserne Vorhang" noch nicht gefallen, Deutschland noch nicht geeint und in der Kommunikation gab es zwar schon klobige, unhandliche Autotelefone (C-Netz) und das Fax (korrekt Telekopierer) hatte sich für den Informationsaustausch durchgesetzt.
Das INTERNET war zwar schon erfunden, war aber (neben einer breiteren Anwendung in ARPANET und MILNET in USA) in der übrigen Welt meist nur einer elitären Gruppe von Forschungseinrichtungen und Universitäten vorbehalten, für Dienste wie Remote Login (TELNET), File Transfer (FTP) und auch schon Email (SMTP).
Es fehlte aber die breite Anwendung und das World-Wide-Web (WWW) war noch nicht existent. Informationen holte man sich aus Büchern und Fachzeitschriften in Bibliotheken aber noch nicht aus dem "INTERNET".
Die Lebensgewohnheiten der Menschen waren noch ganz andere als heute, es gab noch kein Mobiltelefon und wer "unterwegs" telefonieren wollte, musste eine Telefonzelle aufsuchen. Rechner standen in Büros, zum Teil vereinzelt zu Hause, aber noch nicht auf den "Schößen" (Lap(Tops)) der Menschen.
Lange vor dieser Zeit hatten aber Sie (und einige andere) bereits eine Vision – und nicht nur eine Vision – sondern natürlich auch eine genaue Vorstellung von der technischen Umsetzung und Realisierung und hatten

diese bereits am 17.10.1983 (fast auf den Tag genau vor 25 Jahren) unter dem Titel "Funknetz mit einer Vielzahl von mobilen Stationen" beim Deutschen Patentamt angemeldet, wofür 3 ½ Jahre später am 26.02.1987 die Offenlegung des angemeldeten Patentes erfolgte.
Damals wurde bereits der Grundstein für eine überaus erfolgreiche Tätigkeit als Wissenschaftler, Forscher und Hochschullehrer gelegt, der über die Stationen AEG Telefunken in Ulm (Ihre zweite Heimat in Deutschland), einige, aber wichtige Jahre an der Fernuniversität Hagen, schließlich 1990 zu Ihrem Ruf an die RWTH Aachen, Lehrstuhl für Kommunikationsnetze (europaweit/weltweit besser unter "ComNets" bekannt) führte. Was in der Esse der FernUni geschmiedet wurde, wurde an der RWTH Aachen vollendet und hat als Pionierleistung in vielen Teilbereichen der Mobilkommunikation letztendlich dazu beigetragen und trägt noch immer bei, die oben erwähnten Lebensgewohnheiten der Menschen nachhaltig zu verändern und die mobile Kommunikation als absolut selbstverständliche (fast lebensnotwendige) Tätigkeit zu betrachten.
Dies wurde in zahlreichen Forschungsprojekten der Europäischen Kommission im 4., 5. und 6. Rahmenprogramm und in DRIVE, in nationalen Forschungsprojekten, sowie in einer Vielzahl von Industrie-Kooperationen – gemeinsam mit Ihren Mitarbeitern (gleichzeitig meist etwa 35 Assistenten/Innen) – erarbeitet. Insgesamt in etwa 150 einzelnen – oft wegen der sehr guten Arbeit aber kontinuierlich über die Jahre fortgesetzten – Projekten, u.a. in alphabetischer Reihenfolge: @dwise, 4G-Spektrum, A1, Accord, Adapt. Term, alc-atm, AMBIENT NETWORKS, AMBIENT NETWORKS II, Anette, ant-bosch, ANWIRE, atm-mobil, atm-phil-vorl, bos-atm, Brain, Brain-Mind, cameleon-eu, CDMA-Schutz, cen-pt02, cen-pt03-cw, cen-pt06-gb, cen-pt07-cw, cen-pt07-roki, cen-pto4, CoCar, CoCoNet, CoCoNet II, CoCoNet III, ComCar, commerce2-eri, commerce-eri, COOLWAVE, Coverage-Multihop, DCA-mmo, dect-nokia, DELTA, dfg-11-1-at, dfg-13-1-cg, dfg-7-1-gb, dfg-7-2-gb, dfg-7-3, dfg-8-1-sha, DISTEL, dkm-dfg, Dmotion, DRIVE, eed-atm, electra-eu, eval-eri, fel-correia-eu, fel-rodigr-eu, Fireworks, Ford, Future, GPRS, GPRS-GSM_MMO, GPRS-Qos, GPRS-Video, Huawei, indonesien, insured-eu, IPonAIR-eric, IPonAIR-eric. II, IPonAIR-T-Nova, IRAT-P3, Irma, mbs-eu, Media Point, meganet-sha, metwerk, MIND, miniWatt, MobCon-siem.MobCon II-siem., MobCon III-siem., MobCon IV-siem., , MobCon V-siemems, mobil-eu, Mobil-Funk-siem., Mobilkom, motiv-BMW, motiv-bosch, Multifunk, Multihop-comnets, Multihop-phil, multimedia-mobil, MYCAREVENT, Net-cologne, NEXWAY, OverDrive, phil-atm, PoSSuM, Powerline, rekening, SAILOR, saint-eu, samba-eu, Samo, SatNEx, ScaleNet-siem., ScaleNet-Telekom, schn-netze, SGOOSE, Smartshoppping, Sorbas, SorgN-phil., speet, STAR, STRIKE, SVC-phil, SVC II-phil.,

SVCIII-phil, TakeOFDM, TakeOFDM II, tdma-siem., TETRA II, tetrapol.nrw, thes-eu, top-nets-dfg, transit, UMIC, UMTS Frequ., umts-freq-siem., UMTS-L23-phil., umts-signal-phil., URMEL-P3, UTRA/DCA, Utran, vasco-eu, Verifikation, virtuous, VMTL, VMTL II, WIGWAM-phil., WIGWAM-siem., WINNER, WINNER II, wireless-shop-eri, WSI, WWRI.

So haben Sie als Lehrstuhlinhaber – sicher ohne Übertreibung – eine Stellung als "Papst" der Mobilkommunikation erreicht (vielleicht mit einigen Gegenpäpsten). Sie waren Fels in der Brandung, Leuchtturm und Wegweiser für eine ganze Generation von jungen Forschern und Doktoranden zu neuen Konzepten in der Mobilkommunikation.

Im Namen aller Mitarbeiter möchte ich Ihnen für diese Ihre wegweisende, unermüdliche Tätigkeit auf vielen interessanten Gebieten, auf denen zum Teil bahnbrechende Entwicklungen erarbeitet wurden – wie zahlreiche Publikationen und europäische sowie internationale Standard-Spezifikationen (in die diese Arbeiten Eingang gefunden haben und nachhaltig wirken) eindrucksvoll beweisen – herzlich danken!

H.-Prof. Dr. Carl-Herbert
Rokitansky
FB Computerwissenschaften
Universität Salzburg

COGNITIVE RADIO IN IEEE WIRELESS STANDARDIZATION

An Overview

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Abstract Regulation of spectrum will undergo revolutionary changes in the future allowing a less restricted and more flexible access to radio spectrum. Intelligent mobile radios, or so-called cognitive radios, will realize the dynamic usage of frequency bands on an opportunistic basis, by identifying and using under-utilized spectrum. Such radio systems will share spectrum either horizontally with distributed spectrum access such as listen-before-talk and equal rights to access the radio spectrum, or vertically, where so-called primary radio systems have higher priority to access the radio spectrum than the secondary radio systems. Cognitive radio techniques will considerably change the wireless communication market as barriers for market access are essentially lowered. Operator assistance for cognitive radio systems enables operators to provide a new type of service for spectrum owners (such as regulators or governments), or alternatively to the individual licensees: Spectrum usage is assisted in a reliable way with the help of a radio network that has clearly defined site locations and a trusted operator. Joint efforts of academia, regulators, infrastructure suppliers and telecom operators are required to establish trust in the cognitive radio technologies and to enable a sustainable go-to-market. In this context the cognitive radio standardization is crucial. This article introduces the ongoing standardization activities of the IEEE as one of the leading research organizations in the cognitive radio area

The Department of Communication Networks (ComNets) headed by Prof. Dr.-Ing. B. Walke was for many years a world leading institution in the field of cognitive radio research. Years of scientific work in the field of spectrum sharing, coexistence, interference mitigation and generic protocol stacks laid the foundation for this success.

1. COGNITIVE RADIO OVERVIEW

A Cognitive Radio (CR) is a self-aware communication system that efficiently uses spectrum in an intelligent way [1]. It autonomously coordinates the usage of spectrum in identifying unused radio spectrum on the basis of observing its radio environment and regulatory rules known to or learned by the CR. The classification of spectrum as being unused and the way it is used involves regulation, as this spectrum might be originally assigned to a licensed communication system. This secondary usage of spectrum is referred to as vertical spectrum sharing. The sharing between equals as for instance in unlicensed bands is referred to as horizontal sharing.

"Operator-assisted CR" adds a centralized mean for coordinating spectrum access to the distributed paradigm of CR. Such a centralized approach has the advantage of facilitating the regulator's control of spectrum usage, and to allow them directing how the spectrum is used. Such a control is in particular interest in vertical spectrum sharing. In contrast, the decentralized approach is more flexible, which is especially helpful in horizontal spectrum sharing. To protect licensed radio systems in vertical spectrum sharing scenarios, but to gain from the flexibility of decentralized approaches, supplementary radio systems owned by operators may assist CRs in identifying under-utilized spectrum.

The introduction of CR will be rather a smooth evolution than a radical revolution of communication technology. Standardization, spectrum regulation, confidence in business opportunities and a go-to-market of commercial products based on CR will most likely be incremental on a step-by-step basis. The "digital dividend", i.e. the freed TV broadcast spectrum resulting from transition from analog to digital TV, is an initial play ground and proof of concept for CR. Many initiatives are targeting to develop communication technologies for the reuse of TV bands and vertical sharing with incumbent TV broadcast signals.

With this article we aim to provide a detailed overview on what we believe are the most promising approaches in CR research and standardization. The rest of the article is structured as follows: A general introduction of major standardization bodies in the field of CR is given in Section 2. More detailed standardization of incremental building blocks on the way to the CR vision is discussed in Section 3. Each subsection introduces the scope and purpose of the respective IEEE Working Group (WG). Thereafter the WG's CR context is described and the work of researchers from ComNets with relation to the corresponding topic is summarized. After a short introduction of the DARPA XG Program in

Section 4 and highlighting of the business impact of CR in Section 5 this article is completed in Section 6.

2. COGNITIVE RADIO STANDARDIZATION BODIES

A strong coordination between academia, regulation, infrastructure suppliers and operators is required to guarantee the wide acceptance and commercial success of CR. The standardization of CR is an essential part of this coordination that helps to channelize the joined, interdisciplinary research and development activities all over the world.

2.1 IEEE

The IEEE is in the forefront of CR standardization. The industrial relevance of CR in the US is the result from regulation authorities that advocated many years the liberalization of spectrum regulation. This founded the necessarily required industrial momentum for the development and commercial realization of CR in the US. Many IEEE standardization bodies are working on incremental steps towards the vision of CR as summarized in Table 1 and described in detail in Section 3.

Table 1. Overview on standardization activities towards CR within the IEEE.

IEEE Standardization Body	Cognitive Radio Feature/Building Block
SCC 41 / P1900	Series of general standards on Cognitive Radio
802.11n	Phased Coexistence Operation for phased 20/40 MHz channel usage
802.11s	Common Channel Framework for multi channel operation
802.11k	measurement, reporting, estimation and identification of characteristics of spectrum usage
802.11y	High power contention based medium access and a flexible spectrum management framework
802.19	Coexistence Technical Advisory Group
802.22	Wireless Rural Access Network for the (Re-) use of TV bands with a radio technology similar to IEEE 802.16 / WiMAX

2.2 SDR Forum

The SDR (Software Defined Radio) Forum, founded in 1996, is a non-profit international industry organization promoting the development, deployment and use of SDR technologies for advanced wireless systems. The commercial pillar of the SDR Forum's strategy foresees the creation of standards and specifications that will reduce costs and time-to-market for SDR and CR based systems and products. 3rd party standards supported by the SDR Forum are explicitly wanted and own specifications and standards only intended when no one else can or will.

2.3 ETSI – Reconfigurable Radio Systems

CR is likely to be based on SDR platforms, even though this is not a necessity. The ETSI regards these technologies as Reconfigurable Radio Systems (RRS) exploiting reconfigurable radio/ networks and self-adaptation to a dynamically changing environment, with the aim to ensure end-to-end connectivity. ETSI recently established a technical committee on RRS with the following responsibilities:

- to study the feasibility of standardization activities related to RRS encompassing radio solutions related to SDR and CR research topics
- to collect and define the related RRS requirements from relevant stakeholders
- to identify gaps, where existing ETSI standards do not fulfill the requirements, and suggest further standardization activities to fill those gaps

The RRS technical committee shall address (from the perspective of ETSI so far uncovered) topics in standardization like improved spectral utilization and inter-operator coexistence, using flexible usage modes and a variety of different Radio Access Technologies (RATs). As CR is related to national security aspects (like for instance flexible encryption algorithms) this European driven standardization effort might additionally have strategic motives from the EU member states to be independent form the US dominated IEEE.

3. COGNITIVE RADIO FEATURES IN IEEE STANDARDIZATION

3.1 IEEE SCC41/P1900

The IEEE Standards Coordinating Committee (SCC) 41 has initiated a series of standards, the IEEE 1900 series on next generation radio and advanced spectrum management, to stimulate the research and development of CR. SCC 41 evolved in April 2007 from the IEEE P1900 Standards Committee that was established in 2005. The scope of SCC 41 includes improvement of spectrum usage, new techniques and methods of dynamic spectrum access, interference management, coordination of wireless technologies, network management and information sharing.

The SCC 41 is currently divided into 5 WGs:

- IEEE 1900.1 on Terminology and Concepts for Next Generation Radio Systems and Spectrum Management
- IEEE 1900.2 on Recommended Practice for Interference and Coexistence Analysis
- IEEE 1900.3 on Recommended Practice for Conformance Evaluation of SDR Software Modules
- IEEE 1900.4 on Architectural Building Blocks Enabling Network-Device Distributed Decision Making for Optimized Radio Resource Usage in Heterogeneous Wireless Access Networks
- IEEE 1900.5 on Policy Language and Policy Architectures for Managing CR for Dynamic Spectrum Access Applications

3.1.1 IEEE P1900.1

As many research groups have defined the CR and related terms differently, P1900.1 is creating a glossary of terms and concepts in the context of CR. Technically precise definitions are standardized and explained to provide a common understanding on CR and to found a framework for the standardization efforts of the other IEEE SCC41 WGs.

Related Work @ ComNets:

An example for this is the notation of vertical and horizontal spectrum which was contributed to P1900.1 in [2].

3.1.2 IEEE P1900.2

One major objective of CR is to improve the overall efficiency of spectrum usage. For this reason, the accurate measurement of interference has become a crucial requirement for the deployment and evaluation of CR technologies. P1900.2 is therefore developing interference analysis criteria. Additionally, a framework for measuring and analyzing the spectral coexistence between different radio systems is established. The resulting common standard platform shall facilitate the resolution of (inevitable and necessary) disputes on the introduction of CR technologies. The flexibility and adaptability of spectrum usage stimulate fear and doubts in CR especially among incumbent license holders and conservative regulation bodies. The tradeoff between (i.) interference prevention in adaptive spectrum usage and (ii.) cost through precise electrical components shall be quantified. For this reason uncertainty levels in measurements and thresholds of harmful interference are defined by P1900.2.

3.1.3 IEEE P1900.3

SDR is an important part of CR. The P1900.3 WG is developing test methods for conformance evaluation of software for SDR devices. A set of recommendations is defined to assure the coexistence and compliance of the software modules of CR devices before proceeding toward validation and certification of final devices. P1900.3 wants to increase confidence in SDR devices. SDR will comprise different layers of software with different functionalities each to be validated on their conformance to regulatory and operational requirements. The P1900.3 WG is designing testing procedures that will comply with the semiformal software specifications in defining for instance checkpoints and assertions that reflect the specification. Guidelines for operators and manufacturers of SDR technologies shall be established to enable a licensing by regulation authorities.

Related Work @ ComNets:

An initial step towards a semiformal software specification is presented in [3] as a framework for building re-configurable protocol stacks. A high degree of re-configurability is achieved through composing complex behavior of a communication system using Functional Units. A uniform interface allows these units to be connected to form a Functional Unit Network. This work has its origin in [4] and [5] where a design technique for enhancing existing and developing future 3G and 4G protocol stacks is proposed referred to as generic protocol stack.

3.1.4 IEEE P1900.4

The IEEE P1900.4 WG is standardizing means for distributed decision making in heterogeneous wireless networks to optimize the usage of radio resources. Therefore, network and devices resource managers as well as the communication exchange between them are developed. In a first stage, the standard is limited to the architectural and functional definitions while the corresponding protocols are defined in a later stage. The focus on CR capabilities in the context of heterogeneity in wireless access technologies differentiates this WG from other WGs of SCC41. The Cognitive Pilot Channel (CPC) concept is an example for the work of P1900.4. The CPC assists a reconfigurable and cognitive terminal in selecting a specific radio access technology in a wireless communication environment of different access networks with varying spectrum allocations.

3.1.5 IEEE P1900.5

A policy language (or a set of policy languages or dialects) together with a corresponding policy architecture is developed by the youngest member of SCC41, the IEEE P1900.5 WG. A policy framework is specified for an interoperable, vendor-independent control of CR functionality and behavior for dynamic spectrum access. Initial efforts will concentrate on the features required for bounding a policy language to one or more policy architectures. Thereafter, concrete details with a special focus on interoperability are intended to be standardized.

Related Work @ ComNets:

Policy-defined spectrum sharing and medium access for CRs are presented in [6]. This article primarily discusses an approach that intends to enable distributed QoS support in open spectrum. This algorithm is specified as policy in a machine-understandable policy description language, such that the CR is capable of reasoning about spectrum usage. Policies that enable a software defined medium access are the second focus of this article.

3.2 IEEE 802.11n

The high throughput features of IEEE 802.11n imply among others also a (optional) CR feature, namely the Phased Coexistence Operation (PCO). It is designed to allow an 802.11n Access Point (AP) to support 802.11n and 802.11a (and b/g) clients. For this, PCO implements a phased operation with flexible channel bandwidths of 20 and 40 MHz. Two neighboring 802.11 channels, with each 20 MHz bandwidth, are categorized as a primary and a secondary channel. With PCO, the 802.11n AP is able to switch between two times 20 MHz and 40 MHz phases of operation. Thus with PCA an 802.11n AP can support and control legacy devices in the two 20 MHz channels as well as benefit from 40 MHz operation when communicating with 802.11n stations. The PCO of 802.11n is an initial step towards multi channel operation with flexible bandwidth usage, here centrally coordinated on the level of the MAC layer.

The PCO of 802.11n is illustrated in the MAC timing diagram of Figure 1. Two orthogonal neighboring channels with each 20 MHz bandwidth are depicted. Channel f_1 is marked as primary channel through a respective beacon transmission while channel f_2 is regarded as secondary channel.

Additionally, the Network Allocation Vectors (NAVs) of two legacy 802.11 (20MHz) stations operating on f_1 and f_2 respectively are shown. In the last row, the combined channel view of an 802.11n (40 MHz) station is depicted. A centrally operating 802.11n AP is setting the NAVs of all stations with adequate control frames transmitted on the respective channel for channel reservation. The AP is thus coordinating the phased operation of the associated stations with 20 and 40 MHz bandwidth usage.

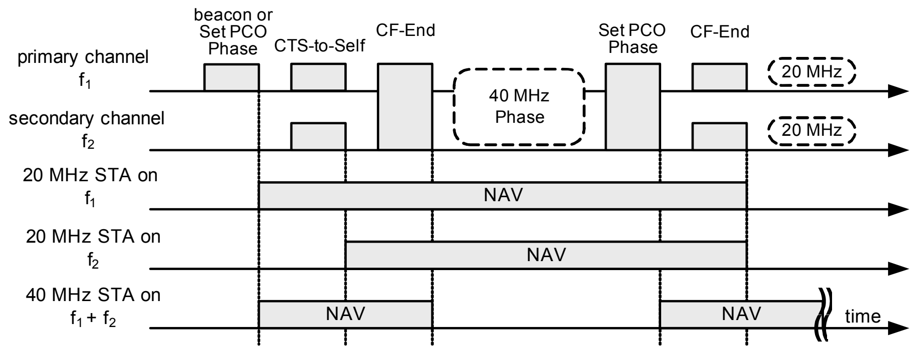

Figure 1. Timing diagram of the 802.11n Phases Coexistence Operation (PCO).

3.3 IEEE 802.11s

The mesh extension of IEEE 802.11 are currently standardized by the 802.11s WG. The primary scope of 802.11s, the creation of a wireless distribution system with automatic topology learning and wireless path configuration, implies many CR aspects. An awareness of the local communication environment is required to allow a dynamic, radio-aware path selection in the mesh. An optional feature of 802.11s, namely the Common Channel Framework (CCF), introduces a coordination channel and the corresponding MAC protocol enhancements for using different orthogonal frequency channels. Contrary to the PCO of 802.11n , these channels are not required to be neighboring. With the CCF, stations periodically switch to the common channel for distributed channel reservation through suggestion and acceptance of destination channels.

The principle of the multi channel MAC protocol implemented through the CCF is illustrated in Figure 2 with a MAC timing diagram. Channel f_0 is used as common coordination channel. On this common cannel f_0, 802.11s stations coordinate their data transmissions on the data channels f_1 to f_n. This is done in a dedicated phase, referred to as Channel Coordination Window, with Request-to-Switch (RTX) and Clear-to-Switch (CTX) messages. The Channel Coordination Window has a Repetition Period which enables, as depicted, normal data transmissions outside this window on the common channel.

Related Work @ ComNets:

Spectrum sharing in IEEE 802.11s wireless mesh networks is evaluated in [7]: The well-known IEEE 802.11 DCF is compared to a distributed, reservation-based approach from the Mesh Network Alliance (MNA).

The contention-based distributed medium access of 802.11 and its extension for the usage in multiple parallel frequency channels is highlighted in [8]. A multi-channel station groups channels for increasing its own achievable throughput. Additionally, methods to enable the coexistence of multi-channel stations and single-channel are introduced.

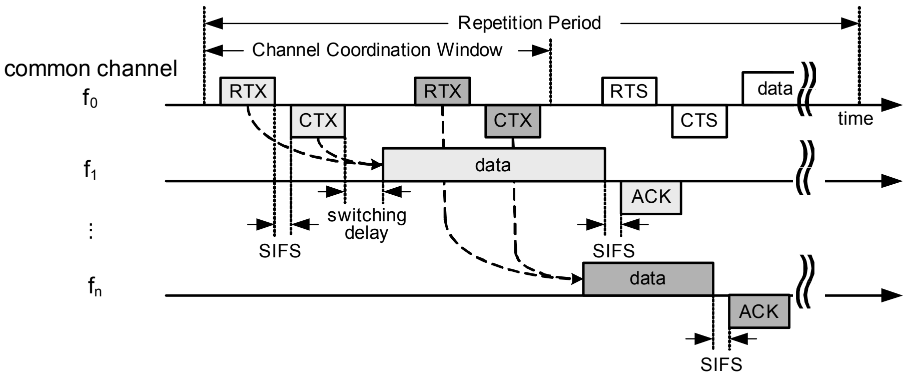

Figure 2. MAC timing diagram of the (optional) 802.11s Common Channel Framework (CCF).

3.4 IEEE 802.11k

IEEE 802.11k is an amendment to the IEEE 802.11 base standard for radio resource management and was finally approved in May 2008. It provides means for measurement, reporting, estimation and identification of characteristics of spectrum usage. 802.11k improves spectrum opportunity identification in unlicensed bands in unpredictable environments and is able to characterize the interference on different frequency channels. New measurements for:

- Channel load (busy / idle fraction)
- Noise histogram
- Neighbor beacons and
- Location

are identified. The measurement reports and frames are standardized in 802.11k, but not the algorithms that use them. This remains an open research topic. 802.11k can be regarded as a basis for the decision making of future CRs based on measuring local spectrum usage.

As an example of radio resource measurements in 802.11k, the Medium Sensing Time Histogram (MSTH) is illustrated in Figure 3. A density diagram of reported Clear Channel Assessments (CCA) times is the basis for the MSTH as shown in the upper right corner. The MSTH is a density vector of 6 values, each representing the density of specific 802.11 MAC idle times. The basis interval for the 802.11 MAC timing is aSlotTime (for 802.11a 9us). The densities are indicators for the probability *p* of occurrence of multiples of this aSlotTime, like SIFS of DIFS. After sensing, the results are reported with standardized MAC frames to the AP and neighboring stations. Such an exemplary 802.11k MSTH reporting frame is also shown in the lower part of Figure 3.

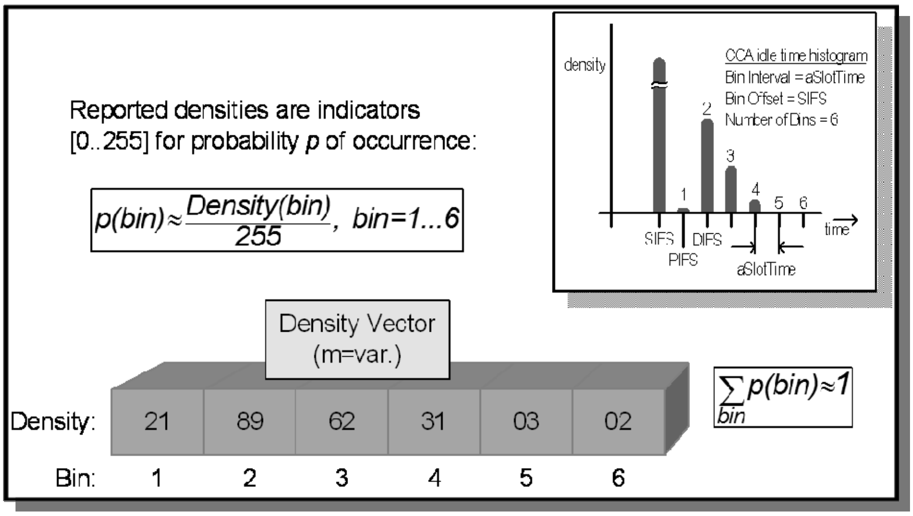

Figure 3. Medium sensing time histogram of 802.11k.

Related Work @ ComNets:

Spectrum awareness for distributed resource sharing in IEEE802.11e/k is described in [9]. Radio resource measurements for opportunistic spectrum usage on the basis of 802.11k are analyzed in [10]. The improvement of confidence in radio resource measurements as approach to reliability of spectrum opportunity identification is considered in [11].

3.5 IEEE 802.11y

At the time this article is written, IEEE 802.11y can be considered as one of the younger WGs of 802.11. 802.11y is working towards a standard for high power Wi-Fi equipment operating in the 50 MHz frequency band at 3.650-3.700 GHz. The standardization is intended to be finalized in September 2008. The FCC issued final rules for a novel "light licensing" scheme in June 2007 for the 3650-3700 MHz band: A small fee for a nation-wide, non-exclusive license is required. Additionally, a nominal fee for each high powered Base Station (BS) that is deployed has to be paid. Contention based protocols are mandatory to allow fair and opportunistic spectrum sharing in case of interference between licensee's devices.

802.11y introduces a combination of higher power limits and enhancements made to the PHY/MAC timings of the 802.11 base standard that allow Wi-Fi devices to operate at distances of more than 5 km. Additionally, a flexible spectrum management framework of regulatory classes is defined for adopting coverage and spectrum usage according to local regulation requirements. Further, 802.11y is introducing smaller channel bandwidths to Wi-Fi as it is capable of operating with 5, 10 and 20 MHz channel bandwidth. The 802.11y Extended Channel Switch Announcement (ECSA) is an enhanced Dynamic Frequency Selection (DFS) procedure and provides the means for notifying stations of changing channels and channel bandwidth. An operator can extends and retract permissions to license exempt devices (referred to as dependent STAs in 802.11y) to use licensed radio spectrum in applying the Dependent Station Enabling (DSE). This mechanism guarantees an operation in licensed spectrum depending on approval by periodic messages received from the operator´s BS. Further, DSE can be extended to channel management, coordination. A location dependent spectrum access and interference resolution is implemented based on the location of the access granting BS. Thus no locating technologies like GPS are required in the 802.11y STA (price, device complexity and indoor failure are reduced).

The light licensing concept of the 802.11y spectrum management framework is not limited to the operation at 3.6 GHz, instead it provides the general means to Wi-Fi of being operated at any spectrum as for example the candidate bands for IMT-Advanced like 450-862 MHz or 2300-2400 MHz. As consequence, 802.11y is also a promising competitor of 802.22 for the reuse of TV bands.

Table 2. Example behavior sets [12].

Behavior limits set	Topic	USA	Europe	Japan
3	transmit power control	Reserved	ETSI EN 301 389-1	Reserved
4	dynamic frequency selection	Reserved	ETSI EN 301 389-1	Reserved
6	4 ms Carrier Sensing	4 ms, no exceptions	Reserved	MIC EO Articles 49.20, 49.21
9	public safety	FCC 47 CFR [B8], Section 90.1209	Reserved	Reserved
10	Part 15 license exempt bands	FCC 47 CFR [B8], Section 15.247	ETS 300-328	MIC EO Article 49.20

Table 3. Example regulatory classes in the US [12].

Regulatory Class	Channel Starting Frequency [GHz]	Channel Spacing [MHz]	Channel set	Transmit Power limit [mW]	Transmit Power limit [EIRP]	Emissions Limits set	Behavior Limits set
10	4.85	20	21, 25	100	-	5	9
11	4.85	20	21, 25	2000	-	5	9
12	2.407	25	1-11	1000	-	4	10
13	3.000	20	133, 137	-	1 W/MHz	6	3, 4, 6, 11, 15
13	3.000	20	133, 137	-	40 mW/MHz	6	3, 4, 5, 6, 12, 15
14	3.000	10	132, 134, 136, 138	-	1 W/MHz	6	3, 4, 6, 11, 15
14	3.000	10	132, 134, 136, 138	-	40 mW/MHz	6	3, 4, 5, 6, 12, 15

3.6 White Spaces Coalition & Wireless Innovation Alliance

The White Spaces Coalition (founded in 2006) and the Wireless Innovation Alliance (founded in 2008) are both planning to provide high speed broadband internet access to consumers via existing "white spaces" in unused analog television frequencies, i.e. the digital dividend. These groups intend to challenge incumbent telephone and cable companies with a fixed, long-distance wireless broadband technology. Dell, Google, HP, Microsoft, Motorola are among other participating in both groups. While for instance Intel, Philips, Samsung and Earthlink are limiting their involvement to the White Spaces Coalition.

Scientific publications and strong attendance in standardization of considerable researchers allows the conclusion that these two groups will apply Wi-Fi as base technology for the secondary access to TV bands. In this context, 802.11y from above forms an adequate basis of technical features.
Obviously, the time consuming and thus expensive process of IEEE standardization is avoided to the benefit of a short time-to-market. The goal seems to be an outrunning of 802.22 and coming up with commercial products to the opening of the digital dividend early 2009. Microsoft, as member of the White Spaces Coalition, submitted an initial prototype to FCC for testing in 2007 and had to go through some cycles of resubmissions due to objections by the FCC.

Related Work @ ComNets:

The paper [12] describes a dual beaconing approach for realizing an operator assisted CR. This approach can be applied for implementing the reliable reuse of TV bands with enhanced Wi-Fi beacons. This is one of the key building blocks of the future radio internet developed in the European research project All-Wireless Mobile Network Architecture (WIP).

3.7 IEEE 802.19

Many of the wireless communication systems standardized by the IEEE are operating in unlicensed spectrum. Within unlicensed frequency bands, radio systems coordinate the usage of radio resources autonomously while operating.

The IEEE 802.19 WG calls itself Coexistence Technical Advisory Group. 802.19 is aiming at the development and maintenance of policies defining the responsibilities of IEEE standardization efforts to consider coexistence with existing standards and other standards under development. If demanded 802.19 evaluates the conformance of the developed standard to the coexistence policies and offers a documentation of the coexistence capabilities to the public.

Related Work @ ComNets:

A framework for defining coordination rules for the radio resource management, referred to as spectrum etiquette, is discussed in [14]. Spectrum etiquette rules for radio systems with different channel bandwidths are defined. The rules are based on a set of actions like channel selection and listen-before-talk. By evaluating the rules with help of simulations, [14] provides an initial approach towards a spectrum etiquette proposal.

3.8 IEEE 802.22

IEEE 802.22 WG is targeting the standardization of a cognitive air interface for fixed, point-to-multipoint, Wireless Regional Area Networks (WRANs) operating on unused channels in the VHF/UHF TV bands between 54 and 862 MHz. Thereby, 802.22 will provide wireless broadband access from a BS to rural areas over distances of typically 15 to 30 km in serving up to 255 fixed Customer Premise Equipments (CPEs). The technical basis of 802.22 is WiMAX with the following key characteristics: OFDMA, TDD, a 10 ms MAC frame, channel bandwidth of 6, 7, 8 MHz and peak data rate of 72.6 Mbps (with optional channel bonding and channel aggregation).

The Geographic data bases and location technologies are not part of 802.22. Instead, incumbent awareness and interference prevention are realized as part of MAC and PHY through:

- Distributed spectrum sensing and spectrum management
- Quiet period and fast/fine sensing management
- Measurements and clustering
- Detection algorithms

All 802.22 devices (BS and CPE) sense the spectrum for three different licensed transmissions: (1.) analog television, (2.) digital television and (3.) licensed low power auxiliary devices such as wireless microphones. A Coexistence Beacon Protocol (CBP) based on beacon transmissions among the spectrum sharing WRAN cells implements decentralized coexistence coordination in a self-coexistence phase. Additionally, centralized coordination via the BS is implemented with the BS beacon being part of the Superframe Control Header (SCH).

A so-called spectrum manager implements CR functions at the BS. It will use inputs from a Spectrum Sensing Function (SSF), geolocation and an incumbent database to decide on the TV channel usage as well as the EIRP limits imposed to specific 802.22 CPEs.

The 802.22 MAC frame structure in the time and frequency domain is illustrated in Figure 4. An 802.22 Superframe has a duration of 160 ms and

consists of 16 MAC frames. Each frame is divided into a downstream (DS) subframe and an upstream (US) subframe with an adaptive boundary in between. The DS subframe contains a single PHY PDU addressed to multiple 802.22 CPEs. Contrary, the US subframe may contain multiple PHY PDUs from various CPEs. Additionally, contention intervals for initialization, bandwidth request and Urgent Coexistence Situation (UCS) notifications are preceding the PHY PDUs in the US subframe. The dedicated self-coexistence phase is located at the end of each 802.22 MAC frame.
In order to apply 802.22 in Europe, the TV signal detection and channel bandwidths has to be adapted to European TV broadcast standards.

Related Work @ ComNets:

An approach to decentralized coordination of reservations on a dedicated coordination channel is described in [15]. Spectrum Load Smoothing is applied for the re-use of TV-bands. It allows prioritization and protection of incumbent signals in secondary spectrum access.

4. DARPA NEXT GENERATION COMMUNICATIONS PROGRAM

The Defense Advanced Research Projects Agency (DARPA), has established a program referred to as Next Generation Communication (XG) Program. The DARPA XG Program can be considered as the number one source of innovation and major driver of CR.

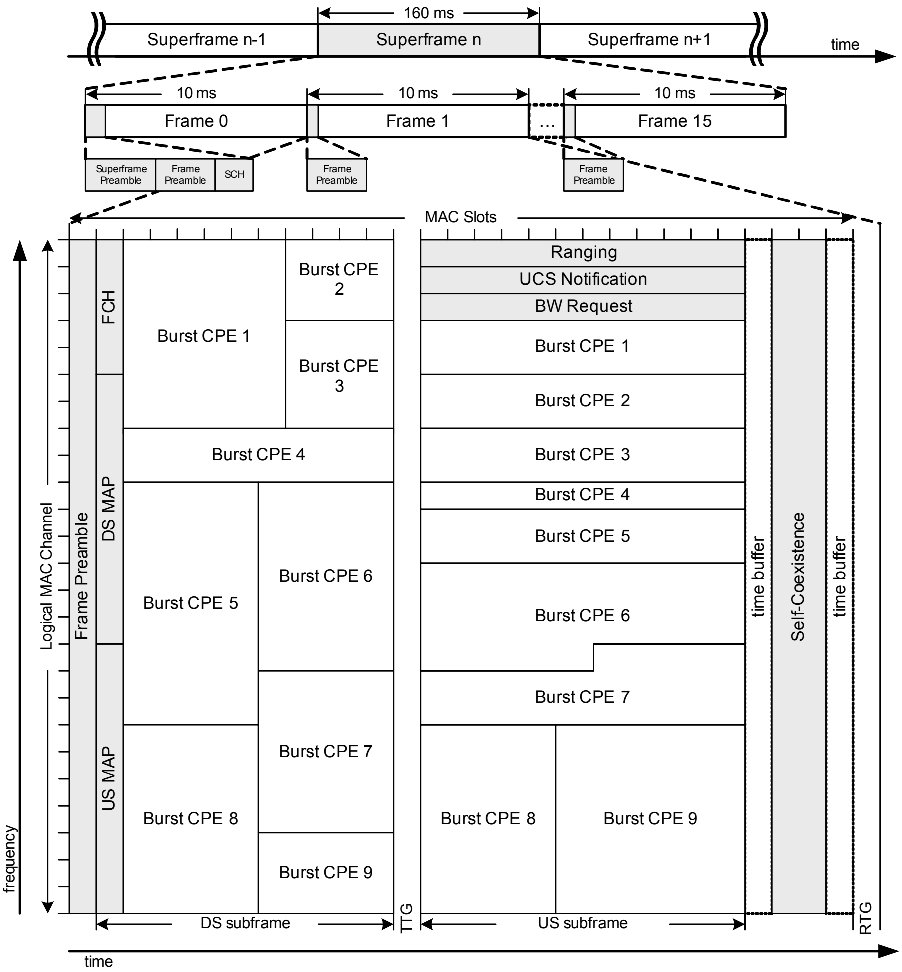

Figure 4. MAC frame composition of 802.22. The time and frequency domain are divided into MAC slots and logical MAC channels in implementing the OFDMA of 802.22.

The XG concept of CR is based on so-called “abstract behaviors, protocols, and a policy language”. The reasons for this approach are mainly “flexibility,” “long-term impact,” and the need for traceability, i.e., regulatory approval of the rules being used by CRs. In other words, behaviors are used instead of detailed descriptions of a standardized protocol, or a set of different standardized protocols, to allow regulators and industry to dynamically align future regulatory requirements and rules for spectrum usage with existing and emerging technologies for future radio systems. Policies use a policy meta language as utility. There is a direct association between policies and technical constraints. Abstract behaviors are derived from policies. A behavior is composed by core behaviors. Protocols are derived from behaviors, realized by the real implementation [16].

Related Work @ ComNets:

An early version of the DARPA XG policy language is applied in [16] to specify spectrum sharing algorithms derived from game theory and water-filling.

5. BUSINESS IMPACT OF CONITIVE RADIO

The much needed introduction of CR will provide new opportunities in the business models in wireless communication. We summarize some basic thoughts on the implications of CR for incumbent operators in a SWAT analysis shown in Table 4. The main advantage of CR for operators is the elimination of the longsome and expensive licensing process. Operators have the chance to realize a multitude of new services. The complexity of devices, primary radio protection and unsatisfying QoS support are technical problems that are currently discussed in the research world. A major threat for incumbent network operators is the breakdown of market entry thresholds. The highly paid exclusiveness of spectrum access to certain frequency bands may be lost and the investment costs for building up a radio network will not consider spectrum license fees anymore. New competitors are then able to easily enter the market and can attack incumbent operators. Existing licenses will in the long term expire without being renewed by the regulator. A timely introduction of operator assisted CR services and coordinated secondary spectrum access are a promising answer to deal with this threat. More details can be found in [17].

Table 4. SWOT analysis of CR from incumbent operator's (today's license holders) perspective.

Strengths	**Weaknesses**
• improved spectrum efficiency • eliminated need for expensive and lengthy spectrum licensing process • more flexible spectrum assignment • less technology dependent access to spectrum • seriously considered by U.S. regulators → potential leverage effect	• protection of licensed radio systems against interference may not be reliable • may not be sufficient for services that require restrictive QoS • low acceptance by license owners and regulators • increased complexity of devices • additional infrastructure may be required • multiple value chain participants
Opportunities	**Threats**
• new types of services like operator assisted dynamic spectrum assignment (to protect licensed services and to enable QoS) may provide new revenue stream • spectrum trading (lease, reselling etc.) and inter-operator spectrum sharing • potential for new very high bitrate systems (Gb/s)	• thresholds for entering market of commercial wireless communication are lowered • new upcoming competitors • cognitive radio devices may be allowed to operate in incumbent operator's spectrum • operators may lose competitiveness if trend is missed

6. CONCLUDING REMARKS & ACKNOWLEDGEMENT

In this article ongoing CR standardization activities are introduced. Evolutionary building blocks towards the realization of the CR vision are currently under development. In putting the discussed building blocks together, a promising potential can be identified: The research and development in the field of CR is building up momentum for commercial break through. But without CR advocating regulation bodies and the general continuous willingness to liberalize spectrum regulation the CR community might fail to leverage this momentum. Joint efforts and an adequate timing within the dynamic telecommunication market will support the success of the CR community.

The authors would like to thank Prof. Dr.-Ing. B. Walke for the many years of strong support, inspiring discussions and fruitful collaboration leading to a multitude of reputable joint publications.

7. DISCLAIMER

Lars Berlemann is with Deutsche Telekom, Product and Innovation and TU Dortmund. Stefan Mangold is with Swisscom Switzerland and with ETH Zurich, Dept. of Computer Science. The comments and statement made in this article are from the authors and do not necessarily reflect the official position of their employers.

References

[1] Berlemann, Lars and Mangold, Stefan: Cognitive Radio and Dynamic Spectrum Assignment. New York U.S.A.: John Wiley & Sons, 2008.

[2] Berlemann, Lars and Mangold, Stefan (2007) Contribution to P1900.1, 05/2007

[3] Schinnenburg, Marc and Debus, Fabian and Otyakmaz, Arif and Berlemann, Lars and Pabst, Ralf: A Framework for Reconfigurable Functions of a Multi-Mode Protocol Layer. In Proceedings of SDR Forum 2005, p. 6, Los Angeles, U.S., 11/2005

[4] Berlemann, Lars and Siebert, Matthias and Walke, Bernhard: Software Defined Protocols Based on Generic Protocol Functions for Wired and Wireless Networks. In Proceedings of SDR Forum 2003, 11/2003

[5] Siebert, Matthias and Walke, Bernhard: Design of Generic and Adaptive Protocol Software (DGAPS). In Proc. of the Third Generation Wireless and Beyond (3Gwireless '01), San Francisco, US, 06/2001

[6] Berlemann, Lars and Mangold, Stefan and Hiertz, Guido R. and Walke, Bernhard: Policy Defined Spectrum Sharing and Medium Access for Cognitive Radios. In Journal of Communications (JCM), p.p. 1-12, 04/2006

[7] Max, Sebastian and Hiertz, Guido R. and Weiss, Erik and Denteneer, Dee and Walke, Bernhard: Spectrum Sharing in IEEE 802.11s Wireless Mesh Networks. In Computer Networks, Vol.51, No. 9, p.p. 2353-2367, New York, NY, USA, 06/2007

[8] Mangold, Stefan and Habetha, Jörg: IEEE 802.11 Contention-Based Medium Access for Multiple Channels. In Proceedings of 11th European Wireless Conference 2005, Vol.2, p.p. 522-526, Nicosia, Cyprus, 04/2005

[9] Mangold, Stefan and Zhong, Zhun and Hiertz, Guido R. and Walke, Bernhard: IEEE 802.11e/802.11k Wireless LAN: Apectrum Awareness for Distributed Resource Sharing. In Wireless Communications and Mobile Computing 2004, Vol.4 Issue 8, p.p. 881-902, 12/2004

[10] Mangold, Stefan and Zhong, Zhun and Callapali, Kiran and Chou, C. T.: Spectrum Agile Radio: Radio Resource Measurements for Opportunistic Spectrum Usage. In: 47th annual IEEE Global Telecommunications Conference, Globecom 2004, Dallas TX, USA, 11/ 2004

[11] Mangold, Stefan and Berlemann, Lars: IEEE 802.11k: Improving Confidence in Radio Resource Measurements. In Proceedings of PIMRC05, p. 5, Berlin, Germany, 09/2005

[12] IEEE P802.11y™/D11.0, Draft Standard for Information Technology - Telecommunications and Information Exchange between Systems - Local and Metropolitan Area Networks - Specific requirements - Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, Amendment 3: 3650–3700 MHz Operation in USA, 06/2008

[13] Mangold, Stefan and Jarosch, Andreas and Monney, Claude: Operator Assisted Cognitive Radio and Dynamic Spectrum Assignment with Dual Beacons - Detailed Evaluation. First International Workshop on Wireless Personal and Local Area Networks. New Delhi, India, 01/2006

[14] Mangold, Stefan and Challapali, Kiran: Coexistence of Wireless Networks in Unlicensed Frequency Bands. In Proceedings of the WWRF9, Zurich, Switzerland, 07/2003

[15] Berlemann, Lars and Hiertz, Guido R. and Walke, Bernhard: Reservation-based Spectrum Load Smoothing as Cognitive Medium Access for Spectrum Sharing Wireless Networks. In Proceedings of European Wireless Conference 2005, Vol.2, p.p. 547-553, Nicosia, Cyprus, 04/2005

[16] Berlemann, Lars and Mangold, Stefan and Walke, Bernhard: Policy-based Reasoning for Spectrum Sharing in Cognitive Radio Networks. In 1st IEEE Symposium on New Frontiers in Dynamic Spectrum Access Networks, p.p. 1-10, Baltimore MD, USA, 11/2005

[17] L. Berlemann, S. Mangold, A. Jarosch and B. Walke, Cognitive Radio: Business Impacts of Dynamic Spectrum Access and Spectrum Sharing, VDE Congress 2006, Aachen, Germany, 10/2006

[18] Walke, Bernhard and Mangold, Stefan and Berlemann, Lars: IEEE 802 Wireless Systems: Protocols, Multi-Hop Mesh/Relaying, Performance and Spectrum Coexistence p. 368, John Wiley & Sons, 11/2006

COEXISTENCE AND RADIO RESOURCE OPTIMIZATION OF WIRELESS NETWORKING TECHNOLOGIES

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Abstract Wireless networks have to meet increasing demands in the future: high data rates for each user, high spectral efficiency and a range of different quality-of-service requirements for different applications. The systems, furthermore, should provide high energy efficiency to minimize interference and provide a fair share of the channel resources to provide coexistence between neighboring systems built on top of the same or even different radio network technologies. Another objective is to maximize the battery lifetime of mobile devices and at the same time reduce the electromagnetic impact on the environment (electric smog). This goal can only be achieved by a dynamic allocation of wireless resources. These issues are covered by three projects: In the CoCoNet project, the focus is on the coexistence between wireless LANs which control their spectrum access dependent on interference measurements experienced from neighboring networks and in this way provide a fair coexistence between each other. The project XLayer considers the resource allocation by an access point which sends data to a number of stations over the downlink. In PoSSuM, the coexistence of WLAN systems introduced in the CoCoNet project is extended to a cooperation of such systems inside meshed networks. This paper gives an overview of the three projects.

1. INTRODUCTION

The demand for high-performance wireless communication platforms is increasing. Time-critical applications like video conferencing require maintaining communication links with providing quality-of-service parameters concerning e. g. throughput, delay or jitter. The rapid deployment of mobile devices results in a high density so that mutual interference becomes more likely which can degrade the performance of the communication. Furthermore, wireless technologies are not only used for the access of end-user terminals in wireless local area networks (WLANs), but also as a backbone in wireless metropolitan area networks (WMANs). Both technologies can be combined to form a fully wireless infrastructure, for example by connecting buildings via an IEEE 802.16 WMAN meshed network and by providing access inside the building by an 802.11 WLAN. Base stations inside buildings then have to be equipped with interfaces for both technologies. This scenario requires coordination on two levels: on the one hand, the WMAN interface which is part of the meshed network has to allocate channel capacities to the neighboring stations according to the amount of data traffic which needs to be transmitted to each of the neighbors. On the other hand, the WMAN interface has to share the frequency spectrum with the WLAN interface so that coordination between the backbone network and the user access is required as well. An issue which has to be considered in all these situations is that the user terminals are mobile so that channel occupation and interference is time-variant.

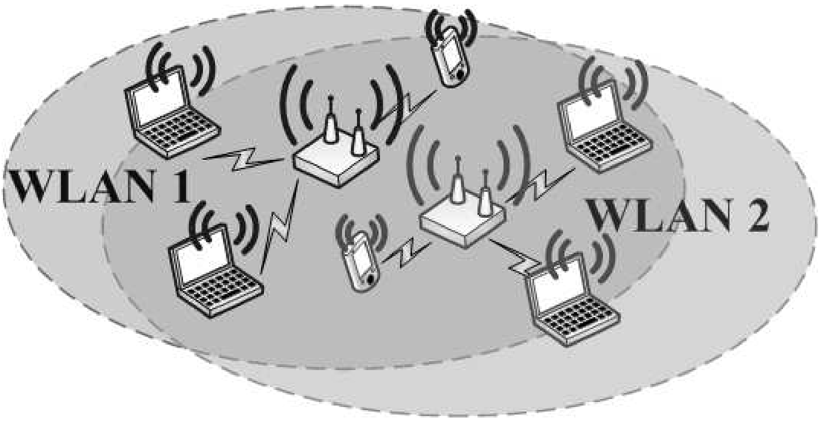

Figure 1. Two WLANs compete for the access to a single, shared frequency [1].

In general, the problems which are investigated in the context of the projects described in this paper deal with the shared access of multiple wireless networks which is illustrated in Figure 1. To cope with this issue, a centralized or rather a decentralized resource management of the available frequency spectrum is needed. The effect of such a resource management is the allocation of spectrum according to certain policies to different wireless connections which contend for the same resource. These policies can either be hardcoded into a wireless station or it can be configurable. In the latter case, the policies need to be defined by a formal description language; they are loaded into the wireless station as a set of rules. An individual connection is characterized by a number of transmission parameters: frequency, transmit power, time interval when the transmission can take place, occupied radio frequency bandwidth, modulation scheme and so on. Spectrum management ensures that two interfering connections modify their transmission parameters so that the interference is reduced. For this allocation, fairness has to be considered; all connections which contend for transmission should be allocated a sufficient amount of resources so that quality-of-service requirements of the application can be met. A strict QoS guarantee is, however, not always possible, for example if the MAC protocol works on a best-effort basis as it is the case for legacy 802.11. In this latter case, only a relative weighting between concurrent connections is possible, rather than enforcing a definite access to the channel at given time.

2. COEXISTENCE OF COOPERATING NETWORKS

Centralized and decentralized approaches how the spectrum usage between different wireless local area networks can be optimized were investigated in the CoCoNet project. The basis of all methods is an estimation of the current interference and channel allocation in the environment of the receivers and, deduced from this, an adaptive tuning of different transmission parameters of the ongoing communication such as frequency, transmit power or point of time when the transmission starts. The goal is a cooperation between the individual networks and stations, where each station tries to minimize the interference resulting from its transmissions and might have to accept compromises between its own demand for transmission capacity and the actual occupation of the transmission media.

By means of coexistence-supporting methods the quality of service of the transmission can be enhanced even in case of a heavy load of the system by uncoordinated systems which becomes more important due to the increasing deployment of wireless LANs. When the CoCoNet project started, there was little awareness of the problem of spectral coexistence and no protocols were available. Nowadays, also due to the contributions of the CoCoNet project, coexistence between wireless systems has become a major research topic which is termed *Cognitive Radio*. An important focus of CoCoNet was the coexistence between IEEE 802.11 and Hiperlan/2 (H2) which both work in the 5 GHz Industrial, Scientific and Medical (ISM) band. While the H2 technology is capable of supporting quality-of-service on demand, the legacy 802.11 did not support such mechanisms. In the CoCoNet project, methods for the coexistence between and within the systems were developed. The coexistence of 802.11 systems has turned out to be the major challenge as H2 could not be established on the market. In the CoCoNet project, protocol extensions were proposed which provide a fair share of the channel between H2 and 802.11 stations. This can for example be achieved in a way that wireless stations of both architectures are controlled by channel reservation mechanisms of an H2 or 802.11 access point.

The problem of coexistence among stations of the same architecture exists, for example due to access points in neighboring hotspots or apartment houses. The legacy 802.11 does not have any means for a suitable coordination between the neighboring networks. Neighboring access points try to optimize the transmission for "their" stations and can while doing so collide with stations in neighboring networks, because the access points are not aware of each other. The developed algorithms for the coexistence of H2 and WLAN could be applied for the coexistence between 802.11 networks and provided the basis for the 802.11e extension. In order to provide fairness between a number of competing stations which share the same channel, game theoretical approaches are investigated in [2] which control the amount of air time that each of the stations occupy. Figure 2 gives an overview of the mechanisms running inside a wireless station inside which the channel access is controlled by game-theoretic methods, Other options to optimize the spectrum access are defined by the IEEE 802.11h standard as transmit power control (TPC) and dynamic frequency selection) (DFS). Since the standard only specifies the signalling and packet formats, methods for TPC [3] and DFS [4] were developed.

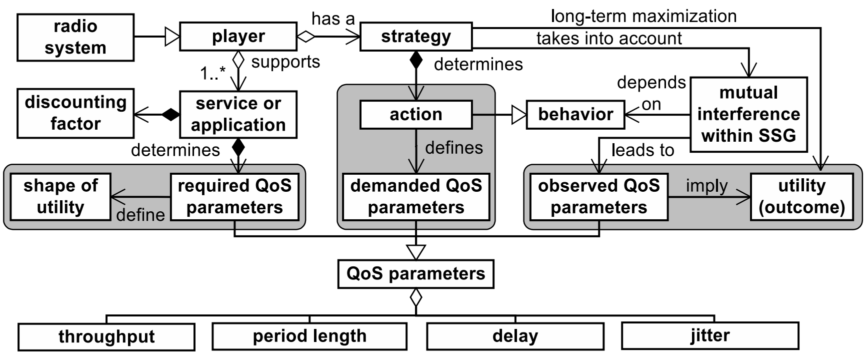

Figure 2. Model of the game-theoretic approach in UML notation [1].

It was found that the developed methods have a big potential to be used with WiMAX or 3GPP-LTE systems which require a decentralized coordination of the spectrum usage. The suitability of the spectrum management methods for WiMAX resulted in the launch of the new project POSSUM which is described in more detail in the next section. The results from the COCONET project were also used to contribute to the IEEE 802.11e standardization, for example in [5].

Besides the horizontal spectrum management discussed up to now where stations belonging to the same network architecture coordinate each other, the vertical spectrum management was also a topic of the project. In this case, every station can allocate any resource on any frequency band provided that they do not interfere with other radio services.

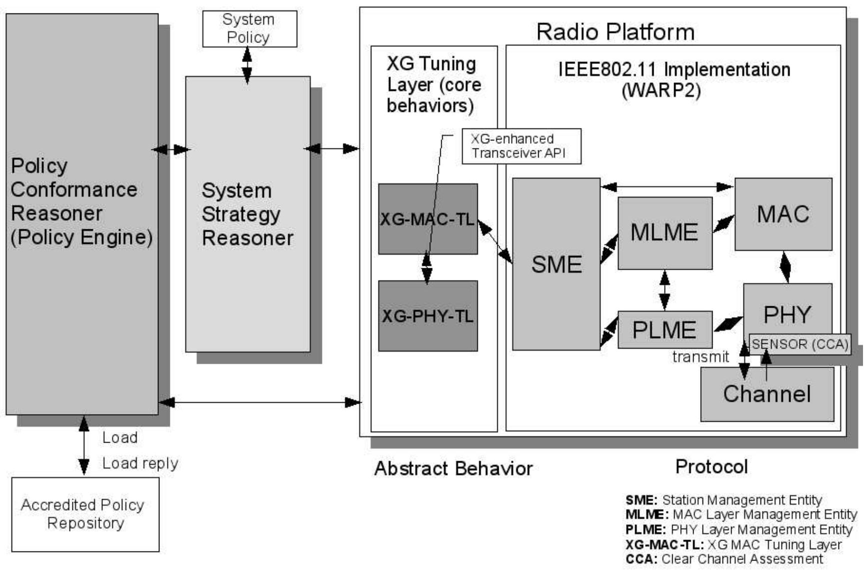

Figure 3. Extension of the 802.11 protocol stack by the DARPA XG architecture.

A framework for such an architecture was provided by the Defense Advanced Research Project Agency (DARPA) in the Next Generation Communication Program (XG). This architecture is based on an opportunistic spectrum usage based on decision policies and an abstract description of the behavior of a station by means of these rules, which is mapped to the physical architecture of the station. In case of 802.11, an abstract value of a maximum channel occupation of 50% can for example be achieved by adapting the length of the interframe spacing before a packet is transmitted. In a feasibility study, it was discussed how the 802.11 protocol stack can be embedded into the framework of a DARPA XG architecture and be adapted for the cooperation with DARPA XG [6], see also Figure 3. Extensions need to be applied to the 802.11 protocol stack so that transmission parameters which are usually determined by the software implementation of the protocol can be obtained from the XG decision policy entity; furthermore, the detection of a free channel provided by the 802.11 stack is shifted to the environment so that signalling is needed for this feature as well. Furthermore, an abstraction layer has to be provided which maps the transmission parameters determined by the policies into 802.11 specific parameters.

The approach defined by the DARPA XG program can be generalized under the topic *cognitive radio* [7] which resulted in the establishment of the POSSUM project described below.

3. CROSS-LAYER DESIGN AND OPTIMIZATION

While the goal of COCONET is the coexistence of neighboring networks which share the same frequency band, another important aspect is the optimization of the resource allocation inside a network considering quality-of-service requirements of the applications which is investigated in the XLAYER project. Application requirements are widely dependent on the application type. As an example, video and audio conferencing requires a minimum throughput so that the data can be transmitted with a given frame rate and screen resolution. It is, however, acceptable if occasionally a data packet is lost. On the other hand, file transfers do not have strict delay constraints; some amount of throughput should, however, be available because the user would not be ready to wait for large amounts of time until the download is complete. The transmission must, however, be performed without packet loss. These requirements can only be met if a channel access scheme is used where one dedicated station has full control over the channel access and can allocate air-time to other stations inside its range. Inside the scope of IEEE 802.11 based networks, the access point maintains this task based on a cross-layer scheduler: Multiple layers inside the protocol stack are included into the allocation of air time to the users which is depicted in Figure 4. The separation between the layers required by the OSI reference model results in loss of control information which can be used to enhance the performance: The MAC scheduler needs to know the requirements of the applications to serve the respective data flows appropriately. The knowledge of the MAC layer packet priorities supports the resource allocation of the physical layer. The PHY layer reports back to the MAC layer which packets could be successfully transmitted which is considered by the scheduler. Finally, the MAC layer informs the application layer if the QoS requirements can be met and on demand requests the application layer to change the communication parameters which in case of a video transmission could be the frame rate or the screen resolution [8].

The physical transmission is based on a MIMO-OFDM system. Different allocation schemes can be tested: TDMA sends the data of the users consecutively, whereas OFDMA allocates different subcarriers to different users so that each user gets the subcarriers which currently yield the best performance, which provides a simultaneous service of a number of users by means of a suitable power allocation method. A further enhancement is SDMA where each subcarrier can be assigned to more than one user at a time by means of a suitable power allocation scheme as for OFDMA and, in addition, a coding scheme which reduces interference between users. However, this interference limits the maximum capacity of an SDMA system when the number of users increases [9].

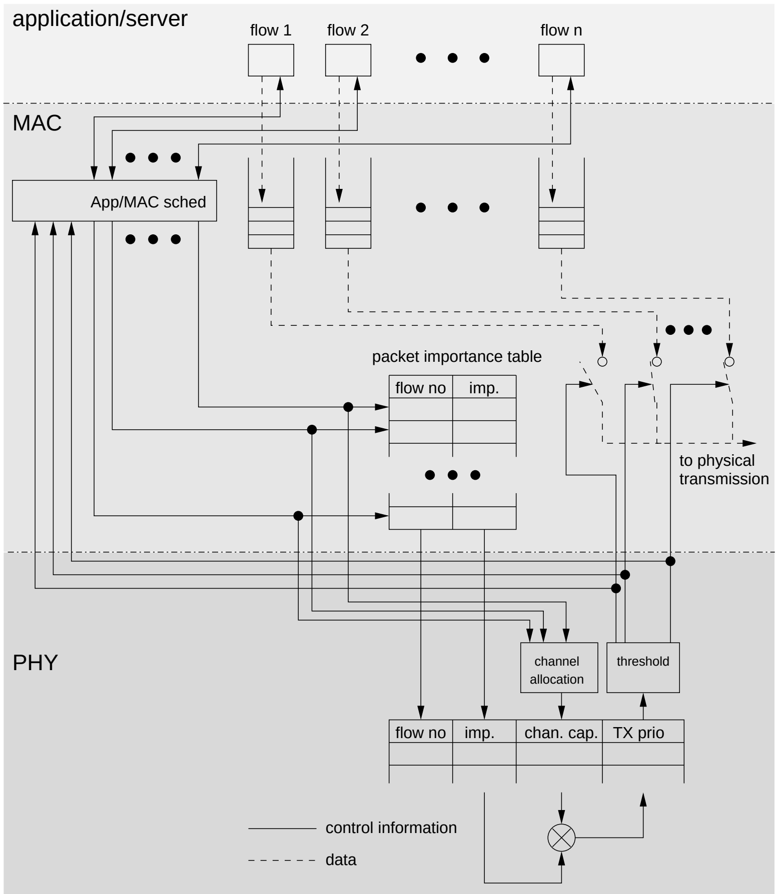

Figure 4. Design of the cross-layer scheduler.

Due to the OFDMA and SDMA resource allocation algorithms, the priorities of the packets at the head of the queue for each data flow are needed so that it is required that the transmissions for all users start at the same time. The packet length and available channel capacity can, however, be variable so that a user with a short transmission time has to wait for other users who take more time to transmit, which results in unused airtime. In order to reduce this problem, packet aggregation is used which means that further packets are taken from the queue and transmitted as long as airtime is available [10].

The beforementioned methods work best if perfect channel knowledge at the transmitter is available, which can be modeled in simulations but is not possible in practical systems. Further investigations have to be performed on the robustness of the schemes in case of imperfect channel knowledge. If the channel capacity is overestimated, this can result in a packet loss, so ARQ needs to be introduced into the MAC. Methods need to be developed which determine how often a transmission should be repeated, which is dependent on the type of application and the delay constraints of the currently transmitted packet. Investigations are also required to reduce the amount of feedback data to report the channel status which is sent from the receiver to the transmitter. Another issue is including adaptive applications into the cross-layer mechanism by defining minimum QoS criteria which have to be met in any case and optimum criteria which should be met if possible. The task of the scheduler is the fair allocation of resources so that each user gets an amount of spectrum which is, as far as the conditions allow, above the required minimum. In addition, connection admission control is needed so that connections whose requirements cannot be met are rejected.

4. POLICY-BASED SPECTRUM MANAGEMENT

With the advent of WMAN as an alternative to cable and DSL in the near future, there will be a possible scenario of an indoor WLAN (aka IEEE 802.11) network connected to the Internet through WMAN (aka IEEE 802.16) backbone network. 802.16 systems operate in licensed and unlicensed bands. However, there are some possibilities where 802.11 and 802.16 systems are candidates for operating in the same unlicensed band (e.g. U-NII band at 5 GHz) in the abovementioned scenario. This scenario can be denoted as “metropolitan apartment scenario”. The spectrum sharing between heterogeneous systems (WLAN and WMAN) hence is an important research topic.

PoSSuM is conceptually an extension of the CoCoNet project, where spectrum sharing among 802.11 WLAN systems has been investigated. PoSSuM aims to develop algorithms for fair spectrum sharing between competing systems like WLANs and WMANs in unlicensed bands. This is an important issue because the radio spectrum is scarce nowadays. Based on the the feasibility study on DARPA XG dynamic spectrum access [6; 12], the challenge of implementing policy-controlled spectrum sharing is an additional objective in this project. One of the main ideas which is being considered is that regular (more predictable) channel occupation by one system in the time domain can support other systems to detect and reliably predict the spectrum opportunity (idle periods). To make the channel occupation more regular, the traffic has to be scheduled in a more deterministic way. An indication of predictable usage of spectrum is discussed in [11]. Moreover, regularity in system behavior helps to establish rules for spectrum sharing.

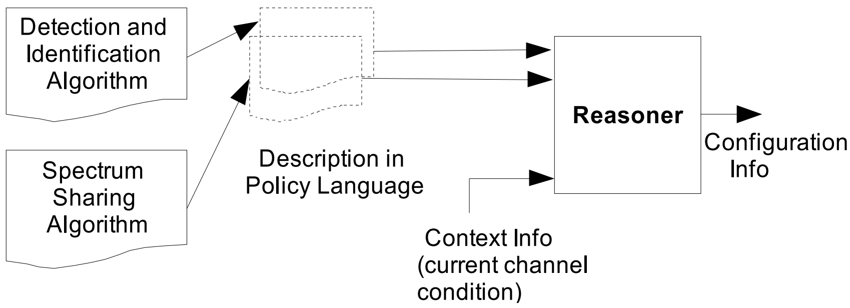

Figure 5. Framework for policy-based spectrum access.

The framework for the policy-based spectrum access is shown in Figure 5. A detection algorithm is required to locate other systems which are operating in the same frequency band. For example, extending an 802.16 station to a policy-based cognitive radio requires detection of 802.11 systems. Detecting some specific patterns by means of correlation might be a solution in this case. An algorithm is required to identify spectrum opportunities or "spectrum holes" which are defined as a frequency channel unused for a certain time period. For example, from the spectrum usage pattern of 802.11 systems, 802.16 systems can identify spectrum opportunities. The above-mentioned algorithms can be developed by analyzing the pattern of measured (noise) data or from their statistics (e.g. idle/busy time histogram etc.). Knowledge gained from identification helps developing spectrum sharing methods or "intelligent medium access". Spectrum sharing supports the radio to select and use identified spectrum opportunities dynamically to improve Quality-of-Service. By implementing above algorithms the radio becomes spectrum agile. This means, it detects when radio resources are sufficient for transmission on the one hand, and ensures that its own interference to the environment meets given constraints on the other hand. These constraints can be specified by policies (rules) which can vary dependent on location and time. Policies which specify a set of rules for spectrum access are required to constrain the dynamic spectrum usage. These rules are interpreted by a reasoner which has two inputs for the policy description provided by above algorithms and the current situation of the channel; as an output it provides the decision about how to allocate which portion of the spectrum and reconfigure the radio to access the spectrum according to the policies. This reasoner is one part of the management plane inside the WLAN protocol stack. The spectrum sharing algorithms in the radio can be soft or hard coded; however, in the latter case the radio will not be flexible enough in case of changing conditions. Detection, identification and spectrum sharing algorithms are required to be described in a common policy language to enable policy-controlled dynamic spectrum sharing. A well-defined policy framework needs to be described by a formal description language to represent policy rules in machine-understandable manner and implement a policy reasoning unit (reasoner). For example, the policy framework provided in the DARPA XG program consists of XGPL (XG Policy Language) and a policy conformance reasoner. The XGPL framework uses OWL (Web Ontology Language) for the machine-understandable representation and a shorthand notation for more human-understandable representation, where the shorthand notation has a one-to-one correspondence to OWL [12].

5. CONCLUSIONS

The three projects CoCoNet, XLayer and PoSSuM which are described in this paper provide important contributions to the enhancement of wireless networks each with a different focus. The aim of CoCoNet is the coexistence between networks which take decisions by observing their environment and drawing conclusions about actions of other networks in order to change their own behavior. In XLayer, the channel access inside a particular network is optimized by a base station which has central control on the channel access and assigns airtime according to QoS requirements of the different stations. Future enhancements will include adaptive applications into the cross-layer mechanism and connection admission control. The focus of PoSSuM is the coexistence between WLAN and WMAN stations. As an enhancement to CoCoNet, the policies for monitoring and accessing the spectrum are no longer hardwired, but defined by a formal description so that they can easily be changed. In future work, the policy-based spectrum access will be used to form meshed networks where WMAN stations provide a backbone and WLAN stations maintain the user access.

ACKNOWLEDGEMENT

This paper gives an overview of the research work performed within several DFG projects. Two of these projects were – and partially are still being – carried out as joint projects between ComNets Aachen and Bremen. This cooperation was established right from the start of ComNets Bremen in 1999. With this contribution the authors want to express their appreciation to Professor Bernhard Walke for his continuous support and many rewarding and enjoyable discussions during these years. We are looking forward to continu-

ing the communication and the networking also within the "Verein der Freunde und Förderer von ComNets" (FFV) in the future.

References

[1] L. Berlemann, "Distributed Quality-of-Service Support in Cognitive Radio Networks," Ph.D. dissertation, RWTH Aachen University, Feb 2006.

[2] S. Mangold, "Analysis of IEEE 802.11e and Application of Game Models for Support of Quality-of-Service in Coexisting Wireless Networks," Ph.D. dissertation, Aachen University, Jun 2003.

[3] A. Könsgen, Z. Hossain, and C. Görg, "Transmit Power Control Algorithms in IEEE 802.11h Based Networks," in *Proc. 16th Annual IEEE Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC)*, Berlin, Germany, 2005.

[4] A. Könsgen and C. Görg, "Performance Evaluation of Dynamic Frequency Selection Strategies in IEEE 802.11h Based Networks," in *Proc. 1st Regional Conference on ICT and E-Paradigms*, Colombo, Sri Lanka, 2004.

[5] G. R. Hiertz, S. Max, L. Stibor, Y. Zang, T. Junge, and D. Denteneer, "Coexistence problem of 802.11s Congestion Control," Melbourne, Victoria, Australia, p. 12, sep 2006, iEEE 802.11 TGs submission.

[6] M. Siddique, A. Könsgen, C. Görg, G. R. Hiertz, and S. Max, "Extending IEEE 802.11 by DARPA XG Spectrum Management: A Feasibility Study," in *Proc. 12th European Wireless Conference*, Athens, Greece, 2006.

[7] L. Berlemann, S. Mangold, G. Hiertz, and B. Walke, "Policy Defined Spectrum Sharing and Medium Access for Cognitive Radios," *Journal of Communications (JCM)*, Apr. 2006.

[8] A. Könsgen, W. Herdt, A. Timm-Giel, and C. Görg, "A Cross-Layer Two-Stage Scheduler for Wireless LANs," in *Proc. 16th IST Mobile & Communication Summit*, Budapest, Hungary, 2007.

[9] A. Könsgen, W. Herdt, H. Wang, A. Timm-Giel, R. Böhnke, and C. Gdüorg, "A Two-Stage QoS Aware Scheduler for Wireless LANs Based on MIMO-OFDMA-SDMA Transmission," in *Proc. 1st IEEE International Workshop on Cross-Layer Design*, Jinan, China, 2007.

[10] A. Könsgen, M. S. Islam, A. Timm-Giel, and C. Görg, "Optimization of a QoS Aware Cross-Layer Scheduler by Packet Aggregation," in *Proc. 10th IFIP International Conference on Mobile and Wireless Communications Networks*, Toulouse, France, 2008.

[11] L. Berlemann, *Distributed Quality-of-Service Support in Cognitive Radio Networks*, ser. Aachener Beiträge zur Mobil- und Telekommunikation (ABMT). Wissenschaftsverlag Mainz, 2006, vol. 50.

[12] "XG Policy Language Framework, Request for Comments," DARPA XG Working Group, Cambridge, Massachusetts, USA, 2004, available on http://www.ir.bbn.com/projects/xmac/rfc/rfc-policylang-1.0.pdf; last visited 2008/05/20.

HSPA EVOLUTION – FIELD EXPERIENCES FROM LIFE NETWORKS

Measuring and Analyzing the High Speed Packet Access Performance

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Abstract With lessons learned from the GSM networks and their enhancements like GPRS and EDGE, UMTS is becoming more and more a high-speed data network competing with wired access techniques. The key enabler is the High Speed Packet Access (HSPA) data transmission with promising data rates up to 14.4 Mbit/s in downlink and 5.7 Mbit/s in uplink direction. This article describes the evolutionary steps in HSPA standardization and deployment. Moreover, with P3 Solutions' long-term experience in life network measurements and analysis, it collects some of the findings while different versions of HSPA have been put into operation.

1. INTRODUCTION

The increasing demand for capacity in order to provide high data rate multimedia services in wireless environments necessitates enhanced radio transmission techniques and network protocol functionality. Such techniques have been integrated into already deployed mobile cellular networks, e.g. the Global System for Mobile (GSM) was upgraded by Enhanced Data Rates for GSM Evolution (EDGE) [1]. For the 3rd generation Universal Mobile Telecommunications System (UMTS) based on Wideband Code Division Multiple Access (WCDMA), the High Speed Packet Access (HSPA) is being introduced to meet this demand and to improve spectral efficiency starting from the year 2005 [2,3]. Like the GSM standard, UMTS is steadily evolving and offers new features mainly for offering increased transmission data rates as illustrated in Figure 1.

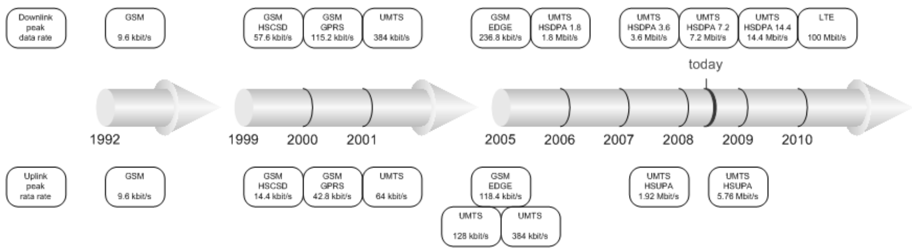

Figure 1. Trend of data rates throughout the years.

These techniques are based on well-known link adaptation by using various Modulation and Coding Schemes (MCS) and with this realizing several different data rates for downlink and uplink transmission. Moreover, fast scheduling mechanisms based on shortened radio frames enable efficient and flexible sharing of the radio resources among different users and services [4]. This article gives a comprehensive overview on the integration of the two components High Speed Downlink Packet Access (HSDPA) and High Speed Uplink Packet Access (HSUPA) in UMTS networks and provides a detailed performance evaluation from life networks collected within the past years since the first introduction in 2005.

The remainder of this chapter is organized as follows. Section 2 highlights the main technical features of HSPA. In Section 3 the measurement methodology and the system design is explained. While Section 4 summarizes the main findings from the step-by-step introduction of HSPA into already operating UMTS networks, Section 5 concludes this contribution and gives a further outlook to the upcoming enhancements within the limits of the UMTS Long Term Evolution (LTE).

2. HSPA TECHNOLOGIES

HSPA caused substantial modifications in the existing protocol architecture, which affects physical (PHY) and Medium Access Control (MAC) layers. Foremost, the PHY layer at the radio air interface has to be enhanced to enable transmission data rates up to 14.4 Mbit/s. This is realized by implementing higher order modulation schemes with fast link adaptation at the HSPA capable User Equipment (UE) and the Node B. Moreover, the minimum Transmission Time Interval (TTI) is reduced to 2 ms, i.e. radio sub-frames, to enable fast adaptation and flexible share of the radio resources. Complementary to the increased PHY capabilities a specialized MAC high speed (MAC-hs) and MAC enhanced (MAC-e) entity with the necessary control functionalities are set-up for HSDPA and HSUPA, respectively [5,6]. As illustrated in Figure 2 these entities provide Hybrid Automatic Repeat Request (HARQ) mechanisms and fast scheduling, facilitating the efficient usage of the radio resources in adaptation to the instantaneous channel conditions and network load. These HSPA enabling techniques are described in detail in the following.

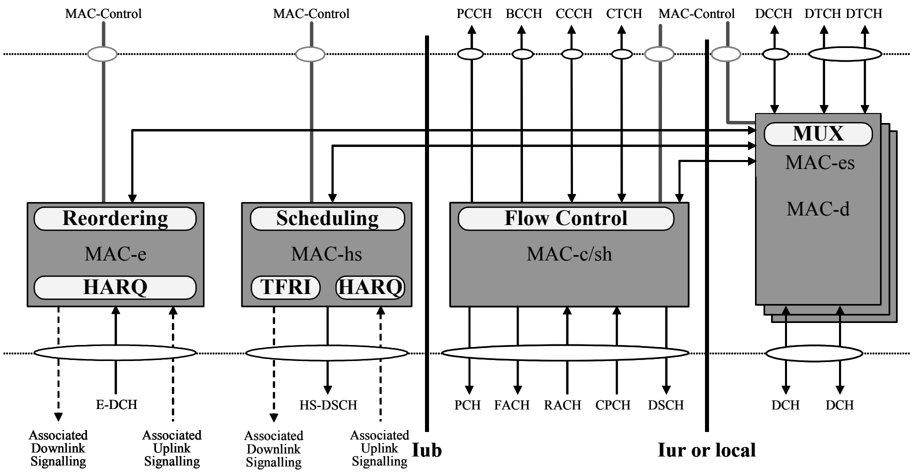

Figure 2. MAC architecture for HSPA (UTRAN side) [6].

Higher order modulation in conjunction with link adaptation is a way of optimising the instantaneous use of the fading radio channel. Besides conventional Quaternary Phase Shift Keying (QPSK) modulation, HSDPA enables the use of 16 Quadrature Amplitude Modulation (QAM) and 64 QAM (within the scope of HSPA+). By transmitting a shared downlink channel at in principle constant power, i.e. without fast power control, the MCS can be selected separately for every UE to maximise throughput while maintaining a certain error ratio. Notwithstanding, higher order MCS are definitely more sensitive towards varying channel conditions as present in high mobility environments. Unlike HSDPA, higher data rates for HSUPA are achieved by less spreading, i.e. a reduced spreading factor of two.

HARQ, combining soft information from additional retransmissions with the original soft information prior to decoding, greatly improves performance and adds robustness against link adaptation errors. It also serves to fine tune the effective code rate and compensates for errors made by the link adaptation mechanism. If all data is correctly decoded, an acknowledgement (ACK) is sent to the Node B, using the associated uplink control channel. But if the data is decoded incorrectly, retransmissions are requested immediately. Once the data has been retransmitted, the UE combines the previous version(s) of data with the retransmitted version. This procedure is called soft combining. Thus, the probability of successful decoding is increased.

Various cell change mechanisms are introduced to exploit macro diversity. The UE indicates the best cell that is currently able to serve data transmission on downlink [5]. Thus, while multiple cells may be members of a "virtually" active set, only the best serving station of them transmits at any time. Hence, downlink interference can be reduced and parallel transmission as well as the need for scheduler synchronisation is avoided.

The scheduler exploits the multi-user diversity and strives to allow transmissions to or from users when radio conditions permit high data rates. Notwithstanding, it also maintains a certain degree of fairness.

2.1 High Speed Downlink Packet Access (HSDPA)

As a key element, HSDPA introduces the High Speed Downlink Shared Channel (HS-DSCH) at a constant Spreading Factor (SF) SF16. HS-DSCH and associated control channels with their signaling functionality are illustrated in Figure 3. The upcoming downlink transmission is indicated via the High Speed Shared Control Channel (HS-SCCH) which carries two parts of information. The first part is specifically encoded with a UE dependent scrambling sequence (encoded Radio Network Temporarily Identifier, RNTI) and contains the modulation and coding information for the following transmission. The second part contains transport format description (block size) and HARQ information. With the first part the user is individually notified about the subsequent transmission of data on the HS-DSCH while the second part describes the decoding procedure necessary to extract the raw data. MCS and transport block size information together are called the Transport Format and Resource Information (TFRI).

In uplink direction, the High Speed Dedicated Physical Control Channel (HS-DPCCH) regularly carries the Channel Quality Indication (CQI) and ACK/NACK information for the HARQ processes. The CQI is dependent on the UE processing capabilities according to the categories listed in Table 1. Category 12 devices as introduced during the first HSDPA launch were able to provide data rates up to 1.8 Mbit/s. Categories 6 and 8 have been available in 2006 and 2007, respectively. Data rates for these devices could be increased to maximum 3.6 Mbit/s and currently 7.2 Mbit/s. HARQ feedback is derived within a data processing interval of approximately 5 ms and combined with the CQI signaling.

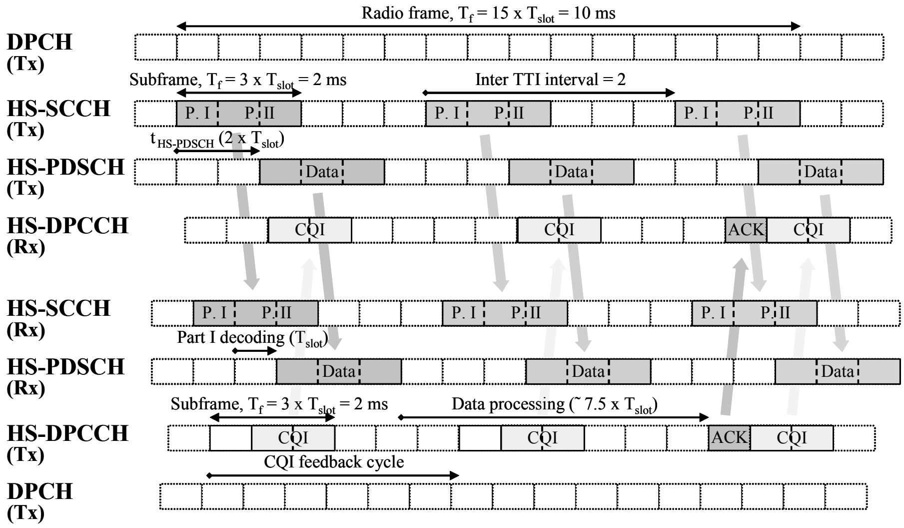

Figure 3. HSDPA channels at UTRAN (upper part) and UE side and their timing relations.

The actual CQI values as defined in [7] have similar meanings across all UE categories. Only at a certain limitation no further increase in data rates is offered. Instead, the HS-DSCH transmission power is to be reduced in steps of 1 dB with increasing CQI. Thus, for e.g. UE category 8, CQI 25 offers the highest data rates of 7.2 Mbit/s whereas the five higher values up to 30 lead to a reduction of transmission power by maximum 5 dB while maintaining a constant MAC throughput (see Figure 4). Category 10 devices without any power reduction will enable physical data rates of up to 12.8 Mbit/s.

Table 1. UE Capabilities for HS-DSCH (HSDPA) [8].

HS-DSCH Category	Max. number of codes (SF 16)	Min. inter-TTI interval	Max. number of bits within a TTI	Total number of soft channel bits	Max. channel bit rate [Mbit/s]
1	5	3	7298	19200	1.2
2	5	3	7298	28800	1.2
3	5	2	7298	28800	1.8
4	5	2	7298	38400	1.8
5	5	1	7298	57600	3.6
6	**5**	**1**	**7298**	**67200**	**3.6**
7	10	1	14411	115200	7.2
8	**10**	**1**	**14411**	**134400**	**7.2**
9	15	1	20251	172800	10.2
10	15	1	27952	172800	14.4
11	5	2	3630	14400	0.9
12	**5**	**1**	**3630**	**28800**	**1.8**

UEs of categories 11 and 12 support QPSK only.

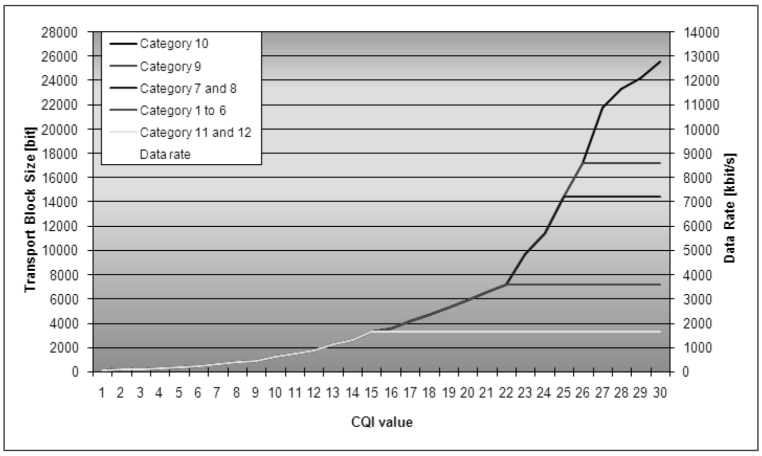

Figure 4. HSDPA instantaneous data rates over CQI for different HS-DSCH categories.

2.2 High Speed Uplink Packet Access (HSUPA)

For HSUPA, the Enhanced Dedicated Channel (E-DCH) is the central element. It is split into E-DCH Dedicated Physical Data Channel (E-DPDCH) and E-DCH Dedicated Physical Control Channel (E-DPCCH). The associated signaling comprises E-DCH Absolute Grant Channel (E-AGCH) and E-DCH Relative Grant Channel (E-RGCH) for scheduling information as well as the E-DCH Hybrid Indicator Channel (E-HICH) for ACK/NACK feedback.

Figure 5 illustrates the HSUPA signaling and data transmission. On the E-AGCH a user-specific Service Grant (SG) defines the maximum allowed E-DPDCH/E-DPCCH power ratio for the E-DCH depending on current uplink noise rise conditions. The UE can then choose from its Transport Format Combination Indicator (TFCI) pool an adequate transmission format. Together with HARQ information and the "happy" bit this is signaled parallel to the E-DPDCH. From the "happy" bit and additional scheduling status information UTRAN can decide to grant additional resources absolutely or relatively. Furthermore, the HARQ cycle is closed via the E-HICH carrying the ACK/NACK messages.

Similar to HSDPA, categories for the UE capabilities are defined which reflect the processing limitations with respect to minimum TTI duration, maximum transport block size and spreading ability, i.e. availability of SF2.

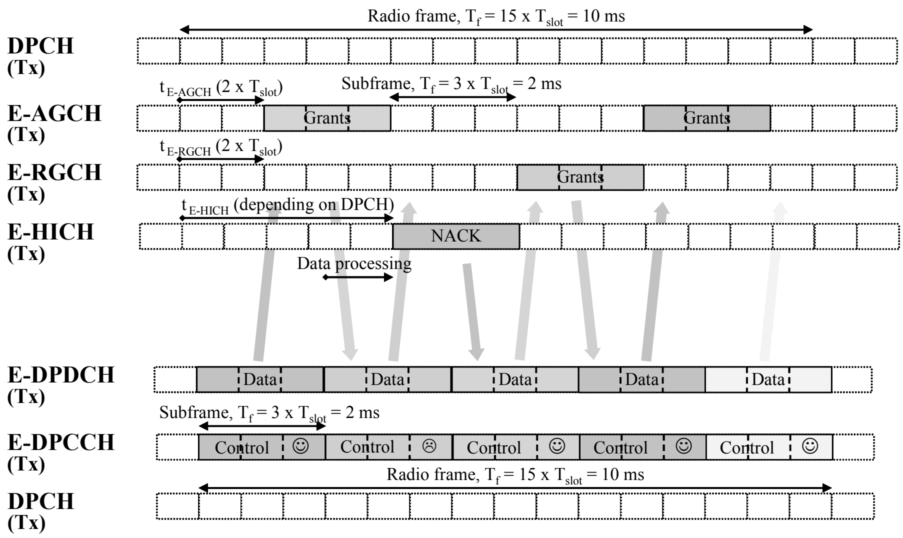

Figure 5. HSUPA channels at UTRAN (upper part) and UE side and their timing relations.

Table 2. UE capabilities for E-DCH (HSUPA) [8]

Category	Max. number of codes	Min. SF	Min. TTI duration [ms]	Max. number of bits in 10 ms TTI	Max. number of bits in 2 ms TTI
1	1	SF4	10	7110	-
2	2	SF4	2	14484	2798
3	2	SF4	10	14484	-
4	2	SF2	2	20000	5772
5	2	SF2	10	20000	-
6	4	SF2	2	20000	11484

When 4 codes are transmitted, two codes shall be transmitted with SF2 and two with SF4.

3. MEASUREMENT SYSTEM AND STRATEGY

P3 Solutions' first comprehensive network test in Germany was realized by auto connect in 2000. The measurements included the networks called at that time D1, D2 Mannesmann, E-Plus and Viag Interkom and were conducted on motorways. Main focus was the evaluation of mobile phone functions (voice service). The result of 500 km driven distance was coverage of at least 84% while using Siemens mobile phones with external antennas.

From 2002 also city tests were included. Narrow streets make coverage difficult; showing only between 61% and 73% of the measured locations providing sufficient coverage in those days, depending on the network operator. Furthermore, customers use their mobile phones in buildings. Therefore, indoor situations are simulated by installing additional attenuators between external antennas and mobile devices at the measurement vehicle resulting in increasing coverage requirements.

Besides voice calls also data transfers are tested. After the GPRS and EDGE packet data standards UMTS has been introduced since 2003 with promisingly high and steadily increasing data rates. With further developments like HSDPA (in 2006) and HSUPA (in 2008) UMTS is becoming a real alternative to fixed access and consequently necessitates steadily ongoing analysis, optimization and performance studies.

According to this steady development, P3 Solutions designed the peeqBOX measurement system for continuous and autonomous measurements in mobile networks. The peeqBOX is a highly integrated and reliable system, which is flexible and adaptable to a requested test scenario.

Figure 6. peeqBOX measurement system.

The HSPA performance tests use a peeqBOX system with a remote access test software which is able to measure packet switched services like FTP or HTTP. For analyzing the network performance multiple scheduled and well-aligned FTP transfers with variable size are configured. All measurements are operated with HS(D)PA data cards. The probes are placed in a single cell for load tests or even across multiple cells for the evaluation of soft and hard handover scenarios. Additionally, a mixed client scenario comprising R’99 and HSDPA is analyzed.

The measurement system records TCP dumps which can be used to evaluate the mean data rate in a defined time interval. The data rate is calculated by summing up the sizes of transferred TCP packets per time interval. Each packet has a maximum size of 1448 byte. The maximum data rate of category 12 UEs is about 1.8 Mbit/s. Hence, a time interval of 0.25 s is small enough to visualize effects like bearer switching or CQI choice.

4. HSDPA PERFORMANCE EVOLUTION

HSDPA performance has been analyzed by P3 Solutions since its first deployment in 2006. Hence, the evolution from lower data rates starting at 1.8 Mbit/s up to 7.2 Mbit/s nowadays as well as the introduction of HSUPA is being monitored. The following sections highlight the results of these studies. Further details of the 2007 German HSDPA campaign performed for the Connect magazine can be found in [9].

4.1 HSDPA 1.8 Mbit/s Performance Trial 2006

With only a single probe in a cell, a peak data rate about 1.5 Mbit/s can be achieved (see upper plot in Figure 7). Due to other data and disadvantageous radio conditions the average data rate is reduced to about 1.2 Mbit/s.

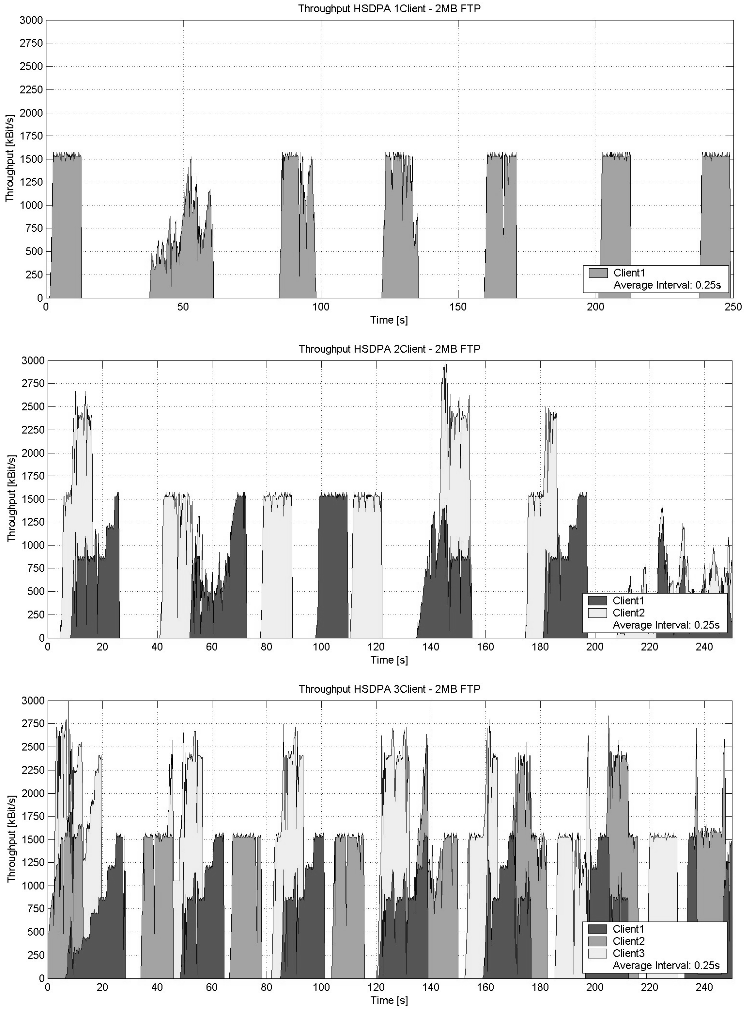

Figure 7. HSDPA throughput (Release 5) with 1, 2, and 3 clients (from top to bottom).

When using two data cards the available bandwidth has to be shared among the users. Caused by the short 2 MByte FTP download time intervals the mean data rate for only each active participant is equal to the measurements with one probe. For two active probes with category 12 data cards the maximum theoretical data rate is about 3.2 Mbit/s which is rarely reached according to the middle plot in Figure 7. The evaluated results show a limitation at about 2.5 Mbit/s. The limit is mainly caused by the fixed link between Node B and backhaul. Usually, a Node B is connected with a 2 x E1 link to the Radio Network Controller (RNC) with a capacity of about

4 Mbit/s including all necessary signaling (about 20%). Hence, the full performance of HSDPA cannot be exploited.

The loading condition with three active probes (lower plot in Figure 7) shows that the mean data rate for each user decreases even though it is just a short time period when all three probes are simultaneously active. The mean data rate for only one client is still at 1.2 Mbit/s. But if there is more than one client active the available capacity is dynamically shared among the different clients. In this load situation the bearer switching is very well visible. The limitation from the fixed network interface can be seen as well.

Furthermore, a mixed scenario with R'99 and HSDPA is analyzed with a maximum of 3 probes per access mode. The evaluation in Figure 8 shows that almost all the time a minimum of one active client exists. The cell limitation at about 2.5 Mbit/s is again visible for this scenario. Data rates for R'99 transmission are mainly limited at 128 kbit/s. But with parallel HSDPA transmission about 1 Mbit/s in average is achieved at the expense of reduced R'99 transmission.

These tests show the general potential of a well-dimensioned UMTS networks. Moreover, they indicate the core network interface capacity as a bottleneck, especially the fixed line connection of the Node B. These interfaces must be adapted to the available radio transmission rates.

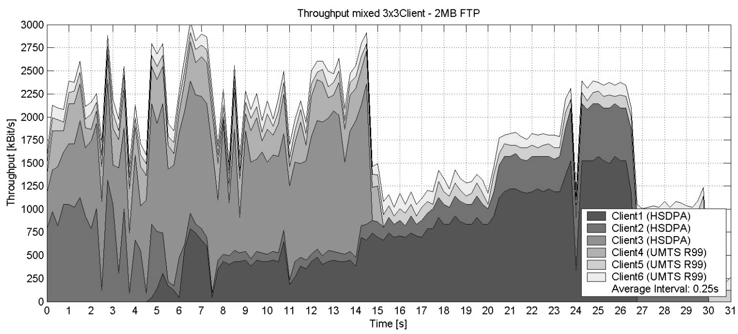

Figure 8. Throughput with 3 x R'99 and 3 x Release 5 probes.

4.2 HSPA Performance Trial 2008

In 2007, many network operators extended their radio access network and core network capacity to be able to provide the increased data rates to the end user. Additionally, newly available techniques are introduced improving the performance of the mobile devices. Receive diversity and advanced receiver technologies are standardized to increase the downlink performance in terms of capacity and coverage. UEs with receive antenna diversity are defined in [10] from Release 6 onwards (see Figure 9).

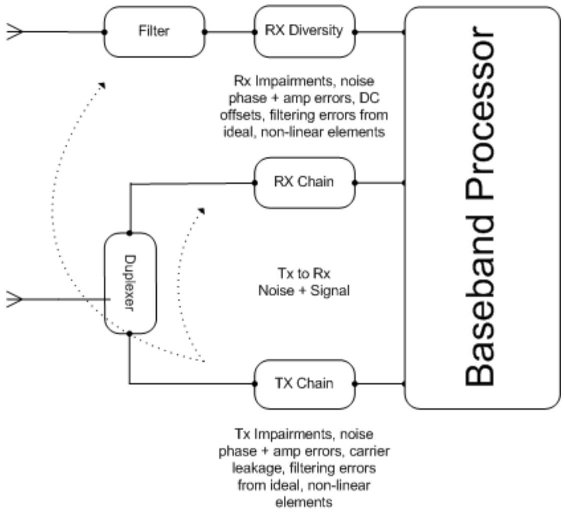

Figure 9. Diagrammatic view of a diversity transceiver.

The Release 6 conform UE employs a rake receiver with receive diversity. A rake receiver is a radio receiver to counter the effects of multipath fading. It does this by using “subreceivers” called fingers. In Release 7, the rake receiver is completed by an equalizer. UEs engaging a rake receiver enhanced with an equalizer are able to achieve twice the performance of UEs with a basic rake receiver.

To take full advantage of diversity technology, the signals received by each antenna should be subject to independent fading. A correlation factor is used to measure the dual-antenna configuration effectiveness. The lower the correlation factor, the more effective is the diversity system at improving the signal to noise ratio and the gain of the combined signal. Low correlations can be achieved by applying:

- spatial separation between antennas,
- different polarization characteristics,
- different radiation patterns.

The measurements in 2008 were intended to work out new strategies and solutions in order to be procurable in benchmarking UMTS Release 7 networks.

For HSDPA, the UE reports information about the channel quality to the Node B every TTI, i.e. 2 ms interval. These CQI values associated with the information from the Node B about buffer level, packet priorities, etc. are deciding for the channel allocation to the UE. In addition to the amount of channels being assigned to the UE, the possible data rate varies. The higher the CQI value, the more downlink channels or uplink power could be allocated to a UE for reasonable usage and the higher the possible data rate. Figure 10 illustrates the actual downlink data rate at MAC layer with respect to the reported CQI. It is obvious that the CQI indicates the upper bound for the data rate whereas the effective data rate is lower due to scheduling restrictions.

Considering only one UE in a cell, an HSDPA peak data rate of about 5.2 Mbit/s could be achieved. Figure 11 illustrates an exemplarily FTP download using an HSDPA 7.2 Mbit/s capable device. Recurrent data transmission or worse radio conditions reduce the data rate to an average of 4.2 Mbit/s.

For HSUPA according to Release 6, a peak uplink data rate of 1472 kbit/s could be achieved. In realistic environments with third party traffic, the data rate is reduced to an average of 1333 kbit/s as illustrated in Figure 12.

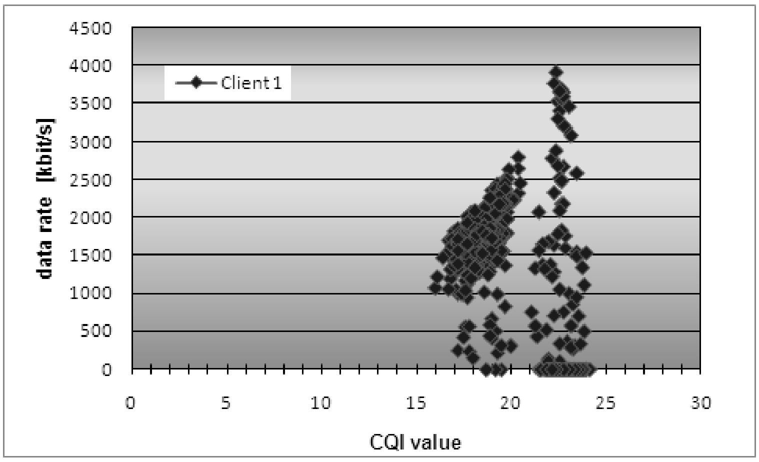

Figure 10. HSDPA data rate to CQI assignment.

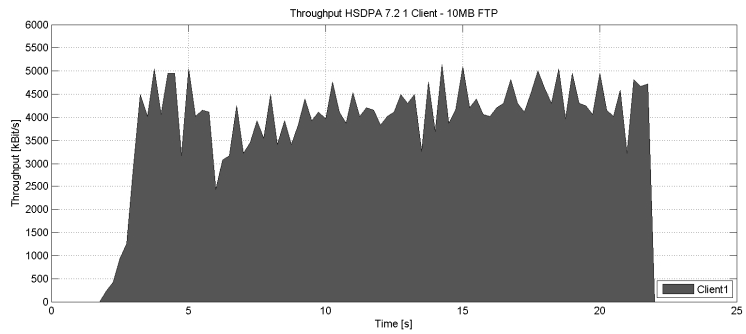

Figure 11. HSDPA throughput (Release 7) during FTP download.

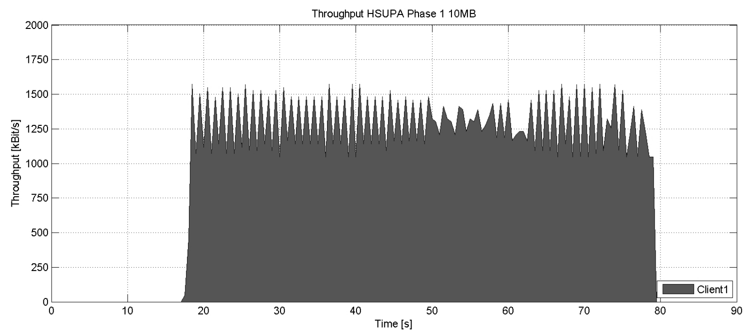

Figure 12. HSUPA throughput (Release 6) during FTP upload.

5. CONCLUSION AND OUTLOOK

HSDPA and HSUPA are promising evolvements for the UMTS radio air interface. Nevertheless, the fixed network and its interfacing capacity have to evolve as well. Since first experiences made during 2006, network operators are more and more expanding their capacity with rich benefits for the end customer. Data rates of 5 Mbit/s in downlink and 1.5 Mbit/s in uplink direction are possible now. This enables UMTS to compete with fixed access while maintaining the user mobility support and wireless freedom.

With further enhancements like HSPA+ and LTE the UMTS is already paving the road towards next generation ubiquitous wireless networking. However, additional techniques like MIMO and relays have to be carefully integrated. P3 Solutions is already aware of new standards and keeps its measurement and analysis systems up to date with emerging standards.

ACKNOWLEDGEMENT

The authors would like to thank their partners from T-Mobile and Vodafone for the support and free access to their networks, even during trial deployments. The inspiration and encouragement of Prof. Walke from ComNets at the RWTH Aachen University of Technology is highly appreciated. Also his personal support for P3 Solutions, the share of ideas and all the motivating discussions during the whole company lifetime are well regarded. Finally, the authors acknowledge the support and advice of Prof. Valenzuela from the Bonn-Rhein-Sieg University of Applied Sciences.

References

[1] P. Stuckmann, The GSM Evolution – Mobile Packet Data Services, Wiley 2003

[2] H. Holma, A. Toskala. HSDPA/HSUPA for UMTS. Wiley 2006

[3] H. Holma, A. Toskala. WCDMA for UMTS – HSPA Evolution and LTE. Wiley 2007

[4] I. Forkel, H. Klenner, A. Kemper, High Speed Downlink Packet Access (HSDPA) – Enhanced Data Rates for UMTS Evolution, in special issue of Computer Networks named Wireless Networks Performance and Protocols, Vol. 49, No. 3, pp. 325-340, The Hague, Netherlands, 2005

[5] 3GPP, TS 25.308: High Speed Downlink Packet Access (HSDPA); Overall description, Stage 2

[6] 3GPP, TS 25.309: FDD enhanced uplink; Overall description; Stage 2

[7] 3GPP, TS 25.214: Physical layer procedures (FDD)

[8] 3GPP, TS 25.306: E Radio Access capabilities

[9] W. Boos, Daten Wettlauf, in Connect Magazine, pp.12-17, June 2007

[10] 3GPP, TS 34.121: Terminal conformance specification, Radio transmission and reception (FDD)

Further Related Standard Documents

- 3GPP, TS 25.211: Physical channels and mapping of transport channels onto physical channels (FDD)
- 3GPP, TS 25.212: Multiplexing and channel coding (FDD)
- 3GPP, TS 25.213: Spreading and modulation (FDD)
- 3GPP, TS 25.215: Physical layer - Measurements (FDD)
- 3GPP, TS 25.301: Radio interface protocol architecture
- 3GPP, TS 25.321: Medium Access Control (MAC) protocol specification
- 3GPP, TS 25.331: Radio Resource Control (RRC) protocol specification

- 3GPP, TR 25.848: Physical layer aspects of UTRA High Speed Downlink Packet Access
- 3GPP, TR 25.855: High Speed Downlink Packet Access; Overall UTRAN Description
- 3GPP, TR 25.858: High Speed Downlink Packet Access: Physical Layer Aspects
- 3GPP, TR 25.899: High Speed Download Packet Access (HSDPA) enhancements
- 3GPP, TR 25.950: UTRA High Speed Downlink Packet Access

ComNets & PHILIPS - PARTNERS IN STANDARDIZATION

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ComNets, the Department of Communication Networks at RWTH Aachen University and Philips have a long history of cooperation. In 1996, the first joint research projects began. Aiming at state of the art technology, the near-market-solutions oriented collaborations always had a remarkably deep industry impact. Until the end of the last national research program (Wireless Gigabit With Advanced Multimedia Support, WIGWAM), twelve ComNets members (Matthias Lott, Andreas Hettich, Jörg Peetz, Jörg Habetha, Stefan Mangold, Georgios Orfanos, Guido R. Hiertz, Yungpeng Zang, Lothar Stibor, Jelena Mircovic, and Sebastian Max) supported Philips in various fields of tomorrow's communications systems.

From the beginning, Philips and ComNets employees introduced the results of the project jointly to different standardization bodies. In the standardization efforts self-organization of home networks and multihop networking was always a key element. The initial efforts focused on the HiperLAN/2 standardization within ETSI, where ComNets researchers acted as editors and chairmen of the home extension of the standard. Many of the HiperLAN/2 concepts have later been integrated into the IEEE 802.11 WLAN standard. Since 2004 we also participated in IEEE's Wireless Personal Area Network (WPAN) group (IEEE 802.15) and its spin-off "Multi-band OFDM Alliance", which later changed its name into WiMedia. From the very beginning of their existence, we supported the standardization groups IEEE 802.11s (WLAN Mesh) and IEEE 802.15.5 (WPAN Mesh) with submissions, proposals and active participation – thus making ComNets truly "Mesh veterans". With further extensions of our joint activities, ComNets also represented Philips in IEEE 802.11p (Wireless Access in Vehicular Environment) and IEEE 802.11n (High Throughput MIMO PHY).

1. IEEE STANDARDS & IEEE PROJECT 802

IEEE project 802 is one of 25 standards committees under the umbrella of the IEEE Standards Association (SA). Project 802 – also known as the LAN/MAN standards committee (LMSC) – is formed and sponsored by the IEEE Computer Society (Figure 1). The LMSC came into existence in February 1980 when development on the first IEEE standard for Local Area Networks (LANs) began. Since then, IEEE 802 has mainly focused on the lowest two layers – PHY and MAC. Already in 1981, IEEE 802 had three Working Groups (WGs) that considered Carrier Sense Multiple Access with Collision Detection (CSMA/CD), the Token Bus and Ring concepts for LAN applications. Each WG is responsible for the development of standards in a certain technological area. A WG and its standard are marked by an extension number that follows the project number "802.". Besides WGs, Technical Advisor Groups (TAGs) also receive a project number. A TAG's expertise lays in the development of recommended practices that deal with subjects related to multiple WGs.

Being the oldest WG in the LMSC, 802.1 is concerned with the overall IEEE 802 system concept and interoperability between the family of 802 standards. While the "Higher Layer LAN Protocols" WG is still active, WG 802.2 – "Logical Link Control" – is currently in hibernation. On top of 802's grass-roots, Ethernet (802.3) became the dominating LAN standard worldwide. Although already in 1981 other LAN WGs were formed in the IEEE 802, none of them are active anymore. 802.4 (Token Bus) has been disbanded and 802.5 (Token Ring) currently remains in hibernation. Because of our active involvement in IEEE 802.11 and its outstanding importance, in the following we focus on the Wireless LAN WG and exemplify the way through standardization by means of IEEE 802.11s – WLAN Mesh.

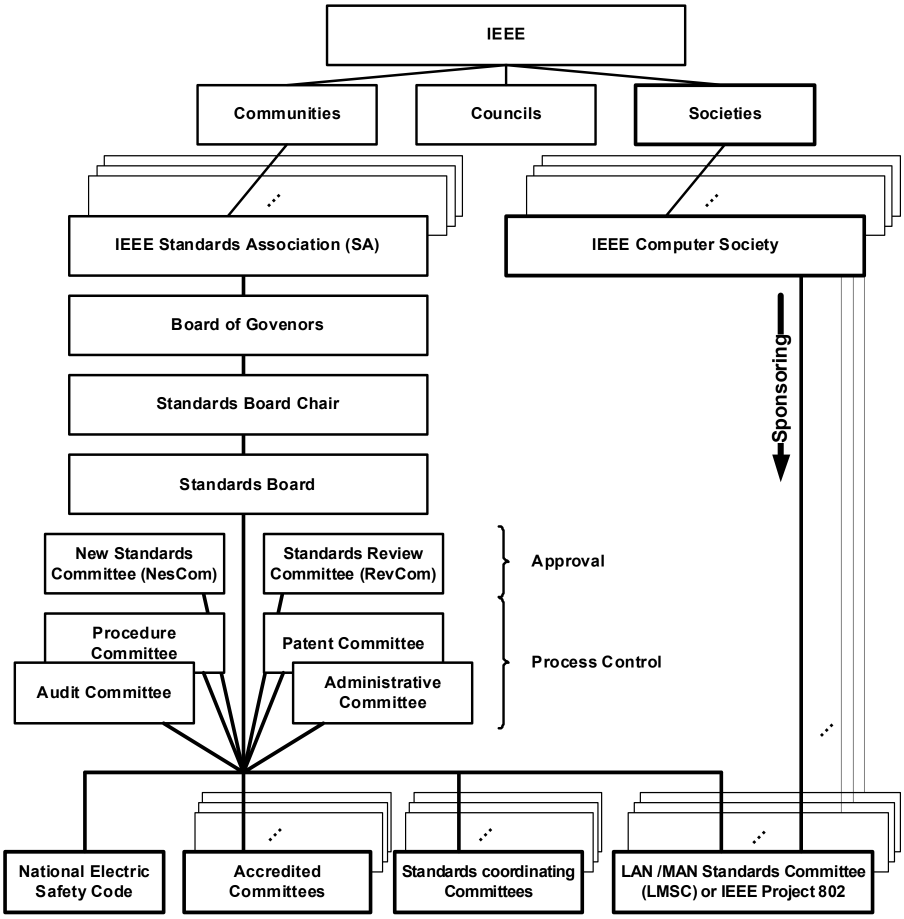

Figure 1. IEEE Standards Association and the IEEE Computer Society lead IEEE project 802.

2. IEEE 802.11s – STANDARD FOR WLAN MESH

As shown in Figure 2, amendments to the 802.11 standard are born in the Standing Committee (SC) Wireless Next Generation (WNG). The SC has a special status as it provides an open forum for new ideas, proposals, and a general platform for discussion on the 802.11 standard. Having socialized an idea and successfully convinced a majority of SC WNG attendees about a new topic for standard development, the 802.11 WG may decide to initiate a

Study Group (SG.) The SG's task is the development of two documents: the Project Authorization Request (PAR) describes the goals, the need for, and the conceptual outline of an amendment. The Five Criteria (5C) document discusses an amendment's market potential, compatibility, distinct identity, technical, and economic feasibility. Once both documents have been adopted, on behalf of the SG the WG may forward them to the New Standards Committee for approval of a new Task Group (TG). Having become the Mesh SG in January 2004, it took only two more IEEE 802.11 meetings until TG "s" was approved. From May 2004 on, the amendment for Mesh networking was known as 802.11s.

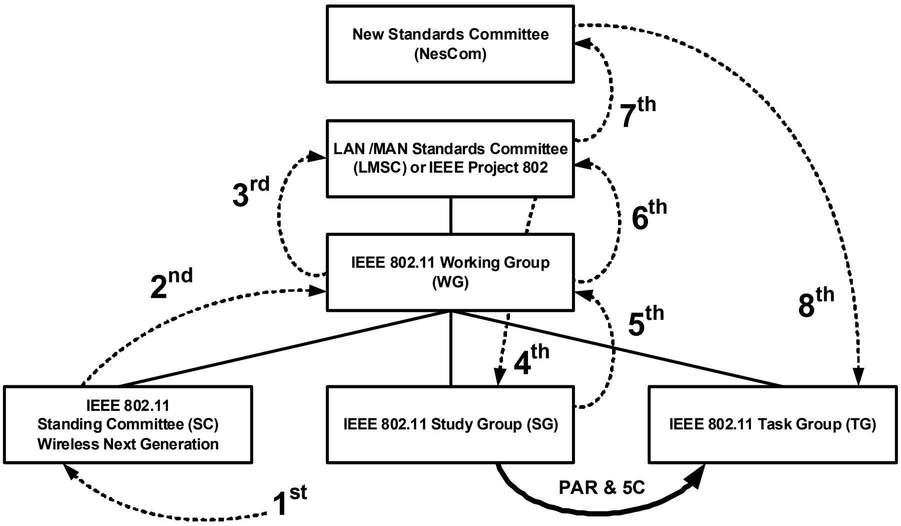

Figure 2. The Standing Committee "Wireless Next Generation" is an incubator for new groups in IEEE 802.11.

Throughout the following months, TGs developed a usage scenario document that describes the foreseen applications to be supported by 802.11s. Accompanied by a down-selection-process related document, 802.11s issued a Call for Proposals (CFP) in January 2005. After 35 notices of intent to submit a proposal were received, fifteen proposals were actually presented during the July 2005 TGs meeting. According to its down-selection procedures, three proposals were eliminated from the list as they received less than 25% affirmative votes. With each ballot in the September and November meeting, further proposals were eliminated or proponents merged their concepts with groups that were still under consideration. The joint ComNets-Philips proposal dropped out on third position in November

2005. Having left so many other companies and consortia behind us, was certainly a success. Especially the detailed and realistic simulative survey of our advanced proposal convinced many attendees until finally company interests forced them to deny further support for our ideas. Since then, ComNets and Philips had officially joint the Wi-Mesh Alliance (WiMA), an industry consortium consisting of Nortel, InterDigital, Mitre, Accton, Thomson, Swisscom and NextHop.

2.1 Down-Selection Process & Letter Ballots

Since the down-selection procedure requires a majority of 75% votes for forming a baseline document that can be further worked on by TGs, the remaining two entities – WiMA and the industry forum SEE-Mesh – merged their proposals in January 2006 (Figure 3). After four more IEEE meetings, TGs found its baseline document in a shape sufficient to be forwarded as the 93rd letter ballot of IEEE 802.11 in November 2006. During a letter ballot, a TG seeks for approval of its draft standard by the WG. Thus, all voting members of the 802.11 WG are requested to submit a motion of approval or disapproval. In case of the latter, a voting member must submit comments that explain the disapproval.

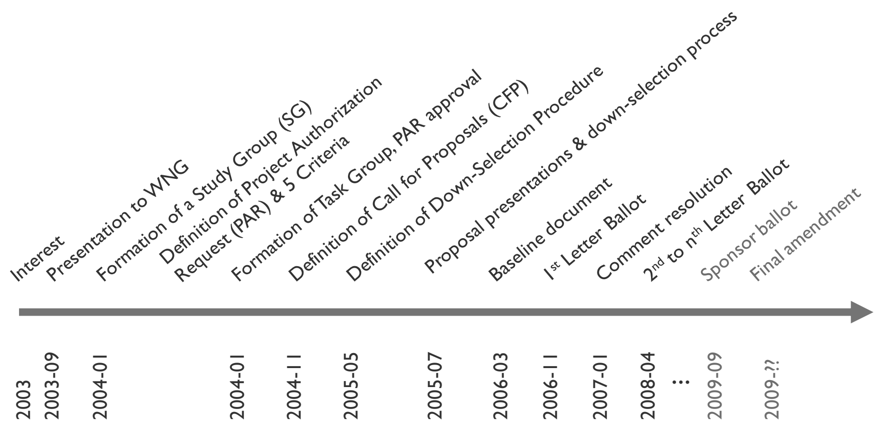

Figure 3. ComNets & Philips contribute to IEEE 802.11s since 2003.

With more than 5,700 comments, TGs received the second highest amount of comments that were ever submitted during a letter ballot. Until today, only the TG of 802.11n received more comments. With such a burden, it took TGs until March 2008 to resolve all comments. The resolution of comments (acceptance, rejection or counter proposal) usually is required before a TG can submit its draft for another letter ballot. After 34 IEEE meetings, six 802.11s ad hoc meetings, nine WiMA and WiMA/SEE-Mesh meetings, the WLAN Mesh standard has still a long way to go, see Figure 4. Once the WG approves the draft, the standard must be confirmed during the so called sponsor ballot. This time, within the sponsor ballot members of the IEEE SA may submit comments. Once the document has passed this final hurdle, a new standard is born. Although the current 802.11s standard is in a pre-mature stage, implementations already exist. The One Laptop per Child initiative certainly is the most prominent implementer of 802.11s. Current laptops rely on an early version of the initial draft. Another open source implementation is available in the open80211s.org project. The latter relies on the WLAN stack mac80211 of the Linux kernel. Future revisions of open80211s will enable any Linux compliant WLAN card with mesh capability.

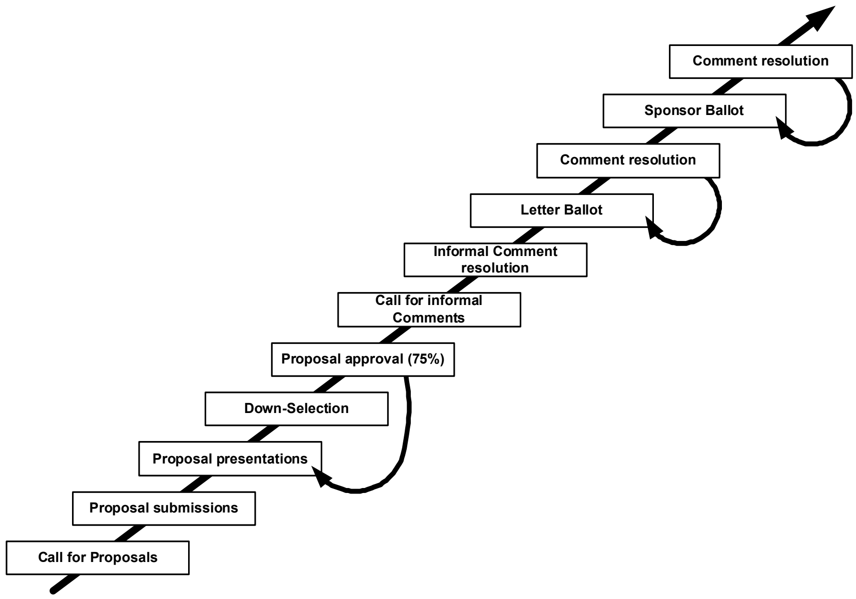

Figure 4. Several steps must be accomplished until a standard's amendment is approved.

2.2 The Business Side – Wi-Fi & WiMedia Alliance.

Since standardization solely targets specification and technology, independent standardization related industry alliances exist that develop certification and interoperability programs. For Ultra-Wideband communication (UWB), the WiMedia Alliance specifies the ecosystem. In the past, ComNets has represented Philips at WiMedia meetings and successfully contributed to major parts of the WiMedia MAC that first became ECMA and then ISO standards. Due to our continued efforts on mesh networking, members of ComNets also began representing Philips in the Wi-Fi Alliance (WFA). Since October 2006, ComNets members attended all of the three annual WFA meetings in addition to six annual IEEE meetings. While IEEE 802 grants voting rights to an individual, voting rights in WFA are per company and ComNets votes on behalf of Philips. In its current state, WFA's Mesh Marketing group develops the marketing and certification strategies for mesh networking. Special attention is given to the integration of such a certification program into the existing family of Wi-Fi brands and marketing strategies. Several questions remain that concern non-mesh products and their integration into the new device categorization, the degree of backwards compatibility to existing networks as well as the efficient and secure auto-configuration of mesh products in the home.

3. CONCLUSIONS

The joint activities in standardization have proven to be very fruitful. Philips and ComNets have made key contributions to the standards HiperLAN/2, IEEE 802.11e, IEEE 802.11n (consortium member of winning proposal), IEEE 802.11p, IEEE 802.11s, IEEE 802.15, and WiMedia (winning proposal). While the industry receives excellent research insides and expertise, ComNets' involvement in standardization provides itself with deep insight to standards, active commitments, influence and access to latest documents. Many of ComNets' publications benefit from this informational advantage. While such broad contribution to invention, proposal submission, marketing and the definition of specific market requirements allows ComNets a very unique and deep inside view to newest technology, participation would not be feasible without Philips' support. Even without participation in industry alliances like WFA, a commitment that includes attendance of regular and non-regular meetings can easily exceed ten trips to international locations per year – a cost factor that ComNets could not afford on its own.

LTE ADVANCED SELF-BACKHAULING

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Ericsson Research

Abstract Multi-hop communication claims to offer cost-efficient coverage extension and capacity increase. Hence it has been a hot research topic over decades and multi-hop functionality has been integrated in all kinds of wireless systems ranging from prototypes (WINNER), over local area networks (IEEE 802.11s) to metropolitan area networks (IEEE 802.16j).

However, apart from simple repeaters, multi-hop communication has not yet been part of commercial cellular networks from the 3GPP or 3GPP2 technology families. But that situation might change. The standardization group 3GPP is discussing multi-hop as one technical component of Long-Term Evolution (LTE) Advanced, the latest evolution of its cellular systems. This paper shows how a multi-hop enabled LTE Advanced system might look like. First, it introduces the current LTE system. Second, the paper outlines the different multi-hop options that are being discussed and it finally details on self-backhauling, a multi-hop solution for cost-efficient backhauling of base stations.

1. INTRODUCTION

Cellular networks are foreseen to cover diverse geographic regions. On the one hand they shall cover urban areas with a high density of buildings and indoor usage, on the other hand cellular networks shall provide access over large geographic regions in remote rural areas. In both scenarios it is challenging to cover the entire service area. Either huge parts are heavily shadowed from the base station or the link distances are very large so that radio propagation characteristics are challenging.

In order to cope with diverse radio propagation conditions, multi-hop communication has been proposed. By means of intermediate nodes, e.g., relays, the radio link is splitted into two or more hops each with better propagation conditions than the direct link. That enhances link quality leading to throughput and coverage enhancements [8]. Due to the enhanced coverage and the increased cell capacity, multi-hop communication promises a cost-efficient deployment and service [15].

Several multi-hop concepts have been developed so far for local and metropolitan area networks. The ETSI-BRAN HiperLAN/2 standard contains a direct mode that allows terminals to communicate directly [9]. Another concept, leverages the HiperLAN/2 option for sector antennas to allow for multi-hop communication [19]. A third proposal for HiperLAN/2 realizes multi-hop communication by allocating periods within the Medium Access Control (MAC) frame to relays, where relays act as base stations [7].

The IEEE 802.16 specification contained an optional mesh mode where traffic could be routed directly between terminals [11]. Due to lack of economic interest the mesh mode has been removed from the standard after four years [13]. The IEEE 802.16 Task Group IEEE 802.16j currently specifies an enhancement for a mobile multi-hop operation without modifying the terminal. Unlike the mesh mode, the task group aims at a tree-based radio network deployment [10, 12]. In parallel, Task Group IEEE 802.16m seeks to develop an advanced IEEE

802.16 air interface that meets the requirements for next generation mobile networks (named IMT-Advanced). That amendment contains a relay option that is not compatible with the one specified in 802.16j. In order to meet back-ward compatibility for legacy equipment, including 802.16j, legacy frames are time-and/or frequency multiplexed into the 802.16m frame structure [16].

The European research project Wireless World Initiative New Radio (WINNER) investigated layer 2 relaying as a key technology that can be applied to enhance coverage and capacity of a base station [17]. Performance evaluation showed the potential to improve coverage, especially to overcome uplink power limitations. However, capacity improvement in terms of spectral efficiency could only be realized by intelligent and dynamic resource partitioning and reuse schemes. An additional cost evaluation showed that the introduction of relays has a positive impact on capital and operational expenditures provided that the cost of a relay is less than approximately 1/3 that of a base station [18].

Up to now, multi-hop communication has not been integrated in any of the relevant cellular systems of the 3GPP (GSM/EDGE, UMTS/HSPA, LTE) or 3GPP2 technology family (IS95, CDMA2000, EV-DO). However, 3GPP members foresee multi-hop transmission as part of their advanced concepts intended for IMT-Advanced systems [3].

The remainder of this paper is structured as follows. Section 2 introduces the architecture and protocols of 3GPP LTE and technology components of LTE Advanced. Section 3 outlines alternatives of forwarding on different protocol layers and it motivates the choice of forwarding on layer 3. Sections 4 and 5 discuss self-backhauled eNodeBs, especially eNodeB requirements, forwarding options, and a potential protocol architecture. The paper concludes with a summary.

2. 3GPP LONG-TERM EVOLUTION

The 3GPP standardization body (3rd Generation Partnership Project) is currently working on the specification of the evolved 3rd Generation mobile system, where the core network (Evolved Packet Core (EPC)) related evolution of the architecture is often referred to as System Architecture Evolution (SAE) while the radio access evolution (Evolved UTRAN (E-UTRAN)) is referred to as Long-Term Evolution (LTE). Hence the overall system is named SAE/LTE. For an overall description of the LTE part of the architecture see [1], for the SAE part see [2] and for both see [4, 5].

2.1 Logical Architecture

EPC and E-UTRAN are evolving in parallel. As shown in Figure 1 the resulting flat architecture is composed of only two nodes in the User Plane (UP): the eNodeB and the Serving Gateway (S-GW). The S-GW executes generic packet processing functions similar to router functions, including packet filtering and classification and it provides the connection to the outside world.

Like in the UP, only two nodes are involved in the Control Plane (CP): the eNodeB and the Mobility Management Entity (MME). The MME handles core network control functions, such as attach/detach handling, mobility functions, bearer management, and security . It terminates the Non-Access Stratum

(NAS) signaling protocols with the terminal (named User Equipment (UE) in 3GPP terminology). Separation of MME and S-GW functionality facilitates optimized network deployments and enables fully flexible capacity scaling.

eNodeBs are connected to the core network using the IP-based EPC to E-UTRAN interface named S1. The logical interface between eNodeBs, i.e., the IP-based X2 interface, supports loss-less mobility and multi-cell Radio Resource Management (RRM). The flat architecture reduces the number of involved nodes and thus optimizes network performance, improves cost-efficiency and facilitates the uptake of mass-market IP-based services.

Existing 3GPP and 3GPP2 systems are integrated to the evolved system through standardized interfaces. Such integration supports both dual and single radio handover, allowing for flexible migration to LTE.

2.2 LTE Physical Layer

LTE can be used in both paired Frequency Division Duplex (FDD) and unpaired Time Division Duplex (TDD) spectrum. With FDD, downlink and uplink traffic is transmitted simultaneously in separate frequency bands.

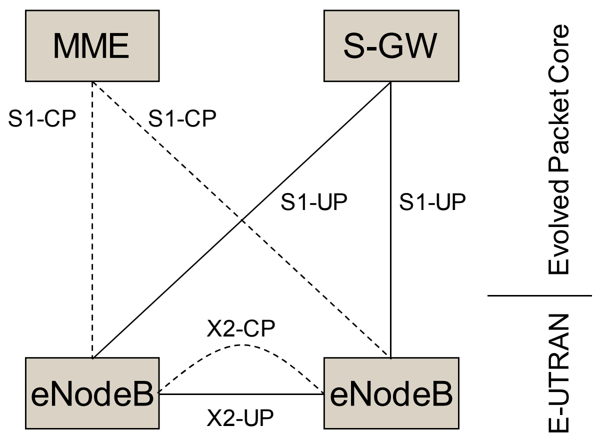

Figure 1. Logical architecture of the System Architecture Evolution.

With TDD the transmission in uplink and downlink is discontinuous within the same frequency band. Nearly all cellular systems use FDD today hence it represents higher device and infrastructure volumes. Furthermore it has some advantages with respect to coverage. TDD is a good complement in unpaired TDD spectrum and it allows advanced antennas to leverage channel reciprocity. Because LTE hardware is the same for FDD and TDD (except for filters), TDD and FDD operators will for the first time be able to enjoy the same economies of scale.

LTE supports flexible carrier bandwidths from below 5 MHz up to 20 MHz allowing operators to introduce LTE on various spectrum bands. LTE can be operated in new bands offering large carrier bandwidth but also in existing cellular bands with smaller carrier bandwidth. Furthermore, flexible carrier bandwidth allows for an initial deployment on small carriers, which will be extended to larger bandwidth as customer demands grow.

Advanced multi antenna solutions are the key component of LTE in order to meet next-generation mobile broadband network requirements for high peak data rates, extended coverage and high capacity. Different advanced multi antenna techniques addresses different scenario. For instance, high peak data rates can be achieved in high Signal to Interference plus Noise Ratio (SINR) regions with multi-layer transmissions (aka spatial multiplexing) while diversity coding or beamforming improves coverage and capacity in low SINR regions.

The overall time domain structure of LTE consists of radio frames of length 10 ms. The radio frame is subdivided into subframes (Transmission Time Intervals (TTIs)) of length 1 ms. A TTI is further subdivided into two slots of length 0.5 ms.

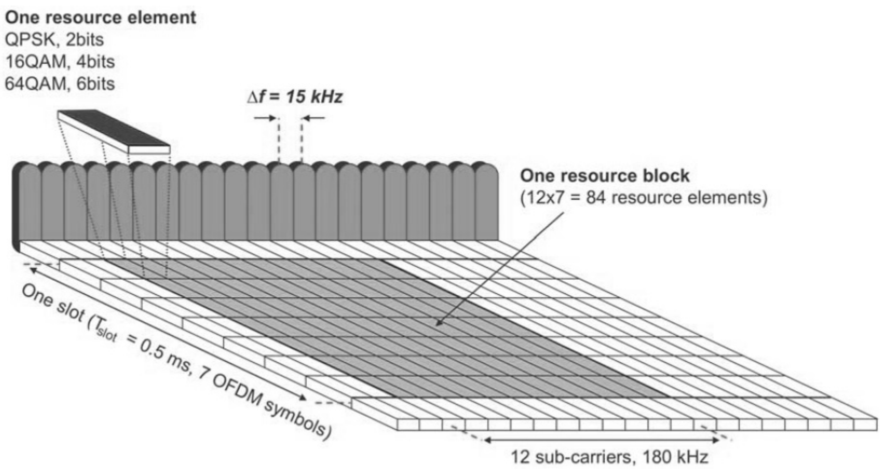

Figure 2. LTE downlink physical resources [5].

Downlink: LTE uses Orthogonal Frequency Division Multiple Access (OFDMA) for the downlink, i.e., from the eNodeB to the UE. OFDMA allows for a flexible spectrum allocation and it enables cost-efficient solutions for very wide carriers.

OFDMA divides the broad carrier bandwidth in a large number of narrow subcarriers that are modulated individually (QPSK, 16QAM or 64QAM). The superposition of all modulated subcarriers forms an OFDM symbol. Subsequent symbols constitute the resource granularity in time while subcarriers represent the granularity in the frequency domain. Thus, the basic LTE downlink physical resource can be seen as a time-frequency grid composed of resource elements as shown in Figure 2. That grid allows for a channel-dependent allocation of physical resources in time and frequency.

In the frequency domain, the regular spacing between subcarriers (1/Δf), is 15 kHz. The OFDM symbol duration time is 1/Δf + *cyclic prefix*. The cyclic prefix is used to maintain orthogonally between subcarriers even for a timedispersive radio channel. Using the regular cyclic prefix, 7 OFDM symbols constitute one slot and 14 symbols constitute one TTI of the overall time domain structure.

In downlink, the first and the sixth TTI of a frame, which is composed of 10 TTIs, contain the broadcast channel carrying system information and the synchronization channel used by UEs for gathering and maintaining time and frequency synchronization.

Uplink: Normal OFDMA transmission causes a high Peak to Average Power Ratio (PAPR) requiring expensive and inefficient Power Amplifiers (PAs). That increases the cost of the UEs and drains the battery faster. Inefficient PAs also reduce coverage and cell-edge performance. In order to overcome the drawback, in uplink, LTE uses a pre-coded version of OFDMA called Single-Carrier Frequency Division Multiple Access (SC-FDMA) which reduces PAPR and hence UE costs. SC-FDMA groups together allocated resources and pre-codes them with a Discrete Fourier Transform. The resulting transmission has Single Carrier (SC)-like characteristics including low PAPR.

2.3 LTE Link Layer

In LTE, the link layer is composed of the protocols Medium Access Control (MAC), Radio Link Control (RLC), and Packet Data Convergence Protocol (PDCP). Above resides the network layer, which uses the IP protocol in LTE. Figure 3 shows the LTE protocol stack which terminates in the UE on the one hand and in the eNodeB on the other. More details can be found in [4].

Packet Data Convergence Protocol: As the interface towards the IP protocol operating in the layer above, PDCP performs two basic functions. Header compression reduces the size of the IP/TCP/UDP/ headers and thus the number of bits transmitted over the radio interface. The compression mechanism is based on Robust Header Compression [14]. PDCP protects the transmitted data by means of ciphering. At the receiver side, PDCP performs the corresponding deciphering and decompression operations.

Radio Link Control: RLC is responsible for segmentation and/or concatenation of RLC Service Data Units (SDUs) into RLC Protocol Data Units (PDUs). The size of the RLC PDUs is controlled by the MAC scheduler.

The receiving RLC entity performs the reverse concatenation and/or segmentation operations.

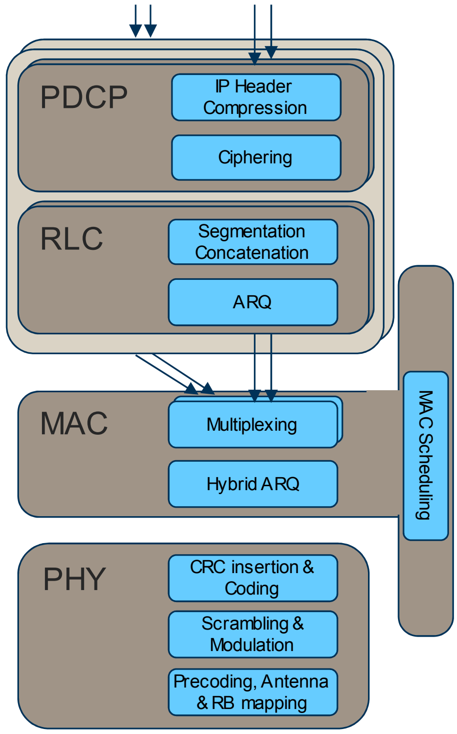

Figure 3. LTE protocol architecture.

Furthermore, RLC provides an error-free, in-sequence delivery of data. The Automatic Repeat Request (ARQ) mechanism handles retransmissions of erroneously received PDUs based on the sequence number. It removes duplicates and re-orders RLC PDUs if necessary.

Medium Access Control: The MAC layer performs multiplexing, Hybrid ARQ (HARQ), and scheduling. The scheduler entity of an eNodeB allocates uplink and downlink resources while the entity in the UE just acts according to the assigned grants. In the time domain, resources are allocated per TTI and in the frequency domain the smallest unit is a resource block composed of 12 subcarriers. The time-frequency grid offered by the physical layer together with appropriate channel measurements allows for channel-dependent scheduling. Besides requesting payload of a certain size from the RLC protocol, MAC scheduling controls HARQ retransmissions and it adapts modulation schemes and coding rates down in the physical layer.

HARQ with soft combining (chase combining or incremental redundancy) provides robustness against transmission errors and thus enhances capacity. The MAC protocol multiplexes different logical channels (per UE) and maps them to transport channels.

2.4 LTE Advanced

In 2008 3GPP started to discuss the evolution of LTE/SAE, named LTE Advanced. LTE Advanced targets to meet the requirements of the International Telecommunications Union (ITU) for next generation mobile systems, named IMT-Advanced. Several technology components have been discussed at the 3GPP workshop [3, 6]:

Wider bandwidth: The widest carrier bandwidth specified for LTE is 20 MHz. The extension to wider bandwidths is part of the LTE evolution towards LTE Advanced since spectrum allocated might have carriers larger than 20 MHz.

Considering spectrum compatibility with legacy LTE, carrier aggregation is the natural choice. Such approach should support combinations of aggregated carriers with various bandwidths.

Spectrum aggregation: Spectrum aggregation is seen as a generalization of carrier aggregation such that non-adjacent carriers are aggregated there. This technique allows leveraging multiple carriers, such as carriers already allocated, carriers newly allocated and even refarming of spectrum used for other technologies. These bands are, in general, scattered in the frequency domain.

Multi antennas: Multi antenna techniques are a key component of LTE and they are further enhanced in LTE Advanced. Spatial multiplexing increases the peak spectral efficiency. In uplink, spatial multiplexing with up to four simultaneous data streams is a strong candidate for LTE Advanced. In downlink, higher-order spatial multiplexing is considered as well. Since beamforming can increase cell edge user throughput, a combined beamforming / spatial multiplexing approach is recommended to be introduced, too.

Coordinated multipoint transmission: Coordinated multipoint transmission is a technology where transmission and/or reception is coordinated across several geographically separated points. In uplink the support for joint processing of signals received at multiple geographically separated points is relatively straight forward and the impact on the LTE radio interface specification is expected to be marginal. In downlink, coordinated multipoint transmission can have different flavors. It ranges from dynamically coordinated scheduling across separate points to joint transmissions from multiple geographically separate points. The impact on the radio interface specification depends on the selected approach, but is expected to be higher than in uplink.

Multihop functionality: Multihop functionality is claimed to offer cost-efficient coverage extension and/or capacity increase [15]. So the introduction of relaying functionality as part of LTE Advanced is being discussed. Different approaches, e.g., Layer 1 repeater, Layer 2 relay, and Layer 3 self-backhauling will be outlined in section 3. As main topic of this paper, self-backhauling is presented in detail in sections 4 and 5.

3. WIRELESS MULTI-HOP COMMUNICATION

3.1 Layer 1 Repeater

A repeater receives a certain signal, amplifies it and transmits it again (aka amplify and forward). Forwarding is performed on Layer 1, i.e., the physical layer. A repeater typically introduces very little delay compared to other multihop solutions operating on higher layers. At the same time, a repeater can not differentiate between received desired signals and received noise and interference since no decoding operation is performed in the repeater. Both noise and desired signal are amplified and forwarded and therefore the repeater can not improve the SINR from input to output.

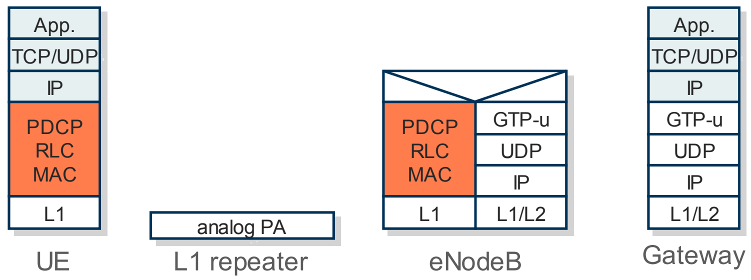

Figure 4. LTE user plane protocol stack of a layer 1 repeater solution.

A repeater can either be frequency translating, where the repeated signal is transmitted on a different carrier frequency relative to the received signal, or on-frequency operating, where received and transmitted signal are on the same carrier. In case of an on-frequency repeater, the repeated signal and any direct signal will add like channel multipath in the receiver. On-frequency repeaters typically need some form of self-interference cancelation functionality.

By example Figure 4 shows the LTE UP protocol stack. On the network side, radio protocols, such as PHY, MAC, RLC, and PDCP terminate in the eNodeB while higher layer protocols, such as TCP and IP terminate in the corresponding application server (here for simplicity the gateway). Conceptually a simple repeater can be thought of as an analog Power Amplifier (PA). However, a repeater could also be more advanced and, e.g., contain a controllable bank of band-pass filters, perform measurements, transmit reference signals etc. An even more advanced repeater could consist of several receive and transmit antennas enabling multi-stream signal repetition.

There are two main target areas of repeaters, firstly, cost efficient coverage extension of uncovered areas and secondly, to increase capacity and achievable data rates in badly covered areas.

3.2 Layer 2 Relay

A relay forwards user and control plane traffic on Layer 2. As the relay node decodes, re-encodes, and forwards received data blocks a delay is introduced. However, no noise is forwarded by the relay node and rate adaptation may be performed individually for each link. The direct and the relayed signal interfere so that relay and eNodeB transmissions have to be separated, e.g., by time or frequency multiplexing.

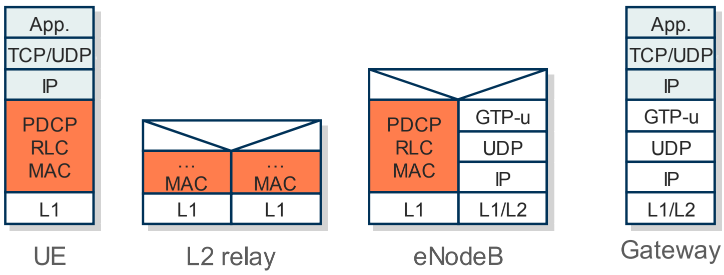

Figure 5. LTE user plane protocol stack of a layer 2 relay solution.

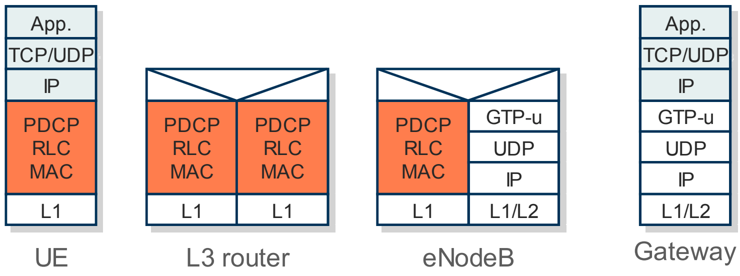

Figure 6. LTE user plane protocol stack of a layer 3 wireless router solution.

Many different technology options exist for a layer 2 relay solution. Layer 2 protocols, such as MAC, RLC, PDCP can operate either end-to-end or on a per-hop basis, see Figure 5. Control functionality, such as resource allocation and system broadcast, can be performed centrally by the eNodeB or it can be distributed amongst relays.

3.3 Layer 3 Wireless Router

A wireless router forwards IP packets on Layer 3, i.e., the network layer. A wireless router has similar capabilities and characteristics as a Layer 2 relay, such that it does not amplify noise and interference but that it introduces processing delays. In contrast to a layer 2 relay, radio protocols (layer 1 and 2) are not affected, see Figure 6.

Such a wireless router provides advantages of decode & forward relaying without requiring new network nodes or modified radio protocols. The characteristics of a wireless router motivate its usage for self-backhauled eNodeBs as it will be discussed in the following sections.

4. SELF-BACKHAULING eNodeB

The network infrastructure that is used to connect eNodeBs to the core network is an IP-based transport network, which can comprise of different Layer 1 / Layer 2 technologies, e.g., leased telephone lines, fibre optic cables, Ethernet or microwave links. The type of transport network and Layer 2 technologies employed is a deployment issue, depending on the availability, cost, ownership, operator preferences, etc., of such networks in the particular deployment scenario. However, the costs of the transport network often play a significant part of the overall operation costs of the network.

This is the reason why LTE Advanced discusses possible solutions to use the LTE radio interface as a backhaul link to connect an eNodeB (named Self-backhauled eNodeB (sNB)) via another eNodeB (named Anchor eNodeB (aNB)) to the core network. This method is called self-backhauling. The purpose of self-backhauling is to reduce the cost for the transport network and in turn to provide cost-efficient backhauling.

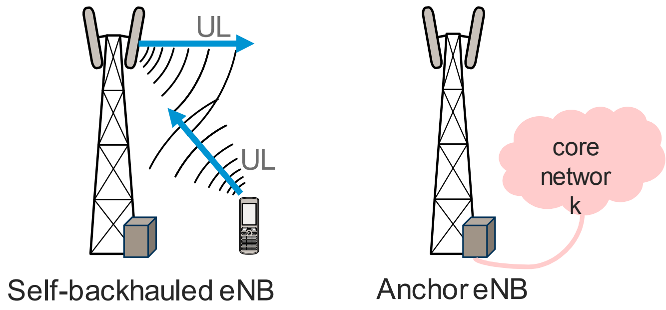

Figure 7. Transmissions on access and backhaul link in UL band.

4.1 Transmission Requirements

LTE uses different transmission schemes for Downlink (DL) and Uplink (UL), see section 2.2. In general, an eNodeB transmits DL signals using OFDMA and it receives UL signals using SC-FDMA. An sNB whose back-haul traffic is served like "regular UE traffic" by the aNB requires additional functionality: such an sNB additionally transmits SC-FDMA UL signals, see Figure 7 and it receives OFDMA DL signals, see Figure 8. Consequently, an sNB requires UE-like transceiver capabilities.

4.2 In-Band Self-Backhauling

In the concept of in-band self-backhauling physical resources are (dynamically) shared between self-backhauling and UE traffic, i.e. backhauling is performed inside the regular spectrum band. Since an sNB transmits and receives in the same band its transmitted signal interferes with the received signal, see Figure 8. So called self-interference occurs at the sNB in DL as well as in UL. Note that the aNB does not generate self-interference.

At the sNB, the receive power of its own transit signal is orders of magnitude higher than the receive power of the desired UE signal so that the

SINR is reduced drastically. Without additional effort the desired signal cannot be decoded successfully. The following sections outline potential technical means to mitigate self-interference.

Coordinated transmission and reception: Self-interference at the sNB can be mitigated by time multiplexing the backhaul into the sNB's access link. Since the sNB's access link is controlled by the sNB and the backhaul link is controlled by the aNB, scheduling of both eNodeBs has to be coordinated. Figure 9 shows the involved DL transmissions on the access links (aNB-UE and sNB-UE) as well as the DL transmissions on the backhaul link (aNB-sNB).

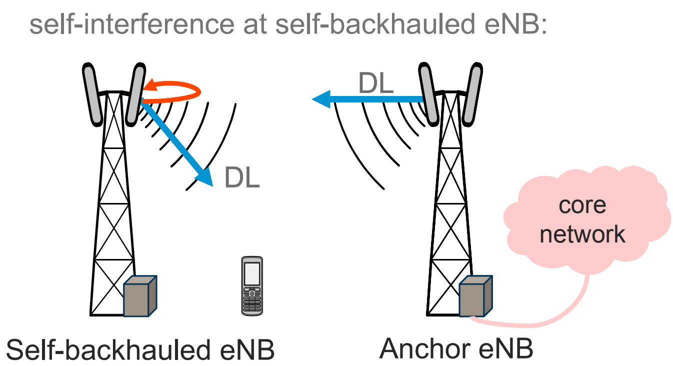

Figure 8. Self-interference at a self-backhauled eNodeB.

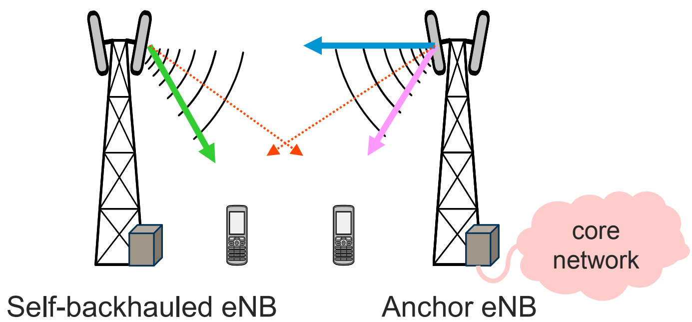

Figure 9. Desired DL transmissions (solid line) and interfering signals (dotted lines) of access and backhaul link.

Figure 10 shows 4 TTIs of the aNB's DL schedule (top) and the corresponding DL schedule of the sNB (bottom). In the aNB, resource allocations for the backhaul link (marked with horizontal lines) are embedded into allocations for regular UEs (marked with vertical lines). All transmissions are scheduled on orthogonal resources by the aNB scheduler.

In order to mitigate self-interference, the sNB's DL schedule is subdivided into subsequent transmit (Tx) and receive (Rx) phases. During Tx phases, e.g., 1st and 3rd TTI, the sNB allocates resources to its own UEs, i.e., the sNB transmits user data. In the 2nd and 4th TTI the sNB switches to Rx so that it can receive transmissions on the backhaul link. During TTIs dedicated to Rx, no UE can be served by the sNB. An analog scheme needs to be applied in the UL.

Rx phases need to be negotiated on a long-term basis so that the aNB can allocate resources and the sNB can switch to Rx mode. They should occur periodically. The length of Rx (and Tx) phases can be one or more TTIs. Within the negotiated Rx phase, the actual resource can be allocated (channeldependent) anywhere in the time-frequency grid. The mechanism to negotiate Rx phases seems to be quite similar to the Discontinuous Reception (DRX) operation of LTE, a UE power saving feature.

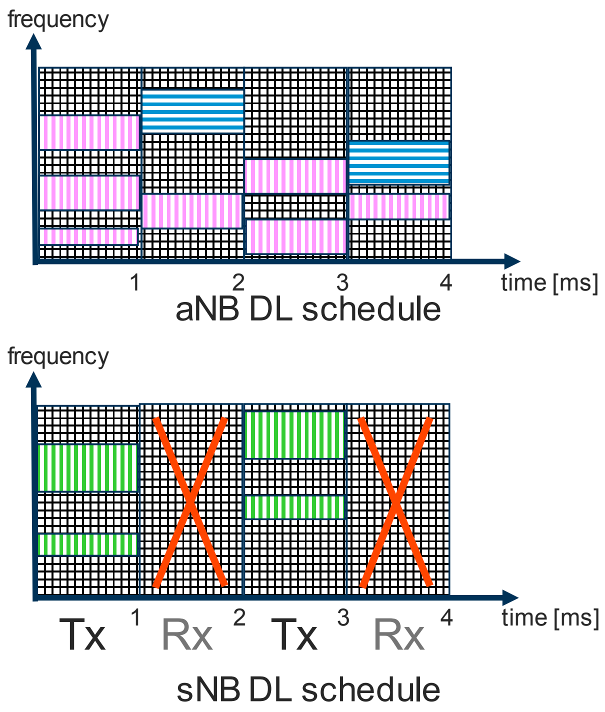

Figure 10. Example resource allocation of aNB (top) and sNB (bottom) using coordinated Tx and Rx phases.

From aNB perspective the backhaul transmission consumes exactly the amount of physical resources allocated to the transmission. From sNB perspective, each backhaul transmission consumes an entire TTI. Long intervals between TTIs with backhaul transmissions lead to large delays, short intervals increase the overhead at the sNB.

Dedicated antenna for self-backhauling: Self-interference can be reduced by deploying a separated antenna dedicated to the backhaul link. Such an antenna could be highly directive, e.g. a parabolic antenna. It could be pointed directly towards the peer eNodeB without sacrificing the downtilt of the eNodeB antenna used for the access link.

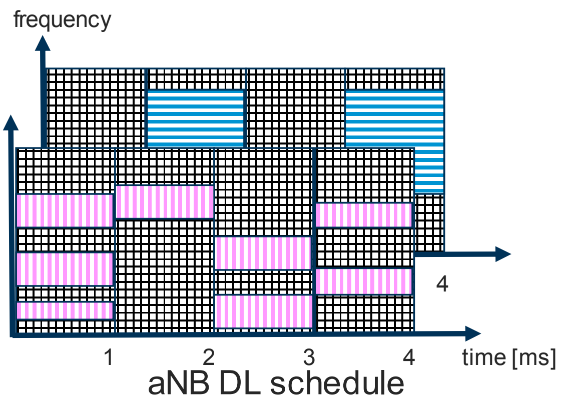

Figure 11. Example resource allocation of an aNB using a dedicated antenna for self-backhauling.

At the aNB a dedicated antenna allows to allocate backhaul resources independently from UE resources. The dedicated antenna could be seen as a separate sector that serves only one UE, which is the sNB. Since physical resources are no longer shared between backhaul and user traffic, UE performance is not affected by self-backhauling.

If the separation of backhaul and access antenna at the sNB is sufficient to suppress self-interference, access and backhaul are independent at the sNB as well. No Tx and Rx phases are required and the sNB can continuously serve UEs while doing self-backhauling. If the antenna separation at the sNB is not sufficient, the sNB needs to coordinate transmission and reception by negotiating Rx phases as shown above.

The drawback of an extra antenna is the extra cost for equipment and deployment, the advantage is a higher capacity for both access and backhaul link.

4.3 Out-Band Self-Backhauling

In contrast to in-band self-backhauling where access and backhaul link share the same spectrum band, self-backhauling could also be performed out-band, i.e., access and backhaul link operate on separate spectrum bands.

At the aNB the out-band solution leads to independent resource allocation of both links. Figure 12 shows the corresponding DL schedules of an aNB where resources for the access link (marked with horizontal lines) are allocated on a different carrier than resources of the backhaul link (marked with vertical lines). Those two carriers can either be adjacent or non-adjacent.

With separate carriers, access and backhaul link do not interfere each other at the sNB. The sNB can continuously serve its UEs on one carrier while performing self-backhauling on the other carrier. Self-interference does not occur.

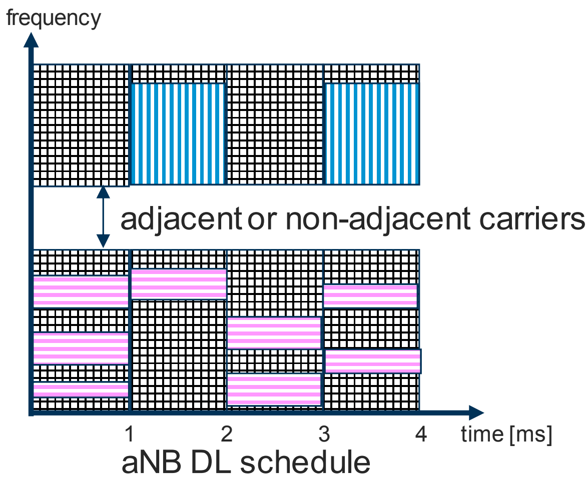

Figure 12. Example resource allocation of an aNB using a out-band self-backhauling.

The drawback of the out-band solution is the extra transceiver costs, the advantage is a better performance of access and backhaul link. Out-band self-backhauling could use spectrum bands which are high up in the radio spectrum (frequencies above 3 GHz) and which are therefore not very useful for Non Line-of-Sight transmission. However, such bands could be used for self-backhauling under Line-of-Sight propagation conditions. Self-backhauling could even utilize unused TDD spectrum. To do so the self-backhauling would be based on LTE TDD while the access would remain FDD.

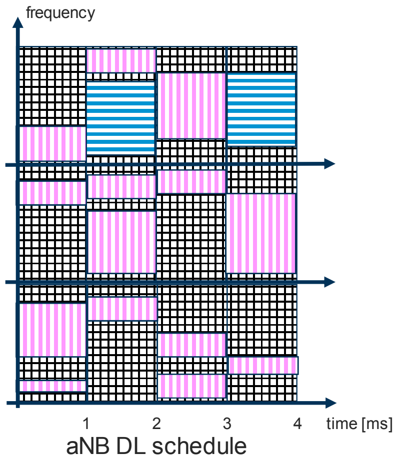

Figure 13. Example resource allocation of an aNB aggregating three carriers.

Out-band self-backhauling and spectrum aggregation. Spectrum (and carrier) aggregation functionality is proposed as potential technical component of LTE Advanced, refer to section 2.4. Such functionality would allow for a more spectrally efficient performance of out-band self-backhauling. Figure 13 shows an example DL resource allocation of an aNB that aggregates three carriers. In the lower two carriers the aNB allocates resources for UE traffic only (vertical lines). On the third subcarrier the aNB allocates resources for backhaul traffic (horizontal lines) as well.

The sNB uses the carriers differently. The lowest carrier is used for the access link only, i.e., the sNB transmits user traffic. The highest carrier is the backhaul carrier, there the sNB receives backhaul data from the aNB. The middle carrier serves as guard band to avoid self-interference, it is not used by the sNB.

With carrier aggregation functionality, the aNB can efficiently utilize resources from all three carriers. Beside the restriction of allocating backhaul traffic to certain carriers only, resources can be dynamically shared between access and backhaul.

5. PROTOCOL ARCHITECTURE FOR SELF-BACKHAULING

5.1 User Plane

Figure 14 shows the user plane protocol stack including the E-UTRAN and the S1 interface of a conventional, i.e., non-self-backhauled system. The radio access uses the protocols as described in section 2.2.

The user plane part of the S1 interface is based on the GPRS Tunneling Protocol (GTP) protocol, which uses a tunneling mechanism ensuring that IP packets destined to a given UE are delivered to the eNodeB where the UE is currently located. GTP encapsulates the original IP packet into an outer IP packet which is addressed to the proper eNodeB. The S1 interface can be operated over various Layer 1 / Layer 2 technologies, e.g., fiber optic cables, leased (copper) lines, or microwave links.

An example TCP/IP based application, such as web browsing, is shown in Figure 14. The corresponding peer entities operate in the UE and at the server hosting the web application. For simplicity, peer protocol entities of the server are drawn in the S-GW, however, in general they are located somewhere in the Internet.

One potential approach to integrate an sNB into the user plane protocol architecture is shown in Figure 15. There, the aNB is seen as part of the transport network, acting like a wireless IP router in between the sNB and the core network. In principle IP packets addressed to the sNB are routed via the aNB based on the sNB IP address. Regular IP routing mechanisms could be used.

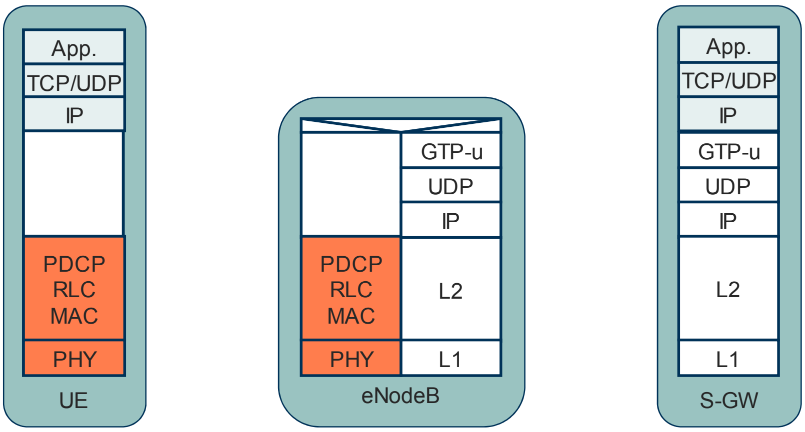

Figure 14. User plane protocol stack (E-UTRAN and S1 interface).

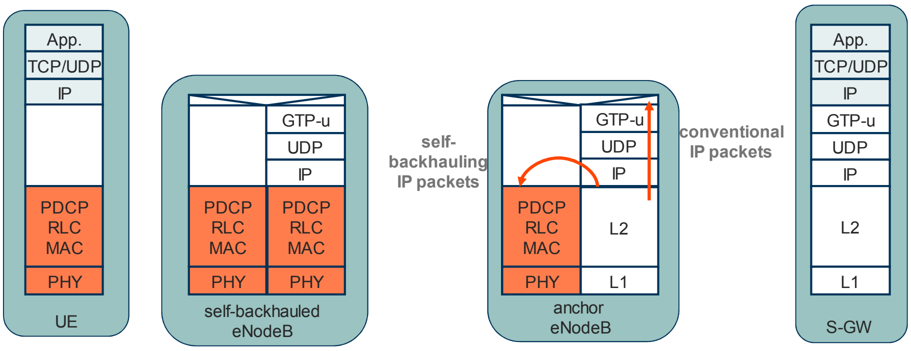

Figure 15. Potential user plane protocol stack for self-backhauled eNodeBs.

That approach imposes that the aNB has to handle (conventional) IP packets destined for a UE associated to that aNB differently than (self-backhauling) IP packets destined for the sNB (actually destined for UEs associated to the sNB). The former IP packets are decapsulated and plain user IP packets are transmitted over the air. The mapping of IP packets to radio bearers at the aNB is done based on the GTP tunnel endpoint. The latter IP packets are directly routed towards the sNB resulting in GTP-encapsulated IP packets being transmitted over the air. Here the mapping to radio bearers cannot rely on GTP tunnel endpoints but, e.g., on IP Quality of Service (QoS) mechanisms such as IP Differentiated Services. As a result the sNB cannot be controlled like a regular UE by the MME. A modified QoS management would have to be introduced.

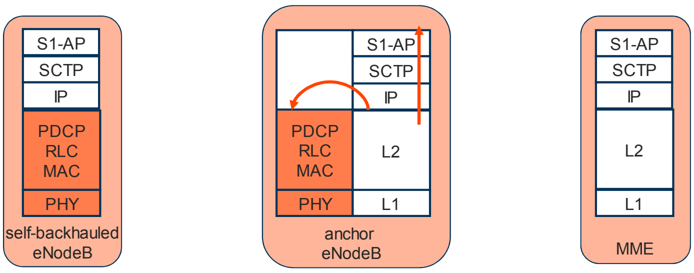

Figure 16. Potential control plane protocol stack for self-backhauled eNodeBs.

5.2 Control Plane

The control plane of the S1 interface connects the eNodeB and the MME. The corresponding protocol is called S1-AP and operates on top of the Stream Control Transmission Protocol (SCTP). The MME uses the S1-AP protocol to establish, configure and tear down bearers, to authenticate UEs, to control ciphering of NAS signaling, and to support UE mobility.

The approach to introduce self-backhauling in the CP could look similar to the one on the UP: IP packets carrying CP data for an sNB are routed via the aNB, which is acting like an IP router in the transport network. IP packets carrying CP data for the aNB are decapsulated at the aNB. Regular routing protocols can be applied. The corresponding control plane protocol stack is depicted in Figure 16.

Although packet forwarding and routing for the CP would be similar to the UP, the CP traffic should be mapped to bearers with higher QoS. The MME acting as the serving MME for the sNB will have to configure the radio bearers and set the packet classification rules in the aNB such that CP traffic is mapped on high priority radio bearers. This needs to be configured when the S1 interface between sNB and MME is established, i.e., at sNB setup/configuration.

6. SUMMARY AND CONCLUSION

This paper shows how self-backhauled eNodeBs might introduce multi-hop functionality into 3GPP LTE Advanced, the future commercial cellular system. After outlining the different options of forwarding on Layer 1, 2 or 3, the paper presents a detailed description of self-backhauling, a cost-efficient solution for backhauling of eNodeBs.

Self-backhauling provides major benefits of conventional layer 2 relaying such as signal regeneration due to decode and forward operation, and per-link optimization of the radio transmission. Furthermore self-backhauling avoids major disadvantages of layer 2 relays: it does not introduce new network nodes and it requires only little standardization effort. These characteristics make self-backhauling a promising technology component.

ACKNOWLEDGMENTS

With his continuous research on cutting-edge mobile communication, Prof. Dr.-Ing. Walke has strongly influenced the development of mobile telecommunication systems over decades. Supported by the staff of his

Department of Communication Networks (ComNets) he developed and promoted multi-hop communication as a key component of next generation systems. Finally, it might become a promising technical component of cellular 3GPP networks.

During his professorship at RWTH Aachen University, Prof. Walke established a strong connection to Ericsson and especially to Ericsson Eurolab's research department. In a series of bilateral and publicly funded research projects Ericsson and ComNets cooperated fruitfully. In particular, this cooperation led to numerous employments of well educated ComNets graduates at Ericsson Eurolab. Thank you and congratulation for your excellent work!

References

[1] 3GPP (2006). 3GPP TS 36.300, Technical Specification Group Radio Access Network; E-UTRA and E-UTRAN; Overall description.

[2] 3GPP (2008a). 3GPP TS 23.401, Technical Specification Group Services and System Aspects; GPRS enhancements for E-UTRAN access.

[3] 3GPP (2008b). REV-080060, Report of 3GPP TSG RAN IMT-Advanced Workshop.

[4] Dahlman, Erik, Parkvall, Stefan, Skold, Johan, and Beming, Per (2007). *3G Evolution: HSPA and LTE for Mobile Broadband*. Academic Press.

[5] Ericsson AB (2007). Long Term Evolution (LTE): an introduction. White paper.

[6] Ericsson AB (2008). R1-082024, A discussion on some technology components for LTE-Advanced. Contribution to TSG-RAN WG1 #53.

[7] Esseling, N., Vandra, H., and Walke, B. (2000). A forwarding concept for hiperlan/2. In *Proceedings of European Wireless 2000*, volume 0, pages 13–18.

[8] Esseling, N., Walke, B., and Pabst, R. (2005). *Emerging Location Aware Broadband Wireless Adhoc Networks*, chapter Fixed Relays For Next Generation Wireless Systems, pages 71–91. Springer Science+Buisness Media, Inc., New York, USA.

[9] ETSI (2001). Broadband Radio Access Networks (BRAN); HIPERLAN Type 2; Data Link Control (DLC); Part 1: Basic Transport Functions; TS 101 761-1; V1.3.1.

[10] Hoymann, C., Klagges, K., and Schinnenburg, M. (2006). Multihop Communication in Relay Enhanced IEEE 802.16 Networks. In *Proceedings of the 17th IEEE PIMRC*, page 4, Helsinki, Finland.

[11] IEEE (2004). IEEE Std 802.16-2004, IEEE Standard for Local and metropolitan area networks, Part 16: Air Interface for Fixed Broadband Wireless Access Systems.

[12] IEEE (2008a). IEEE 802.16j/D5, Draft Standard for Local and Metropolitan Area Networks Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems Multihop Relay Specification.

[13] IEEE (2008b). IEEE Std 802.16Rev2-D5, IEEE Draft Standard for Local and metropolitan area networks, Part 16: Air Interface for Fixed Broadband Wireless Access Systems.

[14] IETF (2001). Robust Header Compression (ROHC): Framework and Four Profiles: RTP, UDP, ESP, and Uncompressed. RFC 3095.

[15] Pabst, R., Walke, B., Schultz, D., and et al. (2004). Relay-Based Deployment Concepts for Wireless and Mobile Broadband Radio. *IEEE Communications Magazine*, pages 80–89.

[16] Taori, Rakesh and Zhu, Peiying (2008). 16jm ad hoc group recommendations on high level constructs for 16m relay frame structure. contribution to IEEE 802.16m.

[17] WINNER, IST-4-027756 (2006). Relaying concepts and supporting actions in the context of CGs. WINNER deliverable D3.5.1.

[18] WINNER, IST-4-027756 (2007). Final assessment of relaying concepts for all CGs scenarios under consideration of related WINNER L1 and L2 protocol functions. WINNER deliverable D3.5.3.

[19] Zirwas, W., Giebel, T., Esseling, N., Schulz, E., and Eichinger, J. (2002). Broadband multi hop networks with reduced protocol overhead. In *Proc. European Wireless 2002*, volume 1, pages 333–339, Florence, Italy.

SIGNAL FLOW GRAPH ANALYSIS - CAPABILITIES AND CONSTRAINTS

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Abstract This article provides an insight into the Signal Flow Graph (SFG) analysis of communication protocols. Different complexity levels of the analysis model are considered by using the example of an idealized Selective Repeat ARQ (SR-ARQ) protocol. First, the model is derived by using a constant block error probability and a deterministic block transmission delay. At a higher complexity level, correlated block errors and varying block transmission delays are considered. Furthermore different model representations, MGF space, state space and probability space, are introduced in order to ease the evaluation of the model and to extend this model with time varying aspects.

1. INTRODUCTION

Performance analysis of communication protocols across different communication layers are in the majority of cases performed by simulations. An analytical analysis is oftentimes only used for calculating mean values. Due to their complexity it is hard to transfer the models to similar problem statements.

Analytical performance analysis has its favors compared to simulations with regards to the model complexity and transparency. Analytical models aim for a reduction of the model complexity by keeping the accuracy of the model as high as needed. Before starting to model a system it is essential to specify the goal of the analysis, i.e. which performance criteria are considered. Hence, the needed model parameters can be deduced that lead to the required modeling accuracy.

Often, analytical models are rather difficult to understand, to extend or to reuse for similar problem statements. Models that can be separated similar to layers of a protocol stack are in favor, since a model layer can easily be exchanged by a different model if boundary conditions are followed. SFG models have been used in the last years to model different Automatic Repeat Request (ARQ) protocols, amongst others in [1], [2]. At the Department of

Communication Networks, RWTH Aachen University a hierarchical model has been proposed in [3] to evaluate the performance of Mobile Web Service calls including transport and radio layer modeling.

2. SIGNAL FLOW GRAPH ANALYSIS AND ITS APPLIANCE TO COMMUNICATION PROTOCOL ANALYSIS

SFGs provide means to model the statistical time response of communication protocols based on a graphical representation. Statistical states, transition probabilities and probability density functions are used for the parametrization of the SFGs. The graphs ease the adaptation, extension, the transition of the model to a different problem statement and the evaluation by using standard graph reduction rules and well established numerical evaluation methods.

In the following sections, the basics of SFGs, their appliance to communication protocol analysis and specific solution methods are provided. All analyzes are adverted to a simple SR-ARQ protocol without loss of generality.

2.1 Signal Flow Graphs

Originally, SFGs are introduced by [4] in order to analyze electronic circuits. In general, a SFG is a network of directed branches connecting a set of nodes. This graph represents a system of equations in which the nodes are values and the branch transmissions are the relations between these values.

For example, the equations

$$x = 3y + 2z \quad (1)$$

$$y = 7z \quad (2)$$

$$\Rightarrow \frac{x}{z} = 23 \quad (3)$$

can be represented by the SFG depicted in Figure 1.

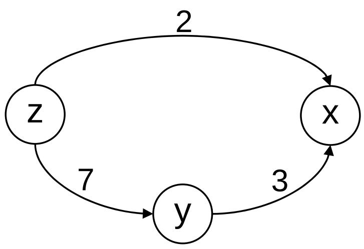

Figure 1. Signal graph example.

The resulting ratio x/z is obviously the product of the branch values from z via y to x added by the value from z to x.

Generally, SFGs consist of three basic topologies, parallel, serial, and self-loop branches. A thorough discussion on SFGs can be found in [4] and [5]. There, several reduction rules are conducted and different graph topologies are discussed. In these articles SFGs are applied to electronic circuit diagrams in order to calculate transfer functions.

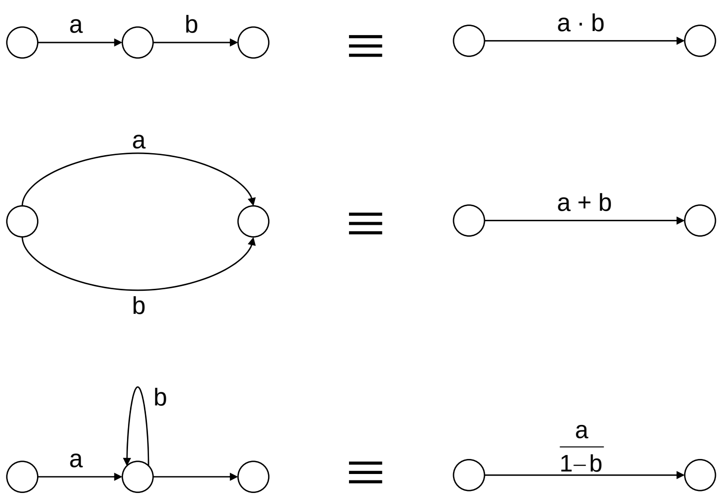

Figure 2. Basic signal graph reductions.

2.2 SFG Analysis of Communication Protocols

In [6] SFGs are used to find the Moment Generating Function (MGF) of the transmission time in order to analyze the mean throughput and delay of ARQ mechanisms. This model has been extended by [1] to matrix signal flow graphs to enable the analysis of ARQ mechanisms with a Hidden Markov Model (HMM) for the forward and reverse channel.

In a first step a model of a SR-ARQ protocol is presented that contains a slotted block transmission over a channel with an uniform block error probability p and a deterministic block transmission delay D. The delay is represented in the Missing Frame Generation (MFG) domain (z-domain) by the delay operator z. The SR-ARQ protocol continuously transmits block with a delay of z (transmission from node S to R in Figure 3). In $(1-p)$ of the cases the blocks are successfully transmitted (transition from node R to A). With a probability of p an error occurs and the block will be retransmitted (transition from node R via B to R) according to the reception of a NACK.

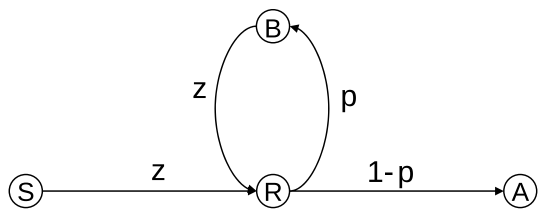

Figure 3. Signal Flow Graph of the basic Selective-Repeat ARQ with deterministic transmission delay D and uniform block error prob. p.

The SFG shown in Figure 3 can be easily reduced by applying the rules of Figure 2 to the MFG of Eq. 4.

$$G_{SR}(z) = (1-p) \cdot \frac{z}{1-pz} \tag{4}$$

2.3 Selective-Repeat ARQ Analysis with Varying Transmission Delay using Signal Flow Graphs

Data blocks of fixed size are transmitted over a communication channel with a varying delay. Assuming an unreliable channel with the block error probability p. The data blocks are delayed randomly by the value D characterized with a Probability Mass Function (PMF) $P_D(k)$. The delay is modeled as a discrete random process of values $k \in \mathbf{N}$. The MGF of D is respectively $G_D(z)$.

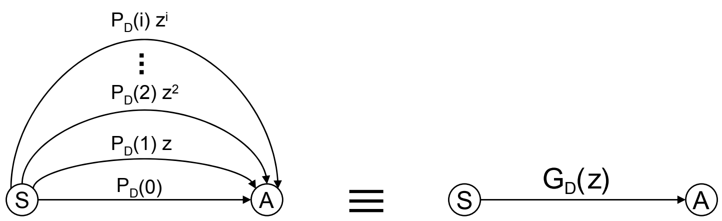

Figure 4. Signal Flow Graph of a delayed segment transmission.

Figure 4 illustrates a transmission of a segment over an error-free channel with the varying delay D using a SFG with nodes “Sending” (S) and “Arrival” (A). The transmission from (S) to (A) is delayed by $k = 0$ units with the probability $P_D(0)$, by $k = 1$ units with probability $P_D(1)$ and so on. Thus, the whole transmission process, from (S) to (A), can be statistically described by the MGF $G_D(z)$ (Figure 4).

In the case the channel is no longer error-free, the transmission has to be protected. In case a SR-ARQ is used to retransmit lost segments, they are continuously sent and acknowledged by the receiving node. It is assumed that the reverse (feedback) channel is reliable, and thus, all ACK and NACK messages will be transmitted error free.

If a continuous stream of segments is transmitted and each segment will be acknowledged, the transmission can be divided into 4 states as depicted in Figure 5. Within the sending state (S) a segment is transmitted and after a delay of D the transmission resides in the ACK/NACK receiving state (R). If an ACK has been received, the transmission is finished. In case of a received NACK, the transmission passes the retransmission state (B) causing an additional delay D.

The SFG of the segment transmission delay using a SR-ARQ is depicted in Figure 5.

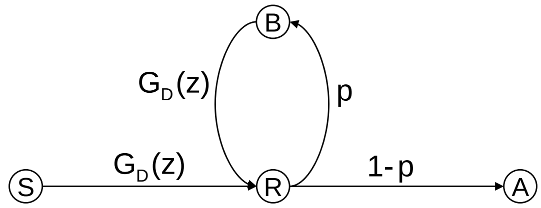

Figure 5. Signal Flow Graph of the Selective-Repeat ARQ with varying transmission delay.

After reducing the SFG, the following MGF results.

$$G_{SR}(z) = (1-p) \cdot \frac{G_D(z)}{1-p\,G_D(z)} \tag{5}$$

In order to derive the PMF of the delay caused by the SR-ARQ, generally, two options are possible. First, the resulting MGF can be derived in the z-domain by evaluating Eq. (5) and transforming the MGF in the co-domain. The second option is to calculate the resulting PMF only in the co-domain without transforming the individual functions to the z-domain. In doing so, a multiplication of MGFs is performed in the co-domain by convoluting the PMFs and a division of MGFs correspond to the de-convolution of the respective PMFs.

In Table 1 the discrete convolution, de-convolution, and n-fold convolution operators are defined. The random variables a and b are statistically independent and take values in the same co-domain k. The first column of Table 1 lists different relations of a and b. Their PMFs are $P_a(k)$ and $P_b(k)$. Accordingly their MGFs are denoted as $G_a(z)$ and $G_b(z)$. The last column additionally lists the corresponding Matlab functions used for numerical calculations.

Table 1. Convolution operator definitions.

	z-domain	co-domain	Matlab function
$a+b$	$G_a(z)\cdot G_b(z)$	$P_a(k) * P_b(k)$	conv(Pa, Pb)
–	$\frac{G_a(z)}{G_b(z)}$	$P_a(k) *^{-1} P_b(k)$	deconv(Pa, Pb)
$N\cdot a$	$(G_a(z))^N$	$P_a(k)^{*N}$	for i:1:N Pa=conv(Pa, Pa) end

Eq. (5) can be transformed to the co-domain by using the operator definitions of Table 1. The result is

$$P_{SR}(z) = (1-p)\cdot P_D(k) *^{-1} (\delta(k) - p P_D(k)) \tag{6}$$

SFG basics, first introduction by Mason [4] and reduction rules for SFGs.

2.4 Selective-Repeat ARQ Analysis with Varying Channel Conditions

In the following a model of an SR-ARQ is introduced which considers an erroneous channel [3]. The block error rate behavior is modeled by using a Gilbert-Elliott model [7], which is a HMM with two finite channel states. Each state hides a Block Error Ratio with value ε_i ($\varepsilon_1 = 0$ for the "Good" state and $\varepsilon_2 = 1$ for the "Bad" state). Thus, the probability of being in state 2 is at the same time the probability of an block error ($\overline{BLER} = p_2$). All the following calculations are valid also for an arbitrary number of channel states, but due to illustration reasons, the calculations are made for a 2-state HMM. The main advantage of a HMM is due to the use of parameter estimation methods, like the Baum-Welch algorithm [8]. The model parameter can be estimated from block error statistics coming from link layer simulations or measurements. The transitions between the channel states are statistically described by transition probabilities p_{ij} from state i to state j. As a whole, all transition probabilities are collected in the channel transition matrix $\mathbf{P}$, where i is the row index and j the column index.

$$\mathbf{P} = \begin{pmatrix} p_{11} & p_{12} \\ p_{21} & p_{22} \end{pmatrix} = \begin{pmatrix} 1-p_{12} & p_{12} \\ p_{21} & 1-p_{21} \end{pmatrix} \tag{7}$$

The hidden Block Error Ratios (BLERs) (ε_i) are consolidated in the row vector

$$\boldsymbol{\varepsilon} = (\varepsilon_1, \varepsilon_2) \tag{8}$$

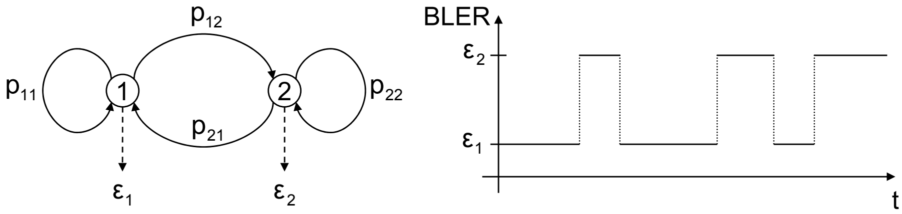

Figure 6. Structure of the analytical model.

In Figure 6 on the left hand side the 2-state HMM is depicted. Each state hides the corresponding BLERs ε_1 or ε_2. Viewing the variation of the BLER in time, the channel switches between these two states according to the transition probabilities. This leads to a correlation of consecutive block errors as subsequently investigated.

In general a HMM channel model is characterized by the state transition matrix $\mathbf{P}$, the error probability vector $\boldsymbol{\varepsilon}$ and the initial probability state vector $\boldsymbol{\pi_0}$. After shifting the channel states k-times, the channel state probability is

$$\boldsymbol{\pi}_k = \boldsymbol{\pi_0} \cdot \mathbf{P}^k \tag{9}$$

The stationary vector $\boldsymbol{\pi}$ of the HMM is the equilibrium probability vector of the channel states and can be derived through

$$\boldsymbol{\pi} \cdot \mathbf{P} = \boldsymbol{\pi} \tag{10}$$

$$\boldsymbol{\pi} \cdot \mathbf{1} = \mathbf{1} \tag{11}$$

The stationary vector of the canonical HMM is

$$\boldsymbol{\pi} = \left(\frac{p_{21}}{p_{12}+p_{21}}, \frac{p_{12}}{p_{12}+p_{21}} \right) \tag{12}$$

The average block error rate is

$$\overline{BLER} = \boldsymbol{\pi} \cdot \boldsymbol{\varepsilon'} = \frac{p_{21}\,\varepsilon_1 + p_{12}\,\varepsilon_2}{p_{12}+p_{21}} \tag{13}$$

The column vector $\boldsymbol{\varepsilon'}$ is the transpose of the row vector $\boldsymbol{\varepsilon}$.

Collecting the channel transition and observation probabilities into the conditional probability matrices $\mathbf{P_1}$ and $\mathbf{P_0}$ it follows

$$\mathbf{P_1} = \begin{pmatrix} p_{11}\,\varepsilon_1 & p_{12}\,\varepsilon_2 \\ p_{21}\,\varepsilon_1 & p_{22}\,\varepsilon_2 \end{pmatrix} = \begin{pmatrix} 0 & p_{12} \\ 0 & p_{22} \end{pmatrix} \tag{14}$$

$$\mathbf{P_0} = \begin{pmatrix} p_{11}\,(1-\varepsilon_1) & p_{12}\,(1-\varepsilon_2) \\ p_{21}\,(1-\varepsilon_1) & p_{22}\,(1-\varepsilon_2) \end{pmatrix} = \begin{pmatrix} p_{11} & 0 \\ p_{21} & 0 \end{pmatrix} \tag{15}$$

The matrices $\mathbf{P_1}$ and $\mathbf{P_0}$ are the transition probabilities under the condition that a block is transmitted correctly ($\mathbf{P_0}$) or not ($\mathbf{P_1}$).

The outcome of the canonical HMM is a discrete time random process $E(k)$, which indicates a block error $E(k) = 1$ or a correct block transmission $E(k) = 0$. Thus, the probabilities p_1 and p_2 of the states are

$$p_1 = 1 - \overline{BLER} = \frac{p_{21}}{p_{12} + p_{21}} \tag{16}$$

$$p_2 = \overline{BLER} = \frac{p_{12}}{p_{12} + p_{21}} \tag{17}$$

By using an SR-ARQ protocol, blocks are continuously sent out without waiting for the acknowledgements of previous blocks. All unacknowledged blocks remain in the receiver buffer. In this simple SR-ARQ, the block transmissions are independent from each other, i.e. multiple block transmission can easily be mapped to the case where only one block is transmitted (for further details see [3]).

The Matrix Signal Flow Graph of transmitting successfully one block by using the described SR-ARQ protocol over an erroneous forward HMM channel is depicted in Figure 7.

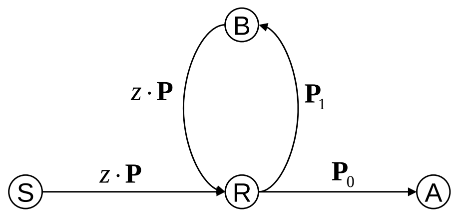

Figure 7. Matrix Signal Flow Graph of a SR-ARQ with error-free reverse (feedback) channel.

Starting from the Sending state `(S)` where the block is transmitted, the following state `(R)` is definitely reached after a delay of D_{Block}, which is indicated by the delay operator z. While moving to this state also the channel state of the HMM changes. Thus, the link from `(S)` to `(R)` is in addition weighted with the channel state transition probability matrix $\mathbf{P}$. In state `(R)` the receiver checks whether a block can be acknowledged or not. If the block can be acknowledged, the transmission is finished and the state `(A)` is reached with the probability $prob(Ch_k = j, E(k) = 0 | Ch_{k-1} = i) = p_{ij} \cdot (1 - \varepsilon_j)$. Thus, this link is weighted with the probability matrix $\mathbf{P_0}$ and there is no delay of the ACK transmission since the reverse channel is ignored. In case that a non-acknowledgement has been received, the state `(B)` is reached with the probability matrix $\mathbf{P_1}$. After the retransmission of the unacknowledged block (transition from (B) to (R) with $z \cdot \mathbf{P}$), the receiver checks again whether a ACK or NACK is required.

The total matrix Moment Generating Function (MGF) $\mathbf{G_{SR}}(z)$ is derived by applying basic graph reduction rules to

$$\mathbf{G_{SR}}(z) = z\mathbf{P}(\mathbf{I} - z\mathbf{P_1}\mathbf{P})^{-1}\mathbf{P_0} \tag{18}$$

The scalar MGF $G_{SR}(z)$ is obtained from the corresponding matrix MGF by pre-multiplying with the probability vector of transmitting a new block $\boldsymbol{\pi}_I$ and post-multiplying with the column vector of ones ($\mathbf{1}$) [9].

$$G_{SR}(z) = \frac{\boldsymbol{\pi}_I \, \mathbf{G_{SR}}(\mathbf{z}) \, \mathbf{1}}{\boldsymbol{\pi}_I \, \mathbf{1}} \tag{19}$$

with

$$\boldsymbol{\pi}_I = \boldsymbol{\pi} \cdot \mathbf{P_0} \tag{20}$$

The average transmission time of one block can be found by evaluating the first derivative of $G_{SR}(z)$ at $z = 1$. According to [9] this is calculated to

$$\overline{D}_{SR} = \frac{\partial}{\partial z}\Big\{G_{SR}(z)\Big\}_{z=1} = \frac{1}{1 - \overline{BLER}} = \frac{p_{12} + p_{21}}{p_{12} + p_{21} - \varepsilon_1 \, p_{21} - \varepsilon_2 \, p_{12}} \tag{21}$$

where $\overline{BLER}$ is the average block error rate of the HMM channel. $\overline{BLER}$ can be derived from the stationary vector $\boldsymbol{\pi}$ of the HMM and the block error probabilities $\boldsymbol{\varepsilon}$.

$$\overline{BLER} = \boldsymbol{\pi}\boldsymbol{\varepsilon}' = \boldsymbol{\pi}\mathbf{P_1}\mathbf{1} = \underbrace{\boldsymbol{\pi}\mathbf{P}}_{=\boldsymbol{\pi}} \underbrace{diag(\boldsymbol{\varepsilon})\,\mathbf{1}}_{=\boldsymbol{\varepsilon}'} \tag{22}$$

For the analysis of a wireless system it is not sufficient to consider only the average BLER, but also its distribution characteristics must be taken into account. The delay distribution is important to be known for the higher communication layer models in order to be able to consider timeouts.

The delay distribution, i.e. the Probability Mass Function (PMF) of the delay P_{SR}, can be derived from the following equations.

$$\begin{aligned} P_{SR}(k) &= \frac{1}{k!} \cdot \left(\frac{\partial^k}{\partial z^k} G_{SR}(z)\right)_{z=0} = Z^{-1}\left(G_{SR}\left(z^{-1}\right)\right) \\ P_{SR}(k) &\quad \circ\!\!-\!\!\bullet \quad G_{SR}\left(z^{-1}\right) \end{aligned} \tag{23}$$

Block Error Rates $\boldsymbol{\varepsilon} = (\mathbf{0}, \mathbf{1})$ In case of a canonical HMM, the probability mass function of the delay can easily be computed. First, the Matrix MGF is calculated by inserting the matrices $\mathbf{P_1}$ and $\mathbf{P_0}$.

Thereupon, the Matrix MGF can be transformed by applying Eq. (19) to the scalar MGF

$$G_{SR}(z) = z \cdot \frac{(p_{21} + p_{12} - 1) \cdot z + 1 - p_{12}}{(p_{21} - 1) \cdot z + 1} \tag{24}$$

The corresponding Probability Mass Function (PMF) $P_D(k)$ is derived by means of the inverse z-transformation of $G_{SR}(z^{-1})$.

$$\begin{aligned} P_D(k) =& (p_{12} + p_{21} - 1) \cdot \Theta(k-2) \cdot (1 - p_{21})^{(k-2)} \\ &+ (1 - p_{12}) \cdot \Theta(k-1) \cdot (1 - p_{21})^{(k-1)} \end{aligned} \tag{25}$$

$\Theta(k)$ is the discrete Heaviside function also known as unit step function.

From Equation 25 it can be seen that the SFG analysis is an appropriate tool to compute not only the mean value but also the delay distribution of an SR-ARQ protocol. This underlines that SFGs are helpful for the performance analysis of communication protocols. However, the next section shows some constraints of the SFG analysis, e.g. in the case of a Carrier Sense Multiple Access / Collision Avoidance (CSMA/CA) protocol when several users have to be considered at the same time, or when the considered system is a heterogeneous system, which means that from the user's perspective the system becomes tim varying.

3. CONSTRAINTS OF THE SFG ANALYSIS FOR HETEROGENEOUS NETWORKS

Signal Flow Graphs have proven to be a powerful tool for the investigation of the message delay. Especially the influence of each layer can be modeled easily. The combination of the single delays results in the overall message delay. However, this section shows some constraints of the SFG analysis by examining two typical networking scenarios. In the previous section only one user has been part of the study, but usually in a network there are several users which are using different services at the same time. Those users as well as their behavior may have an affect on each other. One further degree of freedom arises if the system under consideration is allowed to be heterogeneous. In a heterogeneous environment even the networks consist of a multitude of technologies. This aspect must be incorporated in the SFG analysis. A satisfied user criteria can give information about the experienced performance in those networks, e.g. in the presence of user mobility, and therefore the evaluation of the satisfied user criteria of all users in the system is the preferred performance metric over the pure study of the delay of a single message.

3.1 Defining Appropriate Performance Criteria

As mentioned before one of the challenges is to incorporate the number of users in the analysis as well as theirs specific service demands and last but not least the available technologies. The problem is twofold, on the one hand there is the category of networks with explicit reservation of resources for a user which simplifies the analytical model a lot because each user can be modeled almost independently from other users. The random access procedure used for the channel reservation is negligible as long as the system is designed for an appropriate number of parallel reservation requests. The SFG analysis of the previous section could be extended easily in order to tackle the modeling of multiple users, e.g. if the number of channel errors is a function of the number of users served by the system the parameters of the HMM have to be adapted according to the number of users. Different technologies could be modeled by their own SFG. If users move through an area which is served by those different technologies, handovers are very likely to occur for several reasons. Handover performance may have a non-negligible effect on the satisfied user criteria. From the SFG modeling perspective a handover is then equivalent with an exchange of the SFG. The time to carry out the switching procedure could, e.g. be modeled a constant delay penalty or could have its own SFG representation. On the other hand if the random access procedure cannot be neglected, e.g. if CSMA/CA-like behavior is considered, the analytical model has to be extended essentially which is for further study.

Another challenge is to find appropriate performance criteria for the performance analysis of today's heterogeneous (consisting of more than one radio access technology) communication networks. Under this circumstances it may make sense to define a satisfied user criteria which could be used to make statements about the performance of the whole system and how the different services the users are using are affected. But imagine the following cases:

Scenario A A user terminal is connected via technology one. The user moves and the received quality from technology one indicates that the current technology will not be able to support the quality of service demands of the user's application anymore. Without any support for heterogeneous networks the service delivery will be aborted sooner or later. The user terminal will be disconnected. If the user can (re-) connect the device to another technology the service usage could be continued after a short break (break before make vertical handover). In terms of a satisfied user criteria the user is satisfied as long as the user receives the service in an adequate quality. During a disruption of the service the user is not satisfied. The time needed for reconnecting and restarting the service delivery together with an assigned weight for the overall costs gives the value of the satisfied user criteria. This highly depends on the deployment topology, the user's skills, and of the ability of the service to be restarted (imagine that the service is a live video stream of a football match, a voice call, or a simple web browsing session). The technologies as such do not have such a large influence on the satisfied user criteria, because the association to a radio access network operates on the timescale of milliseconds, whereas IP mobility may result in breaks longer than a second.

Scenario B The user terminal supports heterogeneous networks and prepares the handoff before the connectivity is broken (make before break vertical handover). This behavior results in a seamless handover if the user terminal would detect technology two early enough and the technology is appropriate for the current service demands. Layer three mobility is still an issue but could also be prepared in advance (e.g., together with IEEE 802.21 [10] and fast Mobile Internet Protocol (MIP). In that case the user is not interrupted at all and therefore remains satisfied.

As can be seen from the previous examples the user satisfaction highly depends on the short and seldom service disruption during the vertical handover if any. At the best there is no impact on the satisfied user criteria at all. This arises the following questions:

- How to model those relatively seldom events?
- How important are they in reality? How much do they affect the users satisfaction?
- Is there a preferred solution for loosely coupled networks? What is the role of layer two in that case? In tightly coupled networks, as in Universal Mobile Telecommunications System (UMTS) and Wireless Local Area Network (WLAN) [11] the problem does not exist at all, because layer three mobility is handled within the system itself. So, is the problem relevant or not?

The key to the answers of those questions is how different technologies cooperate. The technologies from the 3GPP family, like General Packet Radio Service (GPRS), Enhanced Datarates for GSM Evolution (EDGE), UMTS, High Speed Packet Access (HSPA), and Long Term Evolution (LTE) have well defined methods of collaboration. This collaboration is achieved through the close coupling of these technologies. Roaming between different operators is assisted as well and therefore handover execution between different technologies of two operators is possible. In the IEEE familiy, like Worldwide Interoperability for Microwave Access (WiMAX) and WLAN the interworking is based on the IEEE 802.21 standard. Although IEEE 802.21 defines also interaction with technologies of the 3GPP family there has not been much progress in that area so far.

3.2 IEEE 802.21 in Heterogeneous Networks

The IEEE 802.21 Media Independent Handover standard helps to understand the role of layer two (respectively layer 2.5) in heterogeneous networks. IEEE 802.21 offers a convergence layer for receiving events, e.g. Link Up or Link Going Down, from 802 technologies, retrieving information, e.g. measurement reports or about neighbor access points, and for sending command, e.g. Connect or Disconnect. This hides the complexity (but also some features) of the underlying technologies and allows a unified interaction with those technologies. Of course this is always a trade off. The IEEE 802.21 standard proposes to incorporate also non-IEEE standard technologies into this framework. This approach sounds very promising. The handover could be assisted and performed using this framework. Whenever specific features of the access technology are needed this belongs to the responsibility of this technology and must not be seen from the outside. Together IPv6 respectively the proposed Mobility for Mobile IPv6 standard [12] the mobility problem could be solved as suggested in [13]. Furthermore additional RRM functions could be assisted, e.g. load balancing. There should be a tool to investigate if the performance of a system using the IEEE 802.21 framework is sufficient or if it can be improved. Besides simulation of the involved mechanisms it is beneficial to have a analytical model to make objective statements about the achievable performance of the system, upper and lower bounds, and expected outcome.

Coming back to the question how can signal flow graphs assist the evaluation of those kind of scenarios. The technologies as such can be modeled for sure if the random access problem is not considered for the moment. However the system's reaction to triggers and commands increases the dynamics within the system, e.g. the access selection mechanism has to avoid unnecessary ping-pong handovers. From the modeling perspective the system becomes a time-varying system.

3.3 From SFG to Time-Varying Systems

The following section explains how the signal flow graph model can be extended to cover also time-varying systems. Therefore the signal flow graph model is translated to a state-space model of a Linear Time Invariant (LTI) system. In literature, e.g. [14], two different forms of the transfer operator to describe the state-space model (pre-/post multiplying the transfer operator **T**) exist. Here we use post-multiplication:

$$y_k = u_k \mathbf{T} \Leftrightarrow \begin{cases} \mathbf{x_{k+1}} & = \mathbf{x_k A} + \mathbf{u_k B} \\ \mathbf{y_k} & = \mathbf{x_k C} + \mathbf{u_k D} \end{cases} \tag{26}$$

where $\mathbf{u}$ is a row vector representing the input signal and $\mathbf{u_k}$ is the input signal at time k. If the different u_k are scalar then the signal is a one-channel signal (Single Input Single Output (SISO)). If the $\mathbf{u_k}$ are rows themselves then $\mathbf{u}$ is a multi-channel signal (Multiple Input Multiple Output (MIMO)). $\mathbf{y_k}$ is the output row vector of the output at time k. $\mathbf{x_k}$ is the state vector of the system. $\mathbf{A}$, $\mathbf{B}$, $\mathbf{C}$, and $\mathbf{D}$ are components of the transfer operator $\mathbf{T}$. $\mathbf{T}$ is not only used to compute the output of the system but also to obtain the state transition $\mathbf{x_k} \leftrightarrow \mathbf{x_{k+1}}$, so that $\mathbf{T}$ has the form:

$$\mathbf{T} = \left(\begin{array}{c|c} \mathbf{A} & \mathbf{C} \\ \mathbf{B} & \mathbf{D} \end{array}\right) \tag{27}$$

The state space representation of the delay distribution shown in Figure 8 can be found by translating the scalar MGF of the SR-ARQ

$$G_{SR}(z) = z \cdot \frac{(p_{21} + p_{12} - 1) \cdot z + 1 - p_{12}}{(p_{21} - 1) \cdot z + 1} \tag{28}$$

to a state space representation. The delay distribution as shown in Figure 8 for $p_{12} = 0.4$ and $p_{21} = 0.4$ is simply the impulse response of the time discrete system. The components of the transfer operator $\mathbf{T}$ are in this case:

$$\mathbf{T} = \left(\begin{array}{cc|c} 0.6 & 1 & 0.6 \\ 0 & 0 & -0.2 \\ \hline 1 & 0 & 0 \end{array}\right) \tag{29}$$

with

$$\mathbf{A} = \begin{pmatrix} 0.6 & 1 \\ 0 & 0 \end{pmatrix}, \mathbf{B} = \begin{pmatrix} 1 & 0 \end{pmatrix}, \mathbf{C} = \begin{pmatrix} 0.6 \\ -0.2 \end{pmatrix}, \text{and } \mathbf{D} = 0 \tag{30}$$

The graphical representation of those components can be explained by Figure 9. There the input $\mathbf{u_k}$ at the moment k is stored in the state variable $\mathbf{x_{1_{k+1}}}$. The output is a combination of the two state variables x_{1_k} and x_{2_k}. x_{1_k} affects the future state of the system $(x_{1_{k+1}}, x_{2_{k+1}})$ whereas x_{2_k} does not.

When the system becomes time dependent, the entries of $\mathbf{T}$ become time-varying themselves. Therefore they receive the subscript k. The dimension of the matrixes $\mathbf{A_k}$, $\mathbf{B_k}$, $\mathbf{C_k}$, and $\mathbf{D_k}$ have not necessarily to be constant as shown in Figure 10. The amount of states the system posses may vary from the moment k to the other $k+1$. Equation 26 becomes:

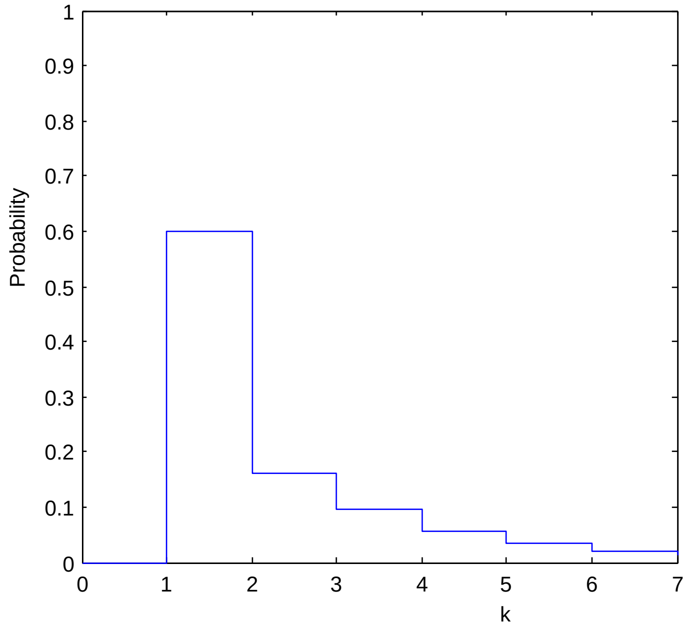

Figure 8. Delay distribution of the link layer SR-ARQ.

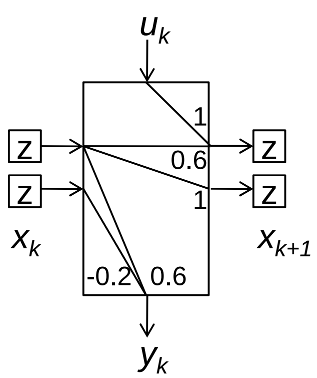

Figure 9. State space model for delay distribution of the link layer SR-ARQ.

$$[\mathbf{x_{k+1} y_k}] = [\mathbf{x_k u_k}]\mathbf{T_k} \Leftrightarrow \begin{cases} \mathbf{x_{k+1}} & = \mathbf{x_k A_k} + \mathbf{u_k B_k} \\ \mathbf{y_k} & = \mathbf{x_k C_k} + \mathbf{u_k D_k} \end{cases} \tag{31}$$

An index free form of the dynamical state equations from Equation 31 is

$$\mathbf{x}Z^{-1} = \mathbf{x\,A} + \mathbf{u_k\,B} \tag{32}$$

$$\mathbf{y} = \mathbf{x\,C} + \mathbf{u\,D} \tag{33}$$

where $Z^{-}1$ is the delay operator which shifts a sequence one position to the left. Then together with the equations

$$\mathbf{x} = \mathbf{u\,BZ}\,(\mathbf{I} - \mathbf{AZ})^{-1} \tag{34}$$

and

$$\mathbf{y} = \mathbf{u}[\mathbf{D} + \mathbf{BZ}(\mathbf{I} - \mathbf{AZ})^{-1}\mathbf{C}] \tag{35}$$

T has the solution

$$\mathbf{T} = \mathbf{D} + \mathbf{BZ}(\mathbf{I} - \mathbf{AZ})^{-1}\mathbf{C} \tag{36}$$

where **A**,**B**,**C**, and **D** are diagonal operators and $\mathbf{T} \in \mathscr{U}$ is an upper triangular matrix as in Equation 37.

$$\mathbf{T} = \begin{pmatrix} \ddots & \vdots & & & \vdots & \\ & \mathbf{D_{-1}} & \mathbf{B_{-1}C_0} & \mathbf{B_{-1}A_0C_1} & \mathbf{B_{-1}A_0A_1C_2} & \cdots \\ & & \mathbf{D_0} & \mathbf{B_0C_1} & \mathbf{B_0A_1C_2} & \\ & & & \mathbf{D_1} & \mathbf{B_1C_2} & \\ & & \mathbf{0} & & \mathbf{D_2} & \cdots \\ & & & & & \ddots \end{pmatrix} \tag{37}$$

A graphical representation of this model can be found in Figure 10. At $k = 0$ the input $\mathbf{u_0}$ affects the output $\mathbf{y_0}$ according to the matrix $\mathbf{D_0}$ and the state $\mathbf{x_1}$ as described by $\mathbf{B_0}$. The old state $\mathbf{x_{-1}}$ influences $\mathbf{y_0}$ by $\mathbf{C_0}$ and $\mathbf{x_1}$ by $\mathbf{A_0}$.

In the example of the SR-ARQ **T** has Toeplitz structure (constant along the diagonals):

$$\mathbf{T} = \begin{pmatrix} 0 & 0.6000 & 0.1600 & 0.0960 & 0.0576 & \dots \\ 0 & 0 & 0.6000 & 0.1600 & 0.0960 & \dots \\ 0 & 0 & 0 & 0.6000 & 0.1600 & \dots \\ 0 & 0 & 0 & 0 & 0.6000 & \dots \\ 0 & 0 & 0 & 0 & 0 & \ddots \end{pmatrix}. \tag{38}$$

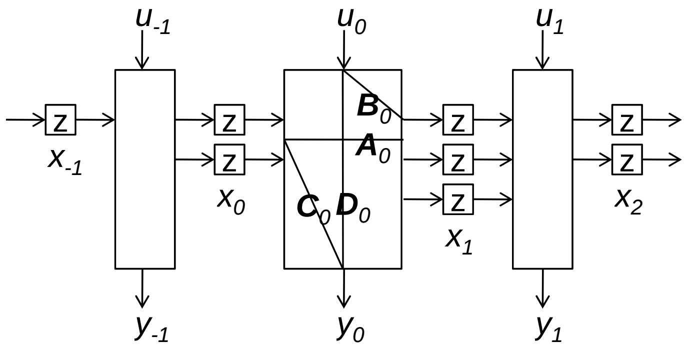

Figure 10. Time-varying state realization [14].

When a system is subject to permanent change the time and data needed for modeling the system is infinite. Therefore the dimension of **T** is infinite, see Equation 37. However, if a system belongs to the class of systems that are "initially time-invariant or periodic, then start to change, and become again time-invariant or periodic after some finite period (time-invariant or periodic at the borders)" then the computations can be carried out in finite time [14] and matrix **T** has finite size. Another class of systems where the computations can be carried out in finite time and **T** is a finite matrix is the class where the input and output for i outside of an interval $I = [1, n]$ are zero. This is the case for the SR-ARQ of a single packet with limited number of retransmission attempts.

In the delay analysis of the SR-ARQ each packet, respectively its delay distribution, has been investigated independently. If the packet is discarded after a maximum number of retransmissions, the SFG does not give the result anymore. In this case the transfer operator **T** is a banded matrix again with Toeplitz structure:

$$\mathbf{T} = \begin{pmatrix} \mathbf{T_{11}} & \mathbf{T_{12}} & \cdots & \mathbf{T_{1d}} & 0 & 0 \\ & \mathbf{T_{22}} & \cdots & \mathbf{T_{2d}} & \mathbf{T_{2,d+1}} & 0 \\ & & \ddots & \ddots & \ddots & \mathbf{T_{n-d+1,n}} \\ & & & & \mathbf{T_{n-1,n-1}} & \mathbf{T_{n-1,n}} \\ \mathbf{0} & & & & & \mathbf{T_{n,n}} \end{pmatrix}. \tag{39}$$

In Equation 39 the width of the band is d. $d-1$ equals the number of transmission attempts, including the first transmission. The input is always delayed for one discrete time increment, because $T_{11} = 0$. Hence $d-2$ is the number of retransmissions. After d transmissions the first input of the sequence $\mathbf{u_k}$ has no impact on the output anymore and the corresponding entries in **T** are zero.

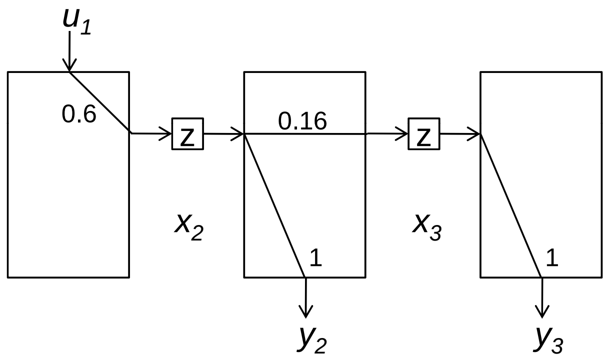

Figure 11. SR-ARQ with one retransmission.

In the example of the SR-ARQ with one retransmission attempt $\mathbf{T}$ is a banded matrix:

$$\mathbf{T} = \begin{pmatrix} 0 & 0.60 & 0.16 & 0 & 0 & \dots \\ 0 & 0 & 0.60 & 0.16 & 0 & \dots \\ 0 & 0 & 0 & 0.60 & 0.16 & \dots \\ 0 & 0 & 0 & 0 & 0.60 & \dots \\ 0 & 0 & 0 & 0 & 0 & \ddots \end{pmatrix}. \tag{40}$$

The entries on the diagonals are still constant. However, the realizations (decomposition of $\mathbf{T}$ into $\mathbf{A}$, $\mathbf{B}$, $\mathbf{C}$, and $\mathbf{D}$) vary with k. A realization of $\mathbf{T}$ can be found by a sequence of realizations ($\mathbf{T_1}$, $\mathbf{T_2}$, $\cdots$, $\mathbf{T_n}$). This realization must not be minimal in the sense, that the number of required states is as small as possible.

If only one packet is considered the input is zero for $i \neq 1$ and the output is zero for $i \geq 4$. Therefore the interval to consider is $I = [1,3]$. Input is only present at $i = 1$ and output is only nonzero at $i = 1$ and $i = 3$. That is why the transfer operator consists exactly of the two components T_{12} and T_{13}. The number of required states equals the rank of T_{12} and T_{13} respectively. In the case of the example T_{12} and T_{13} are scalars with rank one and $T_{12} = 0.6$ and $T_{13} = 0.16$. From Figure 11 it is obvious that the transfer operator changes over time.

3.4 From Time-Varying Systems to SFG

In the previous section it has been shown that it is possible to translate an SFG model to a state space model. Then the state space model has been changed in order to cover time varying aspects. In terms of an SFG the time varying aspects are switches that are turned on or off, depending on which SFG should be used. However, if the system that is represented by the corresponding SFG is not in its steady state when the system changes the analogy may not be that simple and is for further study.

4. CONCLUSION AND OUTLOOK

SFGs are an adequate tool for the analytical study of layered communication systems and theirs protocols. This article showed that it is possible to model protocols like the SR-ARQ with different complexity levels. Further investigations, which are not shown within this paper, have proven that the results obtained through the SFG analysis are in line with results of different analysis techniques and simulations. Furthermore in [3] it was shown, that the model is applicable for link layer protocols like GPRS as well as for transport layer protocols. However, the greatest strength of the SFG analysis is that it can be extended easily to cover the aspects of multiple layers, different technologies, or any arbitrarily number of communication hops. If time-varying aspects of the system have to be considered, e.g. when a handover occurs and the link layer technology is changed, the SFG concept can be extended with the help of state space models. Future research will concentrate on the applicability of those modeling techniques to concrete heterogeneous systems and required interworking functionality.

ACKNOWLEDGEMENT

The authors would like to express theirs heartfelt gratitude to Prof.-Dr.-Ing. Bernhard Walke for his time, advice, and visions.

References

[1] K. Ausavapattanakun and A. Nosratinia, "Analysis of selective repeat arq via matrix signal flow graphs," *IEEE Transactions on Wireless Communications*, jun 2005.

[2] M. Rossi, L. Badia, and M. Zorzi, "Sr arq delay statistics on n-state markov channels with non-instantaneous feedback," *Wireless Communications, IEEE Transactions on*, vol. 5, no. 6, pp. 1526–1536, June 2006. [Online]. Available: www.dei.unipd.it/~rossi/papers/ARQNstate_TW.pdf

[3] G. Gehlen, "Mobile web services - concepts, prototype, and traffic performance analysis," Ph.D. dissertation, RWTH Aachen University, Lehrstuhl fr Kommunikationsnetze, Aachen, Germany, Oct 2007. [Online]. Available: http://www.comnets.rwth-aachen.de

[4] S. J. Mason, "Feedback theory — some properties of signal flow graphs." *Proceedings of the Institute of Radio Engineers*, vol. 41, no. 9, pp. 1144–1156, Sep. 1953.

[5] S. J. Mason, "Feedback theory — further properties of signal flow graphs." *Proceedings of the Institute of Radio Engineers*, vol. 44, no. 7, pp. 920–926, Jul. 1956.

[6] D.-L. Lu and J.-F. Chang, “Analysis of arq protocols via signal flow graphs,” *IEEE Transactions on Communications*, vol. 37, no. 3pp, pp. 245–251, Mar 1989.

[7] E. Gilbert, “Capacity of a burst-noise channel,” *Bell Syst. Tech. J*, vol. 39, no. 5, pp. 1253–1265, 1960.

[8] L. Baum, T. Petrie, G. Soules, and N. Weiss, “A Maximization Technique Occurring in the Statistical Analysis of Probabilistic Functions of Markov Chains,” *The Annals of Mathematical Statistics*, vol. 41, no. 1, pp. 164–171, 1970.

[9] W. Turin, *Performance Analysis and Modeling of Digital Transmission Systems*, 3rd ed., W. Turin, Ed. Kluwer Academic, May 2004.

[10] I. P802.21/D9.0, “Draft standard for local and metropolitan area networks: Media independent handover services,” on the Internet, February 2008.

[11] 3GPP, “3rd generation partnership project; ts 24.234: Technical specification group core network and terminals; 3gpp system to wireless local area network (wlan) interworking; wlan user equipment (wlan ue) to network protocols; stage 3 (release 8),” on the Internet, June 2008.

[12] J. A. D. Johnson, C. Perkins, “Rfc 3775: Mobility support in ipv6,” on the Internet, June 2004.

[13] G. Lampropoulos, A. Salkintzis, and N. Passas, “Media-independent handover for seamless service provision in heterogeneous networks,” vol. 46, no. 1, pp. 64–71, 2008.

[14] P. Dewilde and A.-J. van der Veen, *Time-Varying Systems and Computations*. Kluwer Academic Publishers, 1998.

openWNS: OPEN SOURCE WIRELESS NETWORK SIMULATOR

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Abstract Technological research and innovation are pushed forward as fast as never before in history. To keep the pace up, researchers require effective methods and tools. Computer aided simulation has proved its applicability in various fields of research. It became as important as mathematical modeling and experiments. This also applies to the field of telecommunication research and development. A multitude of simulators and simulator development tool boxes are available. Each has its individual strengths and weaknesses. The Wireless Network Simulator (WNS) developed at the Department of Communication Networks (ComNets) at RWTH Aachen University has been created with focus on simulating mobile radio networks.

This article gives an overview of the WNS. This includes the history of simulator development at ComNets as well as an description of provided radio channel and mobility models. A brief description of the Layer Development Kit (LDK) is given. The LDK provides simulator developers with a tool set to rapidly develop and evaluate current and future communication protocols.

WNS is currently made publicly available as the Open Source Wireless Network Simulator (openWNS).

1. Introduction

As wireless communication systems and protocols get more complex with every new generation, the task of evaluating them becomes more challenging. Analytical modeling can help fulfill this task but has its limits. With no prototype hardware available in the design phase no measurements for performance evaluation are possible. Simulation of network protocols and the environment they operate in can help to fill this gap. Stochastic event-driven simulation has proved its applicability in this field in many different research projects. In the field of wireless network protocol simulation random number generators (RNG) provided by stochastic simulators help to model the influence of net-

work traffic generation, user mobility, and radio propagation on protocol performance. The event-driven characteristic models the effect of protocol state transition on event occurrence, like the creation of new user data or change in position.
The level of detail used to model a communication system is mainly limited by available computational power. Still, the simulator model needs to be implemented, which can be very time consuming. When modeling network protocols, simplifications are possible to focus on relevant effects, while leaving out parts with no or negligible influence. Protocols can even be fully implemented providing a protocol emulation rather than a model. Still, the environment the protocol operates in needs some simplifications and modeling considerations. This includes end user or software activity creating network traffic and the highly complex physics of the wireless channel.
Many software tools for stochastic event-driven simulation exist. Some of them are limited to a minimum, providing random number generators (RN) and scheduling of events in simulation time. Others have been specially designed for communication network simulation. They provide functions and building blocks especially for this task like receivers, transmitters, protocol data units, finite state machines and network protocol layer design support. The Wireless Network Simulator (WNS) developed at the Department of Communication Networks (ComNets) of RWTH Aachen University is a stochastic event-driven simulator specially developed for evaluation of wireless network protocols. Its purpose is to provide researchers a tool to build and develop network simulators. Depending on the research topic, modeling detail can be chosen by flexibly adding or removing complexity and functionality from provided environment models and protocol stack implementations.
This paper gives an overview of the WNS. Section 2 depicts the development history of simulators at ComNets up to the WNS. Section 3 describes the core functionalities provided by WNS. Sections 4 and 5 present two high-level frameworks: the WNS physical (PHY) layer, and channel models and the Layer Development Kit (LDK). Section 6 concludes the paper and gives an outlook for the future.

2. History

The history of simulator development at ComNets has always been a history of protocol specification and verification on the one hand and protocol performance evaluation on the other. The Specification and Description Language (SDL) as recommended by the International Telecommunication Union (ITU) Telecommunication Standardization Sector (ITU-T) [1] was commonly used for the formal specification and verification of protocols. To evaluate the performance of a future protocol with no hardware yet available either analytical modeling or computer simulation was used. While SDL originally only allowed to specify a protocol and verify correct protocol behavior, computer programming languages allow to develop executable protocol implementation. C++ has been the commonly used programming language at ComNets [2].

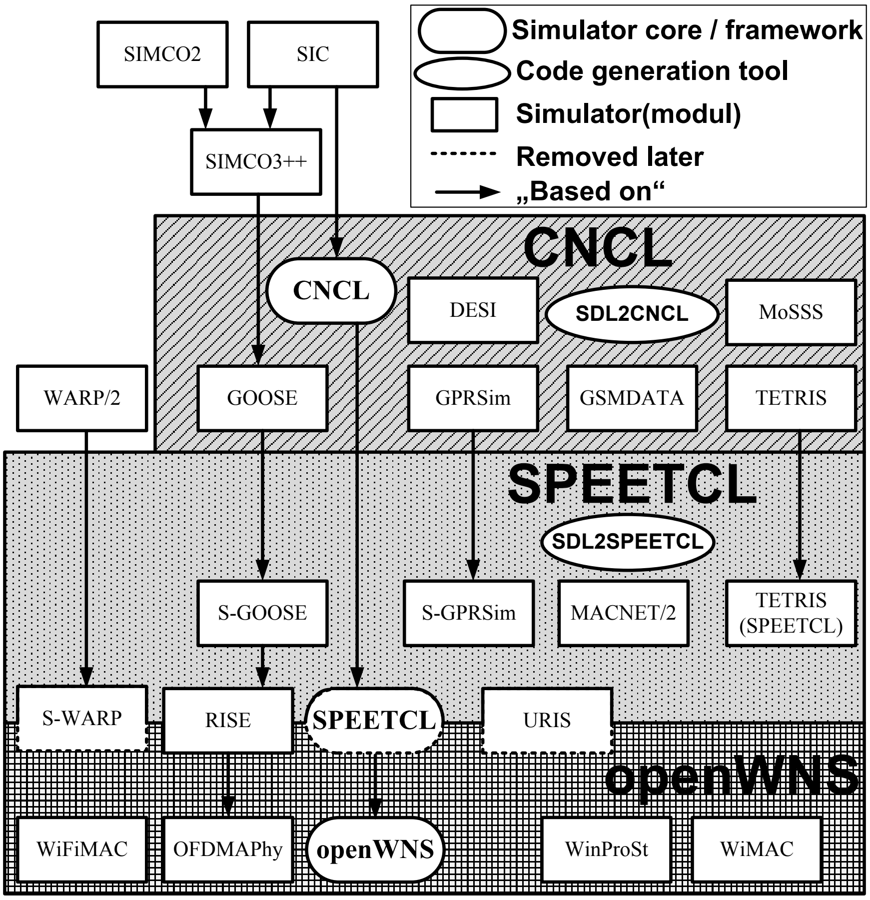

Figure 1. History of Simulator Development at ComNets.

ComNets Class Library (CNCL)

Figure 1 gives a rough overview of how network simulators evolved at ComNets. Evolving in this context means that existing simulators have been further extended and combined. This extension could be done by adding new functionality, refactoring existing code, or by reusing and extending ideas and concept found in an existing simulator to form a new one. Frameworks and class libraries providing common functionality were developed to simplify and speed up simulator development. One of the first frameworks provided for network simulator development was the ComNets Class Library (CNCL) [3; 4; 5]. CNCL provides the essentials required for stochastic event-driven simulation. Those include RNG, event scheduling, and statistical evaluation. RNG and statistical evaluation are based on implementations from Simulation in C++ (SIC) simulator [6]. CNCL includes a tree-like class hierarchy, Run-time Type Information (RTTI), Weak-Typing, and general container classes. Just like SDL, CNCL implements protocols as extended finite state machines with events causing state transitions. Therefore a tool called SDL2CNCL is included allowing to translate formal SDL protocol specifications to executable CNCL code. The following simulators, reflecting the researched protocols at ComNets, were created using CNCL:

- Digital Enhanced Cordless Telecommunication (DECT) Simulator (DESI) [7]
- General Packet Radio Service (GPRS) Simulator (GPRSim) [8; 9]
- Mobile Satellite System Simulator (MoSSS) [10]
- Global System for Mobile Communications (GSM) Simulator (GSM-DATA) [11]
- Terrestrial Trunked Radio (TETRA) Simulator (TETRIS) [12]
- GSM Object Oriented Simulation Environment (GOOSE) [3]

Besides the fact that all simulators were based on CNCL they had some higher-level functionality in common. As mobile network simulators they required mobility, radio propagation and channel models. Those were provided based on existing code from Simulation of Mobile Communication systems (SIMCO3++) [13]. Functionality was bundled to form a framework for PHY layer and channel modeling with a defined interface for higher layers of the protocol stack. This framework was called General Object Oriented Simulation Environment (GOOSE), with the same abbreviation the GSM simulator had. The layers above the PHY layer were usually system dependent. Some of them were fully specified using SDL. For example TETRIS and DESI data link layers (DLL) were implemented in SDL and translated using SDL2CNCL. Traffic sources and sinks were used above DLL to model user traffic in the systems. Those were again generic and could be reused by different simulators.

SDL Performance Evaluation Tool Class Library (SPEETCL)

The idea of coupling higher-level building blocks like protocol layers or even complete simulators was further developed leading to the development of the SDL Performance Evaluation Tool (SPEET) [14; 15]. SPEET provided the required concept to couple simulators defining interfaces to pass events and providing common simulation time. CNCL was further developed adding for example exception handling and became SPEET Class Library (SPEETCL). SDL2CNCL was extended to support more advanced SDL schemes. It became SDL2SPEETCL. Channel and PHY layer modeling framework GOOSE was adjusted to SPEETCL and renamed to S-GOOSE. Soon followed GPRSim and TETRIS. With new protocols being investigated at ComNets new simulators were created based on SPEETCL. Those include Multihop Ad Hoc Network Simulator for HyperLAN/2 (H/2) and 802.11a (MACNET/2) [16] and the Universal Mobile Telecommunications System (UMTS) Radio Interface Simulator (URIS) [17].
As the C++ programming language and SDL tools evolved, some functionalities of CNCL and SPEETCL frameworks became obsolete. The Standard Template Library (STL) became part of standard C++ providing a powerful set of container classes. Weak-Typing and RTTI became part of standard C++ [2]. Also the SDL Design Tool (SDT) by Telelogic [18] further evolved and gained the ability to generate executable C++ code from SDL specifications. This was used to develop the Wireless Access Radio Protocol (WARP/2) IEEE 802.11a and HyperLAN/2 simulator [19; 20]. WARP/2 was fully implemented in SDL including traffic generators, channel and mobility models. Later a SPEETCL version of WARP/2 was built called S-WARP. This was done for research on Media Independent Handover (MIH) with multi-mode enabled devices [21]. Using the SPEETCL framework a simulator environment supporting mobile nodes with IEEE 802.11a (S-WARP) and GPRS/UMTS (S-GPRSim) network access was created [22]. This was one further step to create a simulator, putting all available protocol implementations together as modules with defined interfaces between them and a common environment to configure and deploy simulation scenarios. This simulator was called Wireless Network Simulator (WNS). The S-GOOSE PHY layer and channel modeling framework was further developed and renamed to Radio Interference Simulation Engine (RISE).

Wireless Network Simulator (WNS)

At the beginning of WNS development SPEETCL formed an essential part of the WNS code. This included the RNG, the event scheduler and the possibility to use protocol layers implemented in SDL. This way the interfaces and configuration of S-WARP and URIS could be adjusted to become part of WNS. In a later development step it was decided to remove all SDL from WNS. S-WARP was replaced by a C++ WLAN simulator called WiFiMAC. URIS was totally removed. This step was taken since SDL in its graphical representation proved to be inapplicable for distributed software development. This was contrary to the WNS philosophy of providing a platform for distributed software development including tools for version control and testing. Also present in WNS is a WiMAX simulator called WiMAC [23] and a Wireless World Initiative New Radio (WINNER) simulator called WinProSt [24]. Both of them and WiFiMAC use the Orthogonal Frequency-Division Multiple Access (OFDMA) PHY layer based on the RISE framework. Currently WNS is made available to the public under the Lesser General Public License (LGPL) in the hope that it will prove useful to the scientific community. The goal is to provide a simulator core together with high-level frameworks and protocol implementations using those frameworks. The core and the frameworks are presented in the next sections.

3. WNS Core

The WNS core provides the essentials to execute simulations. Those include an event scheduler, a random number generator, statistical evaluation, and configuration.

Event Scheduler

The WNS scheduler uses functionality provided by the Boost Function and Boost Bind C++ libraries [25]. The Boost Function library provides the required template class to create a function object from any given C++ method. The template arguments specify method parameter count and type and the return type. The scheduler can schedule a Boost Function object for execution at a given simulation time. This way any method with an arbitrary number and type of parameters can be scheduled for execution. Using the Boost Bind library, methods can be bound to objects of a given class and executed considering and modifying the state of the object. Boost Function and Boost Bind are both candidates to be included in the next ISO C++ standard [26].

Random Number Generator

WNS uses the Boost Random library to generate random numbers following different distributions. The basic generator generating uniformly distributed random numbers is a Mersenne Twister MT19937 [27]. The key properties of the basic generator are negligible correlation between sequentially drawn numbers, a period length of $2^{19937} - 1$ magnitude, and a very fast implementation. The basic generator can be used to create RNGs providing different distributions. Boost Random library includes RNGs for Bernoulli, Cauchy, Exponential, Normal, Log-Normal and Uniform on n-dimensional sphere distributions. As a candidate for standard C++, Boost Random will likely replace the C *rand()* method in future C++ compiler libraries.

Statistical Evaluation

Statistical evaluation methods already known from CNCL and SPEETCL [5; 14] are also present in WNS. Measurements, represented by double precision floating point values, are passed to the statistical evaluation system together with the current simulation time and a so called *Context*. The *Context* is formed of key-value pairs with the keys being text strings and the values being text strings or integer numbers. Possible keys could be for example the Medium Access Control (MAC) address of the receiving node, the current X or Y position, or the type of the node. Context cannot only be provided by the probing node but can also come from a received Protocol Data Unit (PDU). Such context information could be the MAC address of the PDU sender, the number of hops the PDU traveled, or even the modulation and coding scheme (MCS) used to transmit the PDU. Context information can be used to build filters and sort measurements to be processed by the desired statistical evaluation object. This way a separate evaluation for example per source address, per destination address or per MCS can be achieved.

Configuration

WNS simulations are constructed and configured using the Python programming language [28]. Python is an object-oriented programming language. The object oriented techniques inheritance and composition, help to set up simulation scenarios by reflecting properties of the real world. Inheritance can be found when network nodes using the same communication protocol act with different roles. For example in 802.11 nodes can be stations or access points. Stations and access points all come with the same parameter set to for example configure the MAC protocol. Additionally stations can have special parameters to configure the scanning process while access points can have additional parameters to choose the frequency channel they operate on. There are many

examples were composition is used to set up simulation scenarios. Scenarios are composed of nodes while nodes can be composed of multiple different protocol layers. Each protocol layer within a node can again be composed of different building blocks.
Using a programming language for simulator configuration also supports tasks like network address assignment by providing arithmetic operations and control structures.

4. PHY Layer and Channel Modeling Framework

WNS includes a PHY layer and channel model framework called Radio Interference Simulator Engine (RISE). The framework offers several channel, scenario and mobility models, and common building blocks for PHY layers.

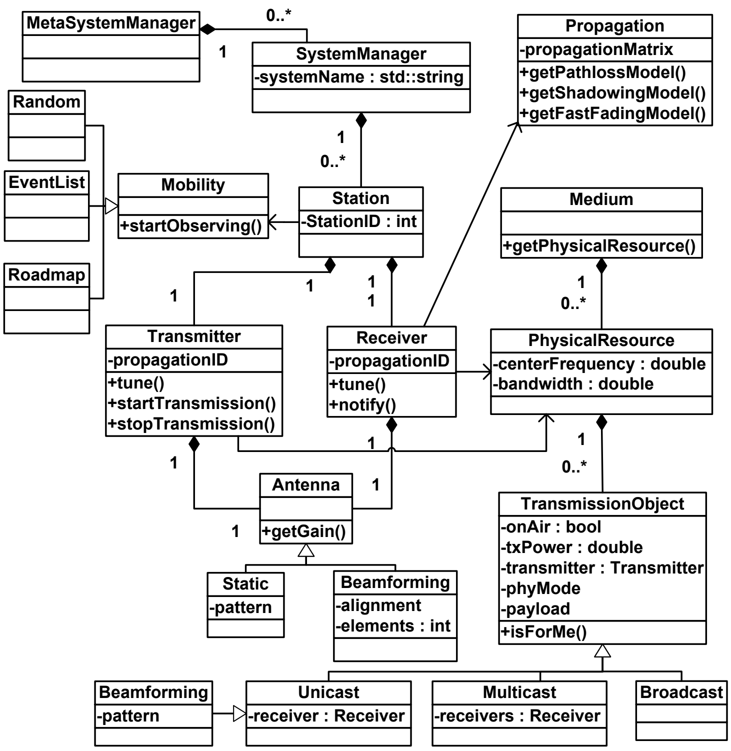

Figure 2. RISE framework classes.

Framework Classes

Figure 2 gives an extract of some of the main classes of the RISE framework. Mobile radio stations belonging to one radio access technology together form a system. Each system is managed by a *SystemManager*. Since it is possible to have different radio access technologies in one simulation scenario all *SystemManagers* register at the *MetaSystemManager*. The *SystemManager* offers the possibility for stations to communicate using the simulation environment rather than the wireless channel. Each station is composed of a receiver and transmitter which can have a static or beamforming antenna. Each receiver-transmitter type pair has an own channel model. *propagationID*s are used to look up the corresponding propagation model in the *PropagationMatrix*.
The wireless medium is represented by multiple *PhysicalResources*. Each resource is defined by its center frequency and bandwidth. Receivers and transmitters are attached to physical resources using the *tune* method. Transmitters can execute a transmission using the *startTransmission* and *stopTrasmission* methods. These methods create a *TrasmissionObject* held by the *PhysicalResource*. Multiple transmissions on one *PhysicalResource* are possible. The receiver is notified about each starting and stopping transmission using the *notify* method. It can access all *TransmissionObjects* to decide which of them supplies an intended signal S or interference I. There are specialized *TransmissionObjects* for unicast and multicast transmissions. They can be used to optimize runtime performance. Receivers receiving a *TransmissionObject* not addressed for them will ignore it as long as it is not required to calculate the interference for an already ongoing reception.

Radio Propagation Models

The channel model is used to calculate total received signal strength for every transmission using formula (1).

$$P_R = P_T - L_{PL} - L_{Sh} - L_{FF} + G_T + G_R \tag{1}$$

P_R is the received power, P_T the total emitted power by the transmitter, L_{Sh}, L_{FF}, L_{PL} the losses due to shadowing, fast fading and path-loss, and G_T, G_R the antenna gains at the transmitter and receiver.

Several models to calculate the path loss between transmitter and receiver are available. Those are:

- Constant (distance independent)
- Free space
- Single slope

- Multi slope

Distance ranges can be defined and a model applied for each range. The single slope model is described by the equation $L_{PL} = (\frac{\lambda}{4\pi d})^{\gamma}$. d is the distance between transmitter and receiver, λ the electromagnetic wavelength and γ the propagation coefficient. In a logarithmic notation γ becomes the slope. Free space propagation is a special case of the single slope model with $\gamma = 2$.

The multi slope model is created by defining multiple distance ranges using single slope propagations with different propagation factors. Constant, distance independent path loss is usually applied for very short or very long distances.

Different shadowing models to describe the scenario are available. These models describe the influence of solid obstacles on radio wave propagation. Three different models are available:

- Map based
- Scenery object based
- Spatially correlated

The map based model assumes fixed base stations communicating with mobile stations. Each base station has a map attached describing the strength of its emitted signal at several sampling points on the scenario. The signal strength at a current mobile station position is then interpolated.
In the scenery object based model the scenario is represented by the composition of geometric objects together with their position on the scenario and the attenuation they cause when penetrated. For every transmission the connecting line between sender and receiver is calculated. In the next step total attenuation is computed by summing up the attenuations of all penetrated objects. This is typically used to create indoor scenarios with walls or outdoor scenarios with whole buildings. In contrast to the map based model this model does not require fixed base stations to be one communication end point. It can therefore be used for mobile-to-mobile station communication.
Spatially correlated shadowing uses a statistic approach to model the scenario. A description of the model can be found in [29]. It is based on a sequence of correlated, log-normally distributed random values.

Additionally to shadowing and path-loss, a fast fading model can be enabled. Currently Rice Fading [30] is available.
Two antenna types are supported. The static antenna is described by its gain in all directions. The beamforming antenna allows to dynamically adjust its

directivity. The algorithm used to calculate the gain is the optimal beamformer algorithm described in [31].
The radio propagation model can be independently chosen for each transceiver type pair. This can be used to for example have different models for different moving speeds or to define line of sight (LOS) and non line of sight (NLOS) connections.

Mobility

The RISE framework supports three mobility models on a two dimensional plane. Random direction, event list based, and road map based. The random direction model assumes a uniformly distributed moving direction in the range $[0;2\pi]$. The distribution of the velocity can be configured. A standard uniformly distributed random number is drawn periodically to decide if a new velocity vector should be calculated by drawing a new random direction and velocity.
Event list based mobility takes a list of simulation time and position pairs. The position of the station is updated whenever the given simulation time of an event is reached.
The road map model allows to define the streets and crossings that mobile stations can move on. Each station has a constant velocity assigned at creation time drawn from an RNG of configurable distribution function. Stations move along a street until they reach a crossing. The next street is then chosen randomly [3]. Each street connected to a crossing has a weighting factor defining its probability to be chosen.

5. Layer Development Kit (LDK)

Network protocols of different layers and technology standards often have functionalities in common. Reasons for that are similar problems and challenges they have to overcome and the fact that often one protocol evolved from another.

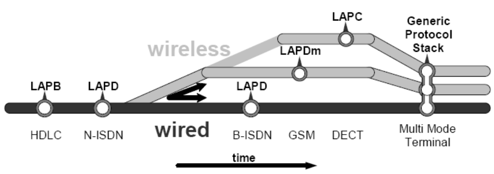

Figure 3. Generic protocol stack as a common basis for wired/wireless networks considering HDLC-based signaling protocols as an example [32].

Figure 3 gives an example of such a protocol evolution from wired High-Level Data Link Control (HDLC) to wireless DECT and GSM protocols. Each protocol consists of similar functionality like Automatic Repeat Request (ARQ), flow management and medium access control. Still, each protocol implements it in a flavour optimized for its working environment. This encourages the idea to build multi-mode enabled devices able to adapt their behavior depending on the environment they operate in [32]. This is achieved by dividing protocols into generic and specific parts.
From the software engineering point of view specialized behavior can be reached by parameterization of generic modules, by inheritance or by combination of both. In the context of network protocol design this simplifies the task of building multi-mode devices following the principles of Software Defined Radio (SDR). In network simulator development this provides a powerful toolset for rapid protocol model development.
This toolset comes with the Layer Development Kit of WNS providing a set of building blocks called Functional Units (FUs) [33]. Those FUs can be either parameterized at configuration time or alternatively be used as base classes providing the required interface and base functionality for specialized FUs implementing intended behavior. FUs can be interconnected to form Functional Unit Networks (FUNs) to jointly fulfill the task of a protocol layer.

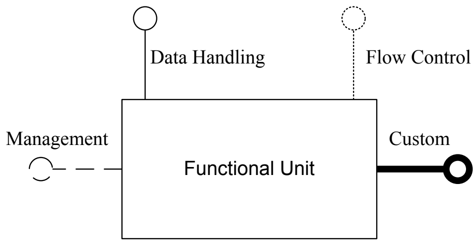

Figure 4. FU Interfaces.

As shown in Figure 4 each FU provides up to four interfaces. Those are Management, Data Handling, Flow Control, and optionally a Custom Interface. Figure 5 shows the methods provided by each interface. The Management Interface provides the *connect* and the *onFUNcreated* methods. The *connect* method is used to form a FUN out of multiple FUs. The *onFUNcreated* method is called for each FU when the whole FUN has been constructed. FUs can then resolve inter-FU dependencies as described later. The methods of the Data Handling interface *sendData* and *receiveData* are used to pass data units between FUs. Protocol data units (PDUs) are called Compounds in the LDK. They comprise the user data part or service data unit (SDU) and the protocol control information (PCI) called the Command Pool. A more detailed descrip-

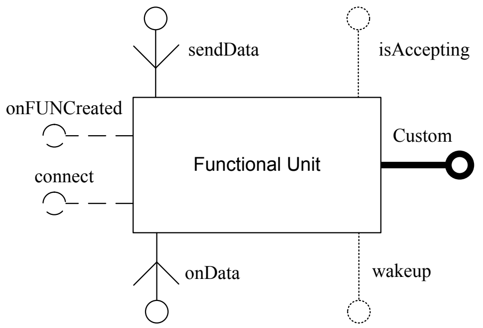

Figure 5. Methods provided by FU interfaces.

tion on Compounds and Command Pools can be found in [33; 24]. The Flow Control interface provides the methods *wakeup* and *isAccepting*. They determine when to pass a Compound between FUs. To demonstrate the application of FUNs an example is given in the next section.

FUN Example: Parameterizable Radio MAC (PaRaMAC)

In order to evaluate Media Independent Handover (MIH) as defined by IEEE 802.21 standard, a simulator module was implemented using many FUs provided by the LDK. While for the task of handover management and execution specialized classes were developed, user data is handled by general purpose FUs. Figure 6 shows the composition of the PaRaMAC FUN. Without going into detail some applications of FUs will be explained on the basis of this example FUN.

Following FUs are present in the PaRaMAC:

Upper Convergence:. This FU provides a defined interface to pass Compounds between PaRaMAC and the next higher layer. Next higher layer can be IP for user data or MIH Function (MIHF) layer sending and receiving handover management data. Upper Convergence FU distinguishes between Compounds from IP and those from MIHF and marks Compounds from MIHF to have higher priority. Upper Convergence FU also writes the source and destination MAC address into the Compound.

Upper, Lower Gate, and Flow Separator:. The FUs between the Upper and Lower Gate can be present multiple times. A Flow Separator FU creates the FUs for each flow. Here a flow is a logical connection between a base station (BS) and a mobile node (MN) associated with it. If no association

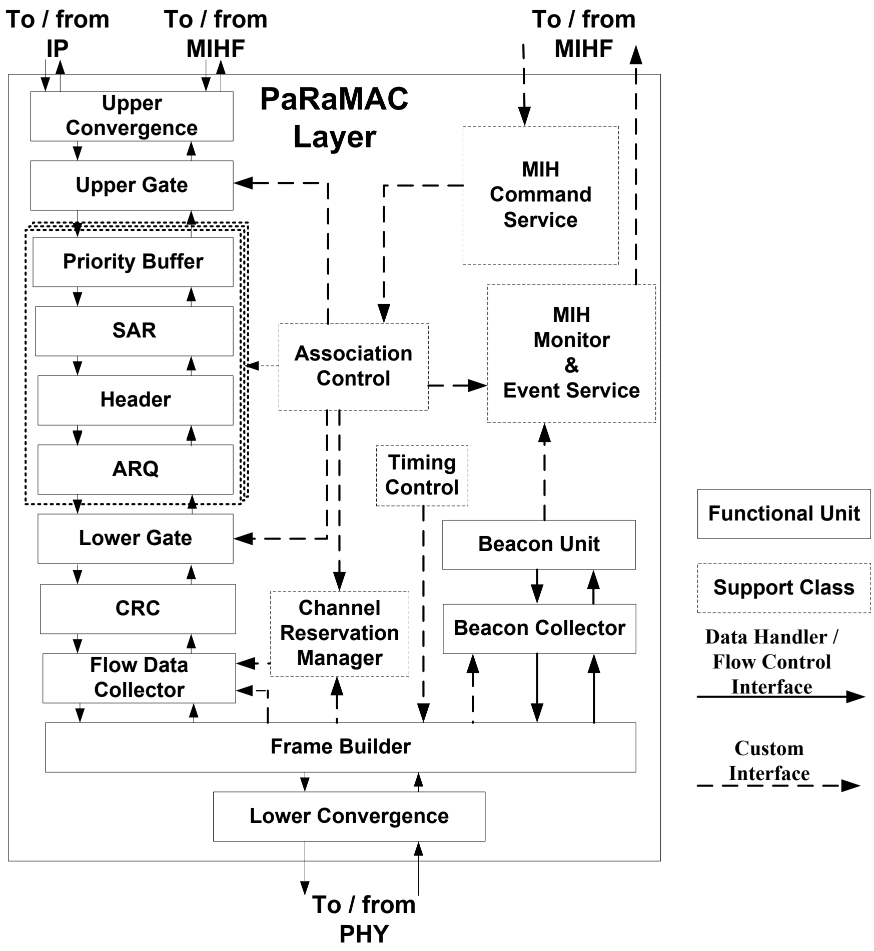

Figure 6. PaRaMAC FUN and Support Classes.

exists, the Gates drop any compounds for flows not present.

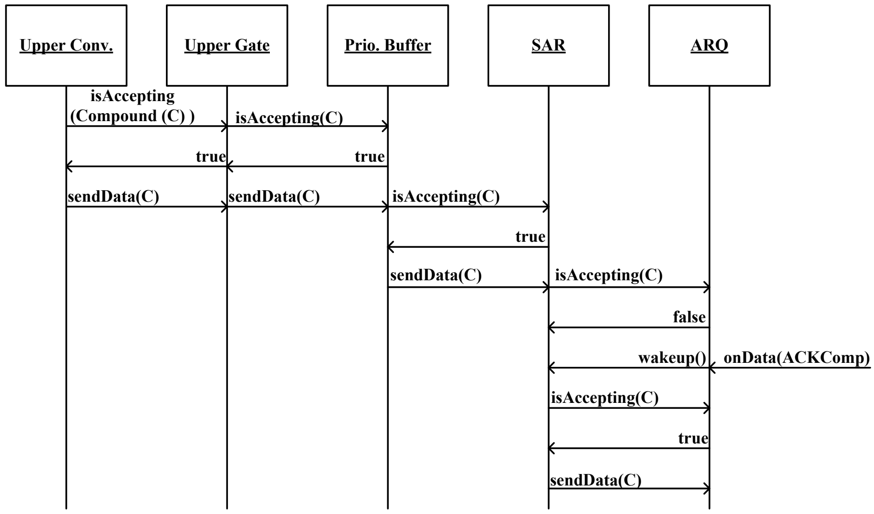

Figure 7. Signaling of the Flow Control and Data Handler Interface.

While Upper Convergence, Gate and Flow Separator FUs only process Compounds by manipulating or dropping them the FUs below them are able to store Compounds. To determine where a Compound is stored the Flow Control interface is used. Figure 7 shows the signaling protocol. Upper Convergence and Upper Gate FUs do not have memory and therefore can not store Compounds. They just pass an *isAccepting* call to lower FUs. The *isAccepting* call takes the Compound as an argument. This way FUs can base their decision if they accept a Compound on the Compound itself. For example a buffer can take into account the size in bit of the Compound to determine whether it has capacity left to store it. In this example the buffer has capacity, so it accepts the Compound. It is then passed down to the buffer. The buffer then tries to pass the Compound further down the FUN. The Segmentation and Reassembly (SAR) FU also has capacity. The ARQ, implemented as a Selective Repeat ARQ, does not have any capacity left in its sending window and therefore refuses to accept the Compound. The Compound is then stored in the SAR FU. If at a later point in time the ARQ FU receives an acknowledgement it can extend its sending window and transmit further Compounds. It therefore signals the SAR FU with a *wakeup* call. SAR FU again has to call the *isAccepting* method before passing the Compound. This is required since for example ARQ could have only limited space in its sending window and has to check if the Compound would fit.

Priority Buffer:. The Priority Buffer stores Compounds until lower FUs are ready to handle them. It has two queues, one for low and one for priority traffic. It reads the Command information provided by the Upper Convergence FU to decide where to queue a Compound. It therefore checks inside its *onFUNcreated* method if another FU that can provide priority information is present in the FUN. If not it acts like a normal buffer. The Priority Buffer is implemented as a dropping buffer. It therefore always accepts Compounds from FUs above but drops them if the queue has no capacity left. LDK also provides buffers without priority and blocking buffers not accepting Compounds when full.

Segmentation and Reassembly (SAR):. The SAR FU divides Compounds into chunks of configured maximum size in bit. On the sending end it stores the Compound currently being processed until the last chunk is transmitted. On the receiving end it waits until all chunks are received before passing the reassembled Compound to the next higher FU.

Automatic Repeat Request (ARQ):. In PaRaMAC a Selective Repeat ARQ with configured window size is present. LDK also provides other ARQ FUs like Stop-and-Wait, Go-Back-N and Cumulative ARQ.

Cyclic Redundancy Check (CRC):. CRC FU provides an error detection model at the receiving FUN. It therefore depends on another FU to provide the packet error rate (PER) of a received Compound. In PaRaMAC PER is provided by Lower Convergence FU which calculates it from the bit error rate (BER) provided by the PHY layer. The CRC FU draws a standard uniformly distributed random number and compares it with provided PER to decide whether to drop or pass a Compound.

Compound Collectors and Frame Builder:. The LDK offers a Frame Configuration Framework (FCF) designed to simplify implementation of MAC protocols. The focus is on periodic, centrally controlled MAC protocols. The FCF provides the interfaces to build FUs implementing a desired MAC protocol. In PaRaMAC those FUs were implemented to provide a basic model for handover evaluation. Only user data traffic and beacons required to determine signal strength are explicitly modeled. Other tasks like connection establishment and release and scheduling are done using the simulator environment. Using the Flow Control interface Compound Collectors open and close data flows depending on the station currently allowed to transmit.

They implement custom interfaces used by the Reservation Manager to start and stop flows.

PaRaMAC has been successfully parameterized and verified against analytical and other simulator results to evaluate WiMAX, WINNER and GPRS handover in heterogeneous and homogenous environments. It demonstrates the strength of the LDK as a toolset of building blocks providing commonly required generalized functionality which can be specialized to model a given protocol.

6. Conclusion and Outlook

WNS is a network simulator specially developed for mobile radio networks. It has therefore a powerful PHY layer, mobility and channel modeling framework. Another goal of WNS is to provide researchers with a tool to speed up the development of network simulators. The included LDK provides a set of Functional Units providing commonly required functionalities in network simulator design. At present the simulator core is publicly available for download under the Lesser General Public License (LGPL) [34].
Further parts are currently prepared for release. In the next step a fully featured IEEE 802.11 WLAN protocol stack will be published including traffic sources and sinks, and TCP/IP. The layers of the protocol stack are constructed using FUs from the LDK. An OFDM PHY layer based on the RISE framework is also included.

ACKNOWLEDGEMENT

The authors would like to thank all ComNets research assistants and their students who helped to develop the WNS in the past years. Special thanks go to the members of the ComNets team who have put great effort into reviewing, documenting and adapting WNS code to be published as openWNS.
WNS reflects the broad field of research done at ComNets. This includes topics on almost all layers of the ISO/OSI protocol stack "from application to antenna". At the same time WNS reflects the interdisciplinary character of research at ComNets bringing together Communication Engineers and Computer Scientists to mutually benefit from each others expertise. This was made possible thanks to Professor Bernhard Walke who has built up, directed, and managed ComNets for many years. Besides just gathering research results Professor Walke always attached great importance on methods and tools to achieve these results.

References

[1] ITU-T, "Recommendation Z.100. Specification and Description Language (SDL)," International Telecommunication Union – Standardization Sector, Tech. Rep. Z-100, November 2000.

[2] International Organization for Standardization, *ISO/IEC 14882:2003: Programming languages — C++*, 2003.

[3] M. Junius, "Leistungsbewertung intelligenter Handover-Verfahren für zellulare Mobilfunksysteme," Ph.D. dissertation, RWTH Aachen University, 1995.

[4] M. Junius and C. Görg, "Ein Programmpaket zur ereignisorientierten Simulation in C++," in *Messung, Modellierung und Bewertung von Rechen- und Kommunikationssystemen, Kurzberichte und Werkzeugvorstellung*, B. Walke and O. Spaniol, Eds., Aachen, 1993.

[5] C. Görg, *Verkehrstheoretische Modelle und Stochastische Simulationstechniken zur Leistungsanalyse von Kommunikationsnetzen*. Verlag der Augustinus Buchhandlung, 1997.

[6] C. Görg, B. Kluth, and U. Salewski, "SIC a system for stochastic simulation in C++," in *Proceedings of the 1991 European Simulation Multiconference*, E. Mosekilde, Ed., 1991, pp. 413–414.

[7] H. Hußmann, "Algorithmen zur Kapazitätsoptimierung schnurloser Mobilfunksysteme nach DECT-Standard," Ph.D. dissertation, RWTH Aachen University, 1997.

[8] G. Brasche, "Prototypische Bewertung und Implementierung von neuen Paket-Datendiensten für das GSM-Mobilfunksystem," Ph.D. dissertation, RWTH Aachen University, 1999.

[9] P. Stuckmann, "Traffic Engineering Concepts for Cellular Packet Radio Networks with Quality of Service Support," Ph.D. dissertation, Jun 2003.

[10] B. Bjelajac, "Modellierung und Leistungsbewertung von mobilen Satellitensysteme mit dynamischer Kanalvergabe," Ph.D. dissertation, RWTH Aachen University, 1999.

[11] P. Decker, "Entwurf und Leistungsbewertung hybrider Fehlersicherungsprotokolle für paketierte Sprach- und Datendienste im GSM-Mobilfunksystem," Ph.D. dissertation, RWTH Aachen University, 1997.

[12] M. Steppler, "TETRIS – A simulation tool for TETRA systems," in *Mobile Kommunikation*, ser. ITG–Fachbericht, B. Walke, Ed., vol. 135, Informationstechnische Gesellschaft (ITG) im Verein Deutscher Elektrotechniker (VDE). Berlin, Offenbach: VDE–Verlag GmbH, Sep 1995, pp. 403–410.

[13] C.-H. Rokitansky, A. Guntsch, B. Bjelajac, and C. Wietfeld, "SIMCO3++: Simulation of Mobile Communications for Performance Evaluation of Cellular Networks, Short-Range Communications and Satellite Communications," in *Proceedings of RACE Mobile Telecommunications Workshop*, May 1994.

[14] M. Steppler, "Leistungsbewertung von TETRA-Mobilfunksystemen durch Analyse und Emulation ihrer Protokolle," Ph.D. dissertation, RWTH Aachen University, 2002.

[15] S. Böhmer, "Entwurf eines ATM-basierten Funknetzes und Software-Entwurfsmethodik zur Implementierung," Ph.D. dissertation, RWTH Aachen University, 1996.

[16] J. Peetz, "Multihop Ad-hoc Kommunikation mit dynamischer Frequenzwahl, Leistungssteuerung und Ratenanpassung für drahtlose Netze im 5 GHz Band," Ph.D. dissertation, RWTH Aachen University, feb 2003.

[17] S. Heier, "Leistungsbewertung der UMTS Funkschnittstelle," Ph.D. dissertation, RWTH Aachen University, 2003.

[18] "SDT a tool for complex real-time systems," in *ICECCS '95: Proceedings of the 1st International Conference on Engineering of Complex Computer Systems*. Washington, DC, USA: IEEE Computer Society, 1995, p. 21.

[19] A. Kadelka, "Entwurf und Leistungsanalyse des mobilen Internetzugangs ber HIPERLAN/2," Ph.D. dissertation, RWTH Aachen University, 2002.

[20] S. Mangold, "Analysis of IEEE 802.11e and Application of Game Models for Support of Quality-of-Service in Coexisting Wireless Networks," Ph.D. dissertation, RWTH Aachen University, Jun 2003.

[21] M. Siebert, "Interworking of Wireless and Mobile Networks based on Location Information," Ph.D. dissertation, RWTH Aachen University, Lehrstuhl fr Kommunikationsnetze, Aachen, Germany, Dec 2006.

[22] E. Weiss, U. Fornefeld, B. Xu, and S. Hischke, "Performance Evaluation of A Mobile Internet Architecture with IEEE 802.11a WLAN and GPRS/UMTS," in *Proceedings of The Seventh International Symposium on Wireless Personal Multimedia Communications, WPMC04*, Abano Terme, Italy, Sep 2004, pp. V3, 130–135.

[23] C. Hoymann, "IEEE 802.16 Metropolitan Area Network with SDMA Enhancement," Ph.D. dissertation, RWTH Aachen University, 2008.

[24] M. Schinnenburg, R. Pabst, K. Klagges, and B. Walke, "A Software Architecture for Modular Implementation of Adaptive Protocol Stacks," in *MMBnet Workshop 2007*, University of Hamburg, Computer Science Department, TKRN - Telecommunication and Computer Networks Group. Hamburg, Germany: Wolfinger, Bernd and Heidtmann, Klaus, Sep 2007, pp. 94–103.

[25] Boost C++ Libraries. [Online]. Available: http://www.boost.org

[26] ISO, "Proposed Draft Technical Report on C++ Library Extensions," International Organization for Standardization ISO/IEC, Tech. Rep. PDTR 19768, Jan 2005.

[27] M. Matsumoto and T. Nishimura, "Mersenne twister: a 623-dimensionally equidistributed uniform pseudo-random number generator," *ACM Trans. Model. Comput. Simul.*, vol. 8, no. 1, pp. 3–30, 1998.

[28] Python Programming Language. [Online]. Available: http://www.python.org

[29] Z. Wang, E. Tameh, and A. Nix, "Joint Shadowing Process in Urban Peer-to-Peer Radio Channels," *Vehicular Technology, IEEE Transactions on*, vol. 57, no. 1, pp. 52–64, Jan. 2008.

[30] B. Walke, *Mobile Radio Networks - Networking, Protocols and Traffic Performance*, 2nd ed. New York, NY, USA: John Wiley & Sons, 2002.

[31] L. C. Godera, "Application of Antenna Arrays to Mobile Communications, Part ii: Beam-Forming and Direction-of-Arrival Considerations," in *Proceedings of the IEEE, 85(8)*, 1997, pp. 1193–1245.

[32] L. Berlemann, M. Siebert, and B. Walke, "Software Defined Protocols Based on Generic Protocol Functions for Wired and Wireless Networks," *2003 Software Defined Radio Technical Conference*, Nov 2003.

[33] M. Schinnenburg, F. Debus, A. Otyakmaz, L. Berlemann, and R. Pabst, "A Framework for Reconfigurable Functions of a Multi-Mode Protocol Layer," in *Proceedings of SDR Forum 2005*, Los Angeles, U.S., Nov 2005, p. 6.

[34] Open Source Wireless Network Simulator (openWNS). [Online]. Available: http://www.openWNS.org

C-DCF: AN EFFICIENT MAC PROTOCOL FOR W-LANS

Georgios Orfanos

Abstract In this work a new protocol for next generation indoor Wireless Local Area Networks (W-LANs) is presented, that is based on Multi Carrier – Code Division Multiple Access (MC-CDMA) and the Distributed Coordination Function (DCF) of IEEE 802.11 [1]. Its performance is evaluated, by means of stochastic event driven simulation. The introduced protocols aim at achieving high Medium Access Control (MAC) protocol efficiency and support of Quality of Service (QoS). For this purpose, solutions to technical problems for the realization of MC-CDMA based protocols are provided and additional adaptation rules are designed to allow good operation of the wireless system under the conditions of real operation scenarios.

1. MC-CDMA

C-CDMA is a recently developed digital modulation technique [2], combining the robustness of Orthogonal Frequency Division Multiplexing (OFDM), with the diversity of frequency spreading at a reasonably low complexity. In an asynchronous transmission system, such as the proposed Coded – Distributed Coordination Function (C-DCF), timing mismatch destroys the orthogonality among different subcarriers and different users' spreading codes [3], resulting mainly in Multiple Access Interference (MAI).

Assuming the use of Minimum Mean Square Error (MMSE) Multiuser Detector (MUD) at the receivers [4] and the application of the K=7 convolutional encoder/decoder, link level results have been generated [5] for the 20 MHz WLAN channel at the 5.2 GHz band. They serve for the evaluation of both, the new transmission technique, and the link quality as a basis for protocol performance simulation studies. In order to evaluate the link quality, the air interface for Signal to Interference and Noise Ratio (SINR) calculation is carefully modelled, taking into account the emissions per codechannel (cch) and the relative delays of asynchronous transmissions of concurrent Mobile Stations (MS)s transmitting in parallel. This is necessary in order to model accurately MAI. An example of the performance results is shown in Figure 1 in form of the theoretical Packet Error Rate (PER) for a 1514 byte long packet as a function of SINR.

2. C-DCF

2.1 Basic Protocol

The proposed MAC protocol, for the support of MC-CDMA, is a modification of the IEEE 802.11a MAC, namely C-DCF. In this case the frequency channel is divided into 4 parallel cchs (SF = 4). After selecting a cch the MS applies the rules of DCF for data tranmission [6]. The only difference is that MSs in idle state monitor all cchs (aided by the MUD) thus maintaining 4 Network Alocation Vectors (NAV) (Figure 2).

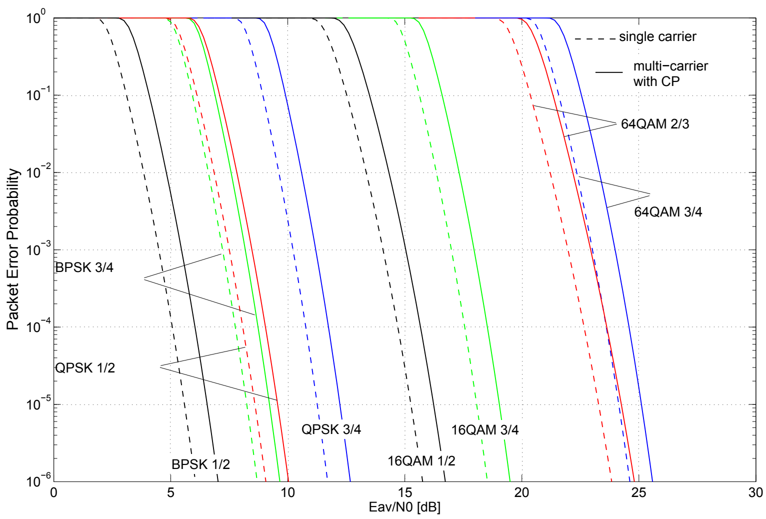

Figure 1. Theoretical PER for a 1514 byte long packet. The solid lines refer to a multi-carrier system with aCP = 80%. Dashed lines refer to a single carrier system without cyclic prefix.

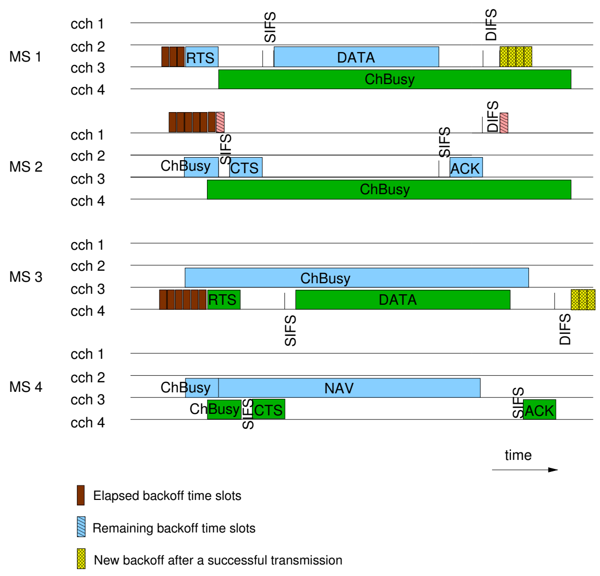

Figure 2. CSMA/CA in four cchs.

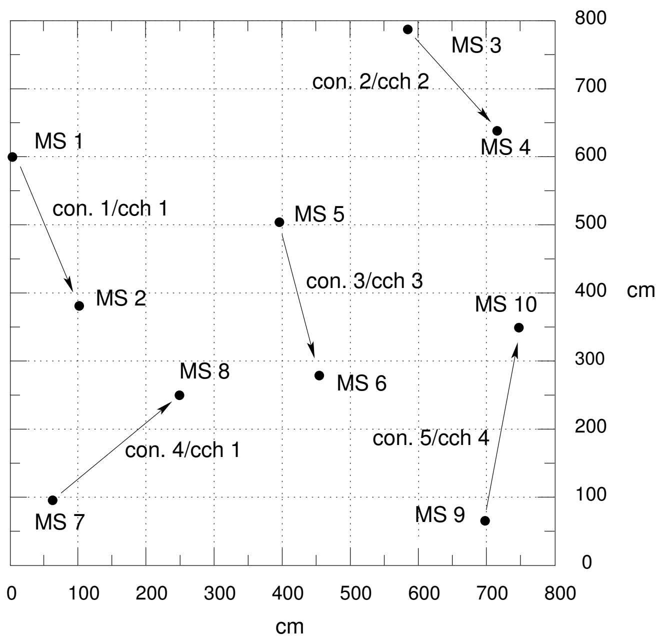

Figure 3. Simulation scenario for C-DCF performance evaluation.

2.2 Performance Evaluation

For the performance analysis of the proposed protocol, the event-driven simulation tool MACNET 2 has been developed, based on C++, the Specification and Description Language (SDL), the translation tool SDL2SPEETCL and the SDL Performance Evaluation Tool Class Library (SPEETCL).

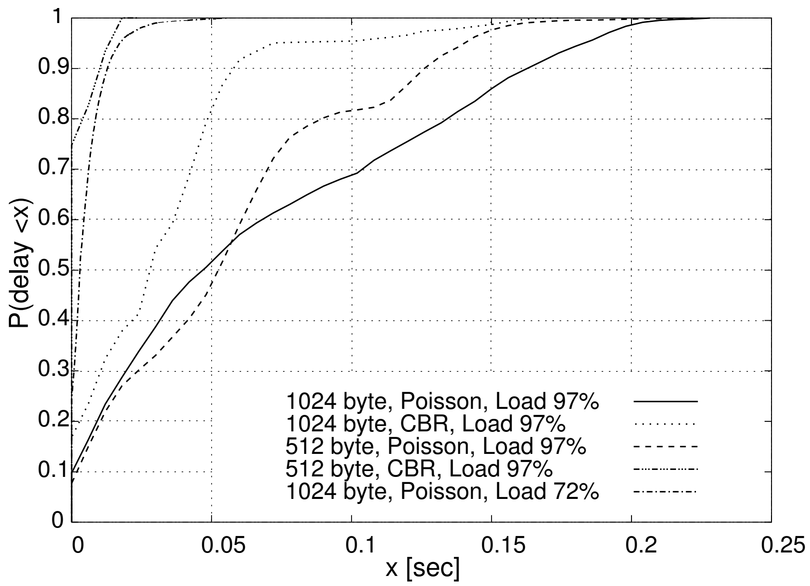

Figure 4. CDF of data packet queueing delay in the network at different load. The applied PHY-mode is QPSK 1/2 for both control and data frames.

Assuming the example scenario of Figure 3, the respective simulation results concerning the Cumulative Distribution Function (CDF) of packet delay are presented in Figure 4. It can be seen from the resulting graphs that even under high load the packet queueing delay of the system can be kept under 150msec for 87% of the transmitted data packets.

2.3 Protocol Efficiency Comparisons

In addition to the simulated performance evaluation [6], the achievable throughput has been analytically calculated [7], [8], and directly compared showing its superiority to an OFDM based system, owing mainly to the longer duration of spread data frames compared to protocol guard intervals and control frames duration. It must be noted that protocol guard intervals are a limiting factor for throughput when broader channels are considered. This makes the use of MC-CDMA inevitable for the emerging wireless systems aiming at a net throughput of 1Gbit/sec and more.

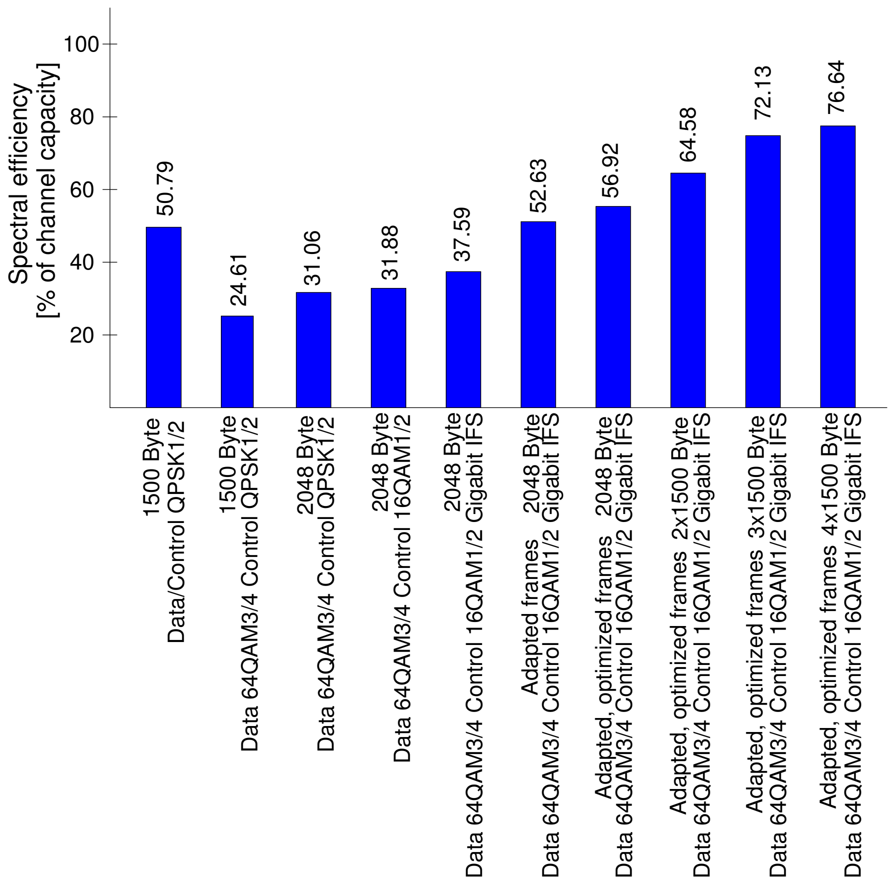

Figure 5. MAC protocol efficiency comparison for different data packet sizes and PHY modes.

In Figure 5, a MAC protocol efficiency comparison between the two modulation schemes (MC-CDMA and OFDM) is presented. The Physical layer (PHY) parameters used are those defined by the Wireless Gigabit With Advanced Multimedia support (WIGWAM) project [www.wigwam-project.de] aiming towards the design of a system for wireless data transmission with speeds up to 1 Gbit/sec. For the Home and Office WLAN networks, which are the focus of this work, the use of 100 MHz wide frequency channels at the 5.25 GHz frequency band is proposed. Further parameters of the suggested wideband channel are: symbol interval 6.8μsec, guard interval 0.4μsec, number of subcarriers 596 and subcarrier spacing 0.15625 MHz. It is apparent from the comparison that MC-CDMA system utilizes better the channel capacity.

3. TRANSMIT POWER CONTROL

Generally, efficient Transmit Power Control (TPC) is essential for a good performance of wireless networks, especially in Coded Division Multiple Access (CDMA) systems, where the capacity is limited by receiver's ability to extract the intended signal from the received one. The proposed TPC algorithm based on an improved handshake [9], that allows the exchange of channel information between the peer MSs prior to data transmission, contributes much to a good network performance, since it fast adapts to the environment conditions. The communicated information contains an interference estimate of the receiver, established with the help of MUD and the last transmission power applied by the transmitter. Transmission power fluctuations due to short term fading and small fluctuations of MAI can be controlled by taking earlier estimates into account, as well as power adjustment 2-3dB higher than necessary to achieve the required SINR value.

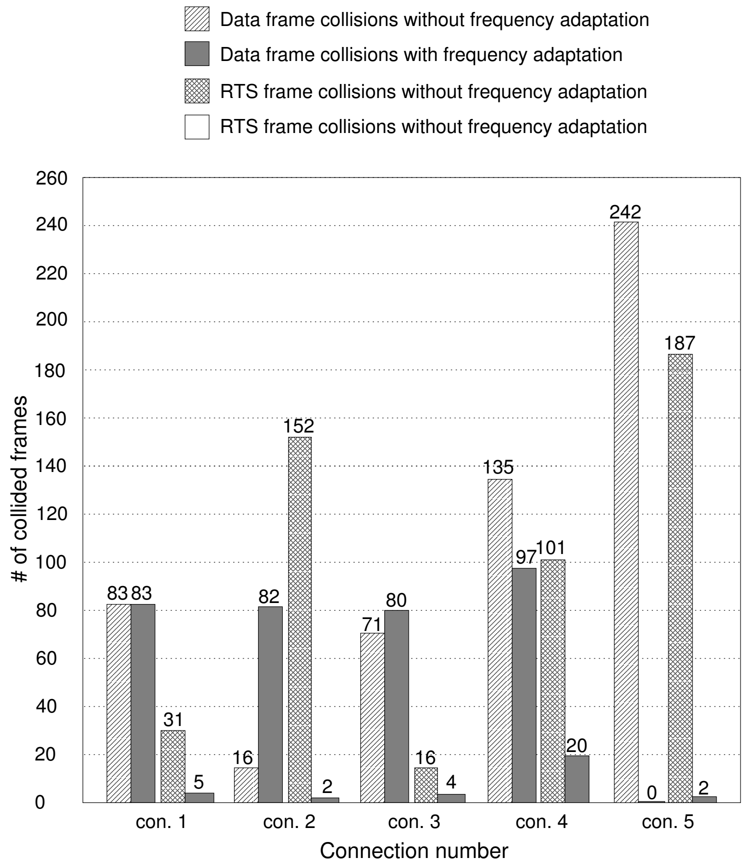

Figure 6. Amount of collided data and RTS frames with and without frequency adaptation. The offered load is 8 Mbit/sec/connection. The applied PHY mode is 64QAM 3/4 for data packets and QPSK 1/2 for control packets.

4. FREQUENCY ADAPTIVITY

Especially for wireless decentrally controlled interference limited networks, where an optimum power adjustment for all transmitting MSs is not always possible, it has been demonstrated [8] that TPC should be combined with adaptation rules, that avoid link failure due to high MAI. In order to overcome this problem, the frequency adaptation method [10] is proposed as supplement to the C-DCF protocol. According to this method, connections that suffer from high MAI due to concurrent transmissions of other MSs, are diverted to operate in other frequency channels. The diversion is performed automatically as soon as the peer entities determine (over the improved handhake) that the necessary link quality cannot be establish in the present frequency channel.

Figure 6 presents the number of collided data and Ready To Send (RTS) frames per transmitting MS, for high offered load in an example scenario. Data frame collisions with frequency adaptation are limited to the period before the steady state is achieved. The enhancement of the network's performance is substantial when frequency adaptation is applied which confirms the effectiveness of this method.

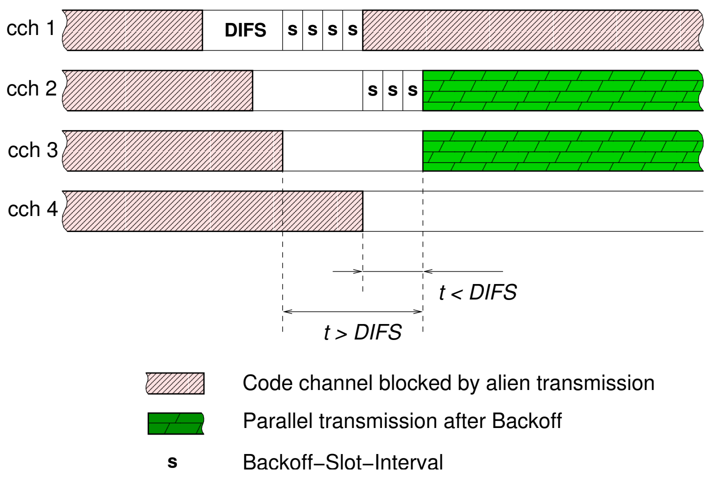

Figure 7. Backoff with parallel transmission.

5. SMART BACKOFF

The C-DCF protocol is further improved by introducing a new backoff algorithm, namely Smart Backoff [11], that optimizes the protocol for the support of a multichannel cch structure. These algorithm allows each transmitter to reselect the cch used for transmission of the next data frame, thus leading to an opportunistic cch usage. Furthermore, parallel transmission in more than one cch becomes possible. As shown in Figure 7, a MS that monitors two cchs in idle after the downcount of its backoff timer can bundle them to a broader cch. Two benefits of these backoff algorithms are the support of prioritized access to MSs (that consequently achieve lower delays), and load balancing among the different cchs.

6. CROSS LAYER OPTIMIZATION

A cross layer optimization function is further incorporated into the proposed protocol. Based on the performance of MUD, that is very dependable on the relative delays of concurrent transmissions in different cchs [8], a new parameter set for the MAC protocol is proposed, that enhances synchronization of the MSs in PHY layer, allowing the asynchronous system to operate in an isochronous mode. In order to improve synchronization, all parameter values of the protocol (SIFS, PIFS, DIFS and the slot duration used for the calculation of backoff) are defined as multiples of 4μsec, the multi-carrier symbol duration. Accordingly, the medium has a slotted structure, where transmissions are initiated only at the beginning of a slot with 4μsec duration. By using this simple arrangement, synchronizationa and accordingly performance improves substancially as the MUDs of the receivers operate at optimum performance and maximize interference suppression.

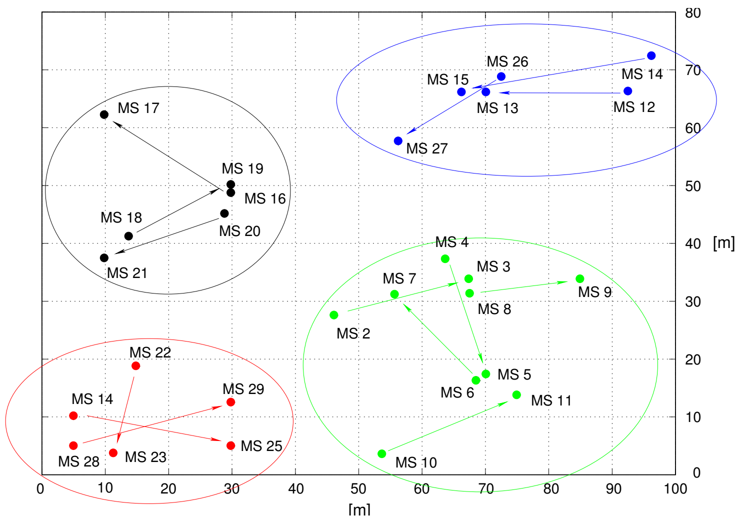

Figure 8. The simulated large area scenario for the cross layer optimization function.

Figure 8 shows the scenario which was used for setting up a comparison between the basic C-DCF protocol (mode A) and the enhanced one (mode D) that uses the above mentioned optimization in addition to TPC (chapter IV) and frequency adaptation (chapter V). As can be seen from the results in Figure 9, in form of carried traffic per source MS, the application of mode

D, not only increases the carried traffic for all MSs in the network, but improves fairness too.

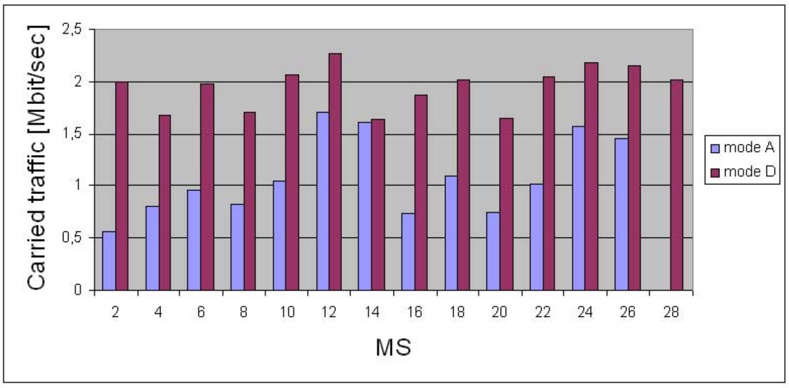

Figure 9. The carried traffic per MS, for two operating modes. The applied PHY mode is QPSK 1/2 for data frames and BPSK 1/2 for control frames.

7. MAC EXTENSIONS FOR MULTIHOP

Multihop connections are essential for W-LANs, used not only for coverage extension, but also as a method to reduce interference. C-DCF has been extended for that with the needed functionality. Multihop packets are treated as normal packets in each multihop-capable MS, and are transmited according to the rules of the C-DCF protocol to the next hop. In order to relief multihop MSs from the higher load, they are capable of Smart Backoff (chapter VI) with parallel transmissions in more than one cch. Further, an extended NAV is used, the NAV per cch and MS. According to the new NAV, each MS receiving a RTS and/or CTS, sets its NAV timer for the denoted duration of transmission, on the channel in which the control frame was received, and marks additionally the involved MS(s) as occupied. This precaution prohibits collisions in multihop scenarios, from MSs trying to transmit data packets to the same receiver, over different cchs, while it enables Smart Backoff deployment in multichannel networks.

In Figure 10, a representative multihop scenario is shown, that is used to evaluate the performance of multihop C-DCF protocol in a direct comparison with IEEE 802.11 DCF. The results of this comparison in form of the complementary CDF of end-to-end delay per multihop connection for DCF and C-DCF are presented in Figure 11 with 0.75 Mbit/sec/con Constant

Bit Rate (CBR) offered load. The difference of measured delay is for every connection at least an order of magnitude in favor of C-DCF, demonstrating the advantages of the proposed multihop protocol.

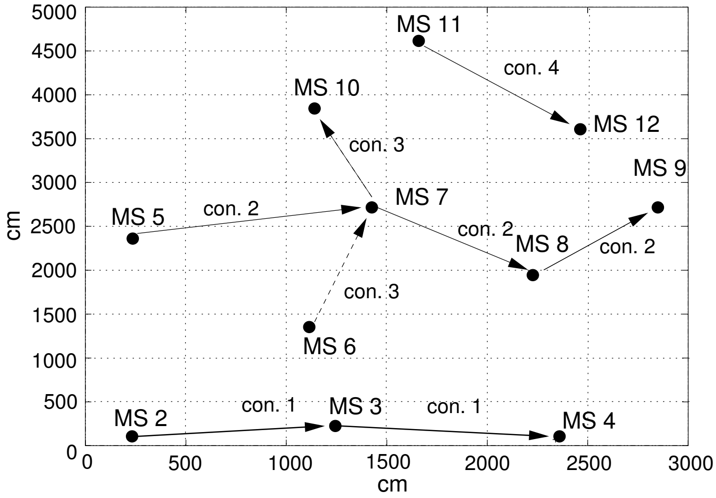

Figure 10. The simulated scenario for multihop transmissions.

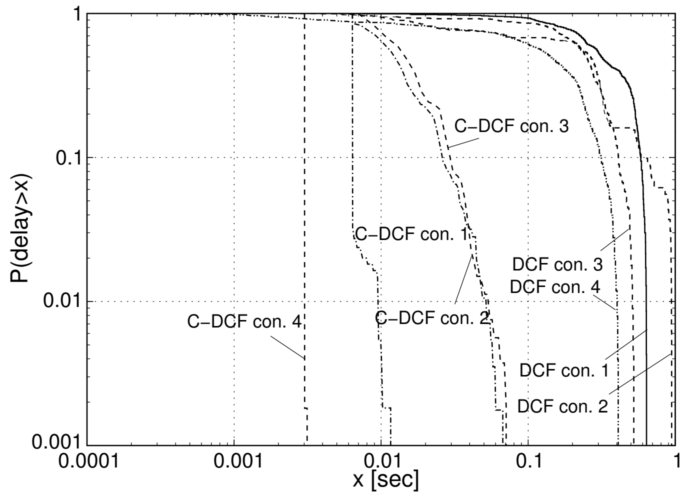

Figure 11. Complementary CDF of end-to-end delay per multihop connection for DCF and C-DCF. The offered load is CBR, 0.75 Mbit/sec/con and the applied PHY mode is QPSK 1/2 for both, control and data frames.

8. CONCLUSION

In this work a distributed WLAN MAC protocol based on MC-CDMA is presented and evaluated, suitable for next generation of WLANs. Adaptaion and optimization functions are proposed for MAC layer which in addition to the sophisticated PHY layer, incorporating a MUD, can mitigate the effects of the near-far problem, and allow WLANs to improve significantly their performance and achieve high efficiency even under heavy multiuser interference, as is typical for ad-hoc operation.

It must be noted, that many of the proposed protocols, can be applied, with some modifications, to other WLANs with multi-channel structure, which does not necessarily emerge from a spread spectrum based PHY layer.

References

[1] Std.802.11. Wireless LAN Medium Access Control (MAC) and physical layer (PHY) specifications: Wireless LAN Medium Access Control (MAC) and physical layer (PHY) specifications, Technical Specification, IEEE LAN/MAN Standard Comitee. IEEE, July 1999.

[2] N. Yee, J.-P. Linnartz and G. Fettweis. Multicarrier CDMA in Indoor Wireles Radio Networks. In Proceedings of Personal Indor Mobile Radio Communications, PIMRC 1993, S. 109–113, Yokohame, Japan, Sept 1993.

[3] K. Wang, P. Zong and Y. Bar-Ness. A reduced complexity partial sampling MMSE receiver for asynchronous MC-CDMA systems. In Proceedings of GLOBECOM 2001, 2001.

[4] S. Yi, C. Tsimenidis, O. Hinton and O. Sharif. Adaptive minimum bit error rate multi-user detection for asynchronous MC-CDMA systems in frequency selective rayligh fading channels. In Proc. IEEE PIMRC 2003, 2003.

[5] G. Orfanos, J. Habetha and W. Butsch. Error Probabilities for Radio Transmissions in MC-CDMA Wireless LANs. In Proceedings of the 65th IEEE Vehicular Technology COnference, VTC Spring 2005, Stockholm, Sweden, May 2005. ISBN 0-7803-8888-7.

[6] G. Orfanos, J. Habetha and L. Liu. A Modified IEEE 802.11 MAC Protocol for MC-CDMA. In Proceedings of the 1st ACM International Workshop on Evaluation of Wireless Ad Hoc, Sensor and Ubiquitous Networks PE-WASUN 2004, S. 52–55, Venice, Italy, Oct 2004b. ISBN 1-58113-959-4.

[7] G. Orfanos, J. Habetha and L. Liu. MC-CDMA based IEEE 802.11 Wireless LAN. In Proceedings of the Twelfth IEEE International Symposium on Modeling, Analysis, and Simulation of Computer and Telecommunication Systems, MASCOTS 2004, S. 400–405, Volendam, Holland, Oct 2004. ISBN 0-7695-2251-3.

[8] G. Orfanos. Development and Performance Evaluation of an adaptive MAC Protocol for MC-CDMA Wireless LANs with QoS Support. Ph. D. Dissertation, Aachen University, ABMT 52, p. 228, 2006. ISBN:3-86130-931-9.

[9] G. Orfanos, J. Habetha and W. Butsch. Efficient Power Control for MC-CDMA based W-LANs. In Proceedings of 11th European Wireless Conference 2005, Band 1, S. 365–371, Nicosia, Cyprus, Apr 2005. ISBN 3-8007-2886-9.

[10] G. Orfanos, B. Walke, W. Butsch and H.-J. Reumerman. An Adaptive MAC Protocol for MC-CDMA Adhoc Wireless LANs. In The 16th Annual IEEE International Symposium on Personal Indoor and Mobile Radio Communications, Berlin, Germany, Sept 2005. ISBN 3-8007-2909-1.

[11] G. Orfanos, J. Habetha and W. Butsch. A new Distributed Coordination Function Adapted to Mc-CDMA based W-LANs. In Proceedings of 11th European Wireless Conference 2005, Band 1, S. 372–377, Nicosia, Cyprus, Apr 2005. ISBN 3-8007-2886-9.

SMART CACHING IN INTERMITTENT WIRELESS BROADBAND NETWORKS

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RWTH Aachen University, Faculty 6, Department of Communication Networks

Abstract Future mobile radio networks will aim on "broadband access for all", anywhere. The performance of the radio network vitally depends on the characteristics of the transmission path between user terminal and Access Point and the degree of network coverage. In urban areas full broadband radio coverage is difficult to provide, causing - from users' perspective - a high variation of throughput capacity and making continuous broadband services hard to realize. In rural regions massive deployment costs prevent a full broadband coverage. Most of the time users have to settle for UMTS-like wide area networks. For mobile users services like video streaming which require continuous broadband connectivity this virtually results in intermittent network coverage. The frequent disruption of the broadband link and its replacement with no or only low performance connections is a problem to be dealt with.

This article introduces a new technique, called Smart Caching, that is able to mitigate variations in network performance so that non-real-time and non-interactive services' quality is improved, substantially. Smart Caching supports pre-fetching from a server and buffering data at the edge of the core network. It transmits data with extremely high speed to be buffered in the mobile terminal when it is in service range of an Access Point. This allows the provisioning of data intensive services even in the case of patchy and intermittent wireless broadband network coverage. The performance of the Smart Caching service is evaluated with a sophisticated queuing model based on the Markov Arrival Process. The performance of the new approach is discussed and dimensioning issues are outlined. The results of a comparison with a legacy network setup is presented.

1. INTRODUCTION

Nowadays ubiquitous telecommunication access is no longer a dream. Mobile users have Internet coverage wherever they are. However, the connectivity is mainly provided by wide area networks like the Universal Mobile Telecommunication System (UMTS) which offer a peak data rate of only up to 384 kbit/s. Contrary to that in so called Hot Spot zones broadband Internet is provided by wireless networks like the Worldwide Interoperability for Microwave Access (WiMAX) system which allow links with a throughput capacity of several tenth of Mbit/s. This combination results in a very heterogeneous network coverage. For mobile users it comes along with the problem that especially broadband services cannot be continuously facilitated with the necessary data rate. In densely populated areas a reliable broadband radio service is almost impossible to achieve since such wireless networks most probably will operate in frequency bands beyond 3.5 GHz. Attenuation of radio waves and shadowing due to obstacles is very high in these bands, resulting in spotty broadband radio coverage.

In rural areas the realization of broadband wireless network coverage fails due to high deployment costs. However, for intensively utilized regions like motorways at least a partial broadband coverage is achievable, e.g. by mounting access nodes at bridges or traffic signs. But all this stresses the fact that in the future for mobile users only intermittent broadband coverage will be provided.

The disruptions of the broadband network connection have to be compensated for mobile users in order to provide sophisticated data intensive services like e.g. high quality video streaming. One way to do so is the introduction of delay tolerance to broadband services. The services should be continued even when a wireless broadband network connection is currently not available. This at the first glance incompatible postulation can be fulfilled by extensive buffering techniques.

A service is only interrupted when the user detects an odd behavior. As long as e.g. the play-back of a video continues the service is not regarded as disrupted although the network connection might be broken for tens of seconds or even minutes. This obviously requires building up a stock of video data in the user terminal.

The same holds for push service of personalized (Emails, RSS feeds, ...) or commonly demanded contents (newscasts, video streams, ...). By stuffing the terminal with a huge stock of personalized information, which might be interesting for the user, it might be possible that he even do not notice connection interrupts. While he is scanning the news of his favored Internet side, which were pre-fetched in advance, no Internet connection at all is required, as all data is stored locally on the user's device. All this can be achieved by - but also requires - a massive transfer of data. Since only a certain percentage of data is really consumed and most of it is discarded unnoticed the download rates drastically increase. Large stocks of pre-fetched data are built in user terminals. This in combination with the frequent disruptions of broadband connectivity makes it of vital importance that periods of good link performance have to be optimally used. Then the throughput of the wireless system is brought to its optimum.

Therefore Smart Caching (SC) integrates two fundamental paradigms. Firstly, the optimal utilization of available network resources if broadband connectivity can be provided and, secondly, the aggregation of data reservoirs in the end device to virtually continue services even in the case of numerous connection interrupts. This concept is further elaborated in the next Section. In Section 3 related work to this concept is outlined while Section 4 discusses the organization of SC in the network. The following Sections present the performance analysis of SC. Finally a summary and a conclusion are given.

2. THE SMART CACHING CONCEPT

In order to optimize the transfer of data in heterogeneous networks a data buffer, the Smart Cache (SCache), is introduced. It is closely located to the Access Point (AP) at the edge of the wireless network as shown in Figure 1.

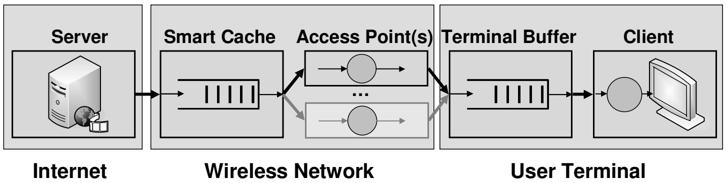

Figure 1. Architecture of Smart Caching.

Caching in this node, as described in this paper, serves for the reduction or even removal of the ”inner resistance” (resulting from flow control algorithms or from capacity constraints) of a fixed telecommunications network, as seen by a user terminal. For this purpose, data from a server is pre-fetched and stored in the SCache so that the mobile terminal can access user data with a transmission speed as high as its wireless link is able to support. Since the distance between user terminal and base station is determining the radio link data rate substantially, caching of user data may be useful - even when wireless broadband coverage without unserved spots is available to user terminals. Which impact SC can have on the data rate is shown in Figure 2. In each graph the data rate of the wireless network, the backbone network and the resulting end2end data rate is displayed. The data rate of the wireless network is of pyramid shape. When a mobile user enters the coverage of an AP the data rate is usually low as at the cell border only the lowest performance of the wireless network can be expected. But the closer the user comes to the AP the higher the data rate gets. When leaving the coverage zone the opposite takes place. The data rate decreases the larger the distance to the AP becomes. Between periods of broadband wireless network coverage there are also periods without any broadband connectivity, so that the radio data rate is equal to zero between two phases of connectivity. The percentage by which the observation interval contains periods of network connectivity is denoted by the coverage ratio (cov_ratio).

Contrary to the behavior of wireless networks the backbone data rate is relatively constant. This data rate is in the following called Backbone Limit (BL) as it prevents the wireless network from achieving the best performance. The value might vary in certain limits but it will usually not achieve the peak rates of the wireless network. In legacy networks the resulting end2end data rate is always the minimum of all involved links which in this case is the minimum of the wireless and the backbone data rate. So if no or only low performance wireless access is provided the data rate of the radio link is the limiting factor while in case of high quality wireless access the backbone limits the performance. This behavior "cuts of" a substantial part of the possible performance of the wireless network (Compare the upper graph of Figure 2).

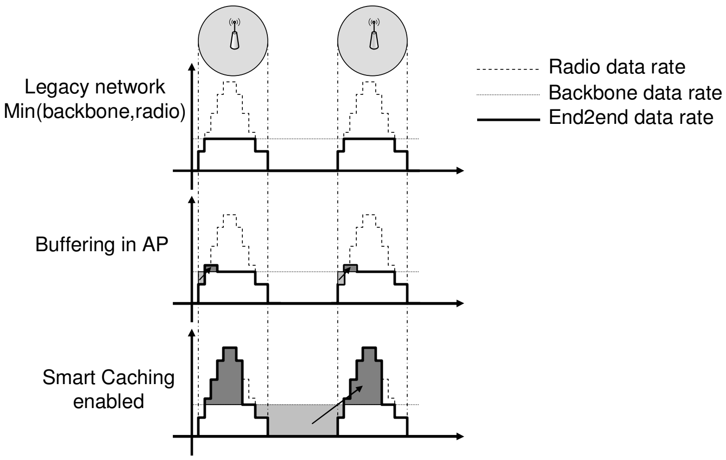

Figure 2. Impact of Smart Caching on Data Rate.

By buffering data in the AP the initial phase of low radio data rates might be used to build up a stock which then is forwarded when enough capacity in the wireless LAN gets available (Second graph of Figure 2). But this has only a minor impact on the overall performance. Only by caching packets which arrive during the gap period it is possible to gather a sufficient amount of data. This makes it possible to substantially utilize the otherwise unused capacity of the wireless network. In order to make use of the cached data it is necessary that it is stored close to the next visited AP. Since the prediction which AP is visited next is usually difficult all AP of a closer proximity have to be clustered and connected to one SCache. Then it does not matter to which AP the user connects next as long as it belongs to the cluster of the SCache which has buffered the data.

Such a buffering operation is not suitable for interactive communication services like speech and video conference owing to strict delay bounds specified for these services' data packets. But these applications may benefit from the service provided by a wide area radio network, since they are not bound to wireless broadband networks.

Non-real-time services like unidirectional video streaming, FTP, email push/pull and similar are tolerating high delay variations. SC is exploiting this property by pre-fetching contents and buffering it at the edge of the fixed backbone network, close to the AP where a mobile terminal is roaming. The buffered data is kept until the terminal comes into service range of an AP or a timer expires. As soon as a mobile terminal enters the service area of some AP the cached data is transferred at maximum possible speed to be buffered in the user terminal. The corresponding storage is called Terminal Buffer (TB). Compared to buffering of video streaming data as supported by RTP/RTCP the buffer size of the end device must be substantially larger with SC, so that even long intervals of low network performance or even service interruptions for, e.g. up to tens of seconds, can be covered by the buffered data.

From the perspective of the communication partners, the application server and the service user, the end2end connection is enriched with two buffers. One is the SCache in front of the wireless link and one is the TB directly behind the wireless link. These two buffers together create a black box as shown in Figure 3 which is fed with data from the server. This data is consumed in the end device by the user.

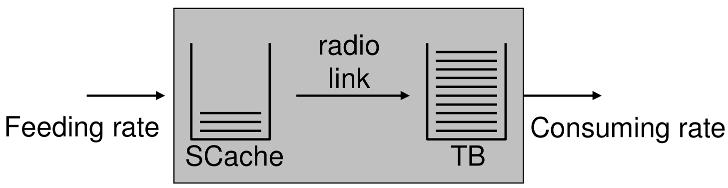

Figure 3. Buffering Concept of Smart Caching.

The task of SC is now that the data flow between both buffers is managed in such a way that the TB never gets drained. As much as possible data should be stored in the TB. But if the wireless link is disrupted the SCache absorbs the incoming data packets. As soon as enough bandwidth gets available the radio link is fully exploited and all data is transferred from the SCache to the TB. In the period of no connectivity the TB on the other hand is responsible for continuing the service so that the disruption can be hidden from the user.

In the case of video streaming this means that the stored video data is used to feed the play-back.

SC allows exploiting the actual available radio link capacity to transmit as much information as possible to the end device so that its buffer is always filled and thus be able to bridge periods of no radio service. The analysis in the Sections 5-7 proves SC supported networks allow continuous video streaming service even under intermittent network coverage with long connectivity gaps. Furthermore the advantage of the new approach over existing techniques is shown.

3. RELATED WORK

One problem which occurs if user data is temporarily stored in a buffer is that it might interfere with network protocols like TCP. As TCP is an end2end connection oriented protocol the buffering would cause massive performance degradations. Therefore it is necessary to decouple the connection into two subsections - one between server and buffer and another one between buffer and terminal. For TCP a solution, called SNOOP, was already presented in [1]. It was developed for the usage of TCP over wireless links which suffers from similar problems.

Since the mobile user continuously changes his point of attachment to the network it is necessary to keep track of his movements. At each new entered cell the data delivery between SCache and terminal has to be restarted. A SIP based protocol for this purpose was developed in the IPonAir project [2].

In [3] a store and forward principle is discussed. It focuses on regions with rudimentary network coverage which should be enhanced by an intermittent broadband connectivity. Such areas should be supplied by so called data mules which physically transport the data. E.g. a small village is connected via one wireless link for basic access and additionally a frequently operating bus piggy back transports the data between an access gateway in the next bigger city and the village. From this approach the organization of the store and forward principle can be applied for the current concept.

The idea to transfer data only in regions where high bit rates can be provided was already followed by the Infostation concept [4]. This approach also inspired the later presented highway scenario. However, in the article only general ideas and concepts are outlined while here the focus is put on the achievable improvement in system performance and network capacity.

4. ORGANIZATION AND POSITION OF THE SMART CACHE

The SCache works as a mediator between the wired and the wireless world by introducing a buffer between them so that the end2end data flow is separated into two subsections. The SCache terminates the first subsection commencing at the service provider. Concurrently it is the starting point of the second subsection which terminates in the TB.

The actual communication partners are the service provider offering e.g. a video stream and the user terminal by which the service is requested. The SCache resides in the middle between the two communication partners. For the service provider the SCache adopts the role of the TCP data flow end point. At this point the end2end connection between service provider and client is interrupted. The ingress data leaves the transport layer and moves up to a data buffer where it is cached until it finally can be forwarded to the user terminal. The first subsection exclusively consists out of wired Internet connections. Therefore TCP or similar protocols are perfectly suitable for the transfer of data in this interval. For the second subsection also TCP can be employed, but as the section basically consists only out of the wireless hop UDP like protocols with additional acknowledgement are sufficient. Nevertheless an end2end achknowledgement has to be supported to ensure the delivery of the complete contents.

Pre-fetching and buffering does only make sense if the data actually reaches its final destination, here the user terminal. Data which is pre-fetched in the SCache but does not finally arrive at the end user wastes network resources. To locate the SCache directly in an AP, like in the SNOOP attempt [1], would imply that all pre-fetched data is useless, if the user leaves the coverage range of this AP and never returns. Making use of already buffered data requires that it must be available also from other access nodes which the user possibly roams to in the near future.

If all APs of a certain area are clustered and connected to one SCache it is possible to benefit from the cached data as long as the user terminal is supplied by one of these APs. Starting a communication session in cell *A* the data stream is routed through the SCache which is responsible for the cluster of that cell and its corresponding AP. If the data rate on the wireless link decreases because the user moves away from the AP or completely leaves the cell the data transmission on the first subsection continues but, as no forward through the second subsection is possible, the data is buffered in the SCache. Entering a new cell *B* and connecting to its AP the communication session will be re-established. If the new AP is included in the same cluster as cell *A* it is sufficient to newly set up the connection between SCache and user terminal while the first subsection between service provider and SCache remains unchanged.

5. URBAN APPLICATION SCENARIO

To analyze the performance of SC and especially its ability to compensate connectivity gaps in urban environments the well-known UMTS 30.03 Manhattan scenario is chosen. All APs in the scenario are connected to and served by one SCache. The street grid consists out of 200 m sized blocks and 30 m wide streets. WiMAX APs are placed on every second crossover. They are operated at a transmission power of 100 mW. WiMAX uses different modulation and coding schemes (PHY modes) to adapt the radio transmission to the current radio link quality. Each of them is assigned with a minimum signal quality necessary to decode the information and an available data rate [5]. The path-loss between sender and receiver is given by an adapted free space model with attenuation factor of 3 (2 free space - up to 5 for indoor). Transmission power, path-loss and minimum signal level allow the mapping of a PHY mode to each terminal position in the scenario. As shown in Figure 4 the streets are divided in annulus shaped zones served by a specific PHY mode and zones without WiMAX connectivity. The network coverage ratio (only for street areas) is given by 82%.

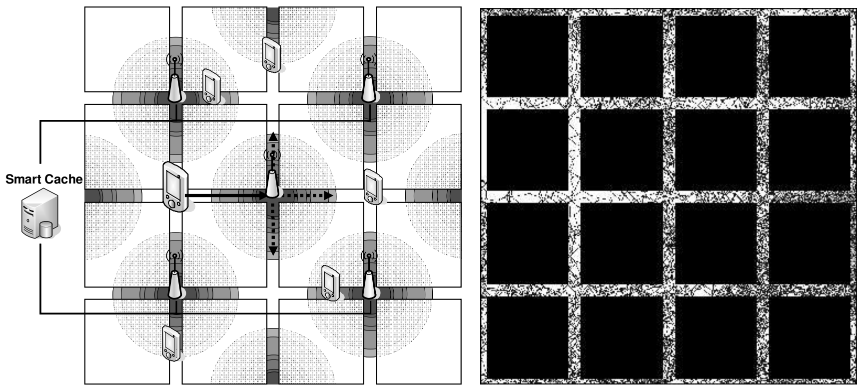

Figure 4. Urban Scenario and Mobility Simulation.

In order to get representative input parameters for the later analysis in a mobility simulation the behavior of users in such a scenario is derived. Pedestrian users are traversing the streets of the scenario with an average velocity of 0.5, 1 or 2 $\mathrm{m/s}$. There mobility is subject to an adapted Brownian motion which includes velocity updates and direction changes of up to ± 45 degree on average every 10 seconds. The outcome (or the user's pathway) of the first 10000 seconds of such a simulation is shown in Figure 4 on the right hand side. Out of much longer simulation runs reliable values for the average residence time in each PHY mode zone and the transition probability of users between different

PHY modes can be gained. Furthermore a complete Cumulated Distribution Function (CDF) for the duration of periods without network connectivity (in the following called gap period) can be derived. Since for later use a pure sample data set of the gap duration is not sufficient it is approximated with support of the EM algorithm [6]. The outcome is a phase type distribution which provides a closed form of the CDF. The result (user velocity 1 m/s) for the gap duration and its approximation are shown in Figure 5.

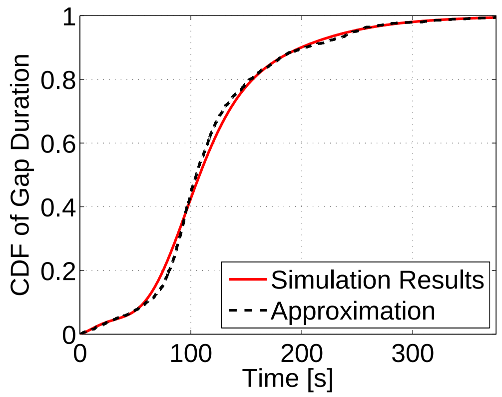

Figure 5. Gap Duration Distribution.

Furthermore the parameters of the different PHY mode zones can be employed to set up later on models for arrival and service processes (compare Section 1).

Out of the coverage ratios of each PHY mode and the corresponding data rates it is possible to calculate the average data rate an user perceives. While for backbone limited setups the performance of WiMAX is cut to a certain boundary SC enabled wireless networks allow an optimal utilization of the network resources. In Figure 6 the available average data rate depending on the BL is shown. The dotted red line displays SC enabled setups and the solid black lines legacy setups without SC. The parameter of the set of curves is the available network capacity share which is not used by other applications. If network resources are bounded to other services the wireless network can only be used partially for the data transport of the SC enabled service. This can be reflected by a reduction of the available data rate. E.g. if 80% of the resources are reserved (20% free network capacity) the data rate available for the SC service is automatically cut to one fifth.

Assuming the mobile user streams and watches a high quality video an average data rate of 3.37 Mbit/s can be assumed (MPEG4 Codec - Compare [7] p. 108). Considering the curves of Figure 6 it can be seen that the advantage of

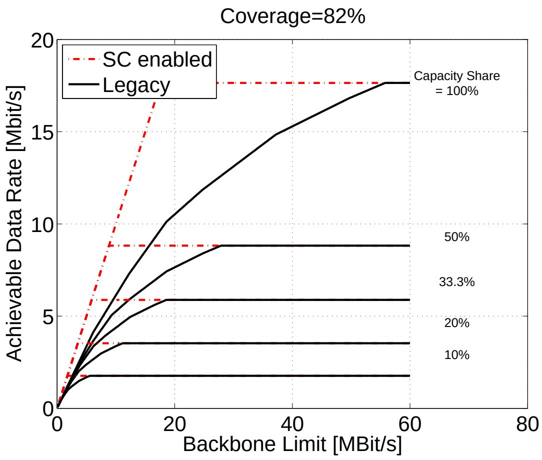

Figure 6. Data Rate depending on Backbone Limit.

SC enabled networks for such a data rate is not directly evident. Especially for high values of free capacity share the difference between the two curves (the red and the black one) is marginal. But for a capacity share of only 20% the advantage increases. While without SC the BL must be above 10 Mbit/s this value can be decreased to the actual streaming rate of 3.37 Mbit/s which is around 33%. Later on it will be shown that the advantage of SC in means of achievable data rate drastically increases if the coverage values get lower (between 20 and 50%). A capacity share of 20% implies that with full network capacity at least 5 mobile video users can be served simultaneously per cell. For approx. 0.022 km^2 covered street area per WiMAX cell this means one user per 4400 m^2, which is in a densely populated urban environment not an unlikely value. Thus, depending on the BL the service might not be realizable without SC as the average data rate would be insufficient. The number of users which can be supported by each cell of a SC enabled network is only limited by the radio resources of the wireless link. Since all data arriving from the server is buffered in the SCache until it can be forwarded to the user terminal, no data is lost and resources are optimally used. Per user video stream the utilization of the network is given by $\rho_{Video} = \frac{\lambda_{Video}}{\mu_{Net}}$. Therefore the number of supportable users (#*user*) is given by

$$\#user \cdot \rho_{Video} < 1. \tag{1}$$

This number does not only include terminals which are currently connected to the AP of a cell, but even more users which join a SC enabled service but are at the moment in a coverage gap.

For legacy networks which are at least temporarily limited by the backbone another approach has to be taken. A test user (tu) is put into the network to calculate how much users can be supported until the services of the test user fails. Due to the BL the service rate of the wireless network is diminished for the test user (Compare Figure 2) so that the utilization is given by

$$\rho_{Video}^{tu} = \frac{\lambda_{Video}}{\mu_{Net}^{BL}}. \tag{2}$$

For the later evaluation the utilization of the other users ($\#user - 1$) is not affected as only their resource consumption is relevant. And whether or not their data flow is limited by the backbone their average utilization of the wireless network does not change.

If the test user has traveled for a time x a zone without coverage and afterwards traverses a coverage zone of length y, then is has to be guaranteed that during the latter period the amount of buffered data together with the currently arriving traffic ($(x+y)\lambda_{Video}$) can be forwarded by the wireless network. Since no SCache is employed both data streams arrive directly from the server so that they are limited by the backbone (μ_{Net}^{BL}). Additionally it has to be taken into account that not all resources of the network can be used for the test user's stream but a significant portion is consumed by the other users ($(\#user - 1)\rho_{Video}$). Therefore in a borderline case the following formula holds

$$x\lambda_{Video} + y\lambda_{Video} = y\,(1 - (\#user - 1)\rho_{Video}]\,\mu_{Net}^{BL}. \tag{3}$$

For the utilization of the other users it is assumed that it is balanced throughout the observation period. This means that the other users do not enter the coverage zone in clusters but in such a way that their accumulated utilization does not vary significantly.

The relation between x and y is predefined by the coverage (cov_ratio). If the network is fully loaded the whole period y is used to transmit buffered data and

$$\frac{x}{y} = \frac{1 - cov\_ratio}{cov\_ratio}. \tag{4}$$

With the last two Equations the number of users which can be supported seen from the perspective of the test user is given by

$$\#user = 1 + \frac{1 - \frac{\rho_{Video}^{tu}}{cov\_ratio}}{\rho_{Video}}. \tag{5}$$

But the prior result holds only for the test user. Assuming a network coverage of 50% it is possible that two test users are traversing the scenario without noticing each other. Depending on the network coverage the number of possible simultaneous test users is

$$\#test\_user = \frac{1}{cov\_ratio} \tag{6}$$

so that the real number of users per cell has to be increased. Hence, Equation 5 has to be adapted to

$$\begin{aligned} \#overall\_user &= \#user + \#test\_user - 1 \\ &= \left(1 + \frac{1 - \frac{\rho^{tu}_{Video}}{cov\_ratio}}{\rho_{Video}}\right) + \left(\frac{1}{cov\_ratio}\right) - 1 \end{aligned} \tag{7}$$

One user has to be subtracted as the original test user is included in Equation 5 as well as in the number of test users in Equation 6. With this Equation the number of users depending on the BL can be derived. In Figure 7 it is done for the Manhattan scenario. Although the graph shows the results for a user velocity of 1 m/s it does not change substantially for other velocities. As coverage and all other network parameters remain unchanged the influence of the velocity is marginal and the graph holds as well for 0.5 and 2 m/s user velocity.

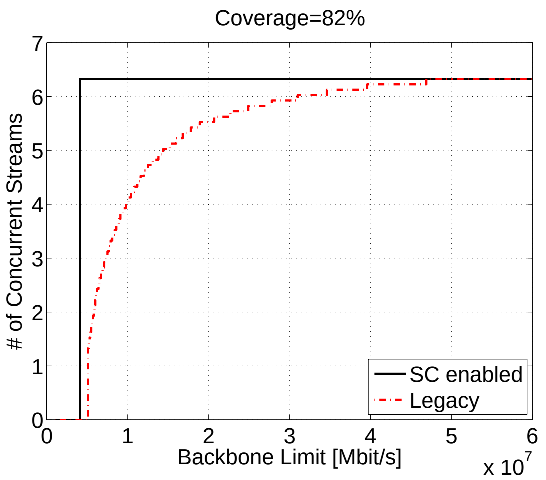

Figure 7. Number of Users depending on Backbone Limit.

In that curve it can be seen that the number of users with and without SC matches if the BL goes close to or if necessary above 55.78 Mbit/s. This is the peak net data rate of WiMAX. In this situation the existence of a BL has no influence on the network and the employment of a SCache is useless. The data is delivered fast enough through the backbone so that the wireless network can always reveal its full potential. The number of users which can be supported in this scenario is limited to 6 users. Depending on the BL this value decreases if no SC is used. Only if the BL gets below 4 Mbit/s even a SC enabled network can no longer support the video service. The reason is that the video rate, plus overhead, is more than 4 Mbit/s. Obviously even in SC enhanced networks the BL must be above the average video rate. Otherwise the backbone could not support the service and whatever performance the wireless network could provide would be of no relevance. But there still exists the problem of the waiting time the packets spend in the SCache. It is mainly sourced by the coverage gaps and in such a scenario it is almost equal to the overall delay. Therefore in the following a sophisticated queuing model is used to develop the delay.

6. MMAP/G/1 QUEUING SYSTEM

New evolutions in queuing theory allow a detailed modeling of networks and the evaluation of their performance. In [8] methods are introduced which allow the modeling of queuing systems with different incoming packet streams, each of them represented by a Markov Arrival Process (MAP). Furthermore each stream can be equipped with a separate service time distribution. The queuing delay of each arrival stream can be calculated separately. Since it can be differentiated between the arrival streams the process is called marked MAP (MMAP) and the resulting queuing system has the Kendall's notation $MMAP(i)/G(i)/1$. This allows a very detailed modeling and analysis of the investigated scenario setup. A MAP can be described by a Markov chain which allows transitions between all states. But contrary to a normal Markov system two types of state transitions can occur. The first type is just the transition between two internal states, witch is not noticeable from the outside. The other is a state transition with a concurrent creation of a packet. These transition rates of the two types are summarized in the matrices D_0 and D_1. D_0 contains simple state transitions; D_1 covers transitions with packet creations. For each arrival process i a separate set of matrices $D_{0,i}$ and $D_{1,i}$ exists. The integration of the different streams can be done by composing the Kronecker sum of the different matrices $D_{0,i}$ and $D_{1,i}$. At the end there is only one matrix D_0 and i different matrices D'_i according to the separate traffic streams. The dimension of the resulting matrices arises from the product of the size of the input matrices $D_{0,i}$.

Two types of arrival processes have to be considered. Firstly, a pure Poisson process ($D_{0,i} = -\lambda$ and $D_{1,i} = \lambda$, λ = average arrival rate) to reflect background traffic which consumes network resources that are not available for the SC supported traffic. This can be used to vary the free network capacity. And secondly, the actual video streaming process which is supported by SC has to be modeled. Already in [9] it was proposed to model variable bit rate (VBR) video traffic by using a certain number M of so called mini sources. To do so the video streaming rate is quantized in chunks of size λ_q. Each quantization step is modeled by a mini source which produces data with the rate λ_q. Depending on how much mini sources are active the accumulated data rate varies like it is done by VBR traffic. The number of active sources is steered by an underlying birth death process as shown in Figure 8.

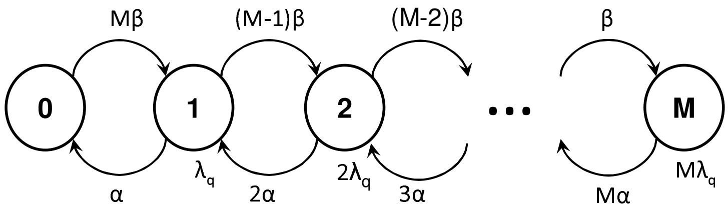

Figure 8. MMPP model for Video Streaming.

The transition rates α and β as well as λ_q can be matched to parameters of the video streaming process. In [9] the average streaming rate $E[\lambda]$, the variance $C(0)$ and the auto-covariance function $C(a,\tau)$ is used for that. With the following equations the necessary parameters can be deduced.

$$\begin{aligned} \alpha &= \left(1+\frac{E^2[\lambda]}{MC(0)}\right) \\ \beta &= a-\alpha \\ \lambda_q &= \frac{C(0)}{E[\lambda]}+\frac{E[\lambda]}{M} \end{aligned} \tag{8}$$

Actual parameters for MPEG4 coded video streaming are taken from [7]:

Table 1. Video Parameters.

	$C(0)$	a	Data Rate $[10^6/\mathrm{s}]$	M
Video (MPEG4)	0.509	0.09	3.37	20

The above modeling of the video streaming process corresponds to a Markov Modulated Poisson Process (MMPP). An underlying Markov chain steers the state changes and to each state a data rate is assigned. A MMPP consists out of two matrices Γ and Λ. The first contains the state transitions and the second the arrival rates per state. In order to translate this to a Markov Arrival Process it is enough to set

$$
\begin{aligned}
D_0 &= \Gamma - \Lambda \\
D_1 &= \Lambda
\end{aligned} \tag{9}
$$

The service process of the queuing model has to reflect the different PHY modes by which a mobile user is served. To reflect the different data rates of each PHY mode in the queuing model a hyperexponential service time distribution is taken - compare Figure 9. According to path probabilities p_i packets are processed with different service rates μ_i. Both parameters can be derived out of the probabilities that a user resides in PHY mode area i and the corresponding data rates r_i of the WiMAX system. These values are provided by the mobility simulation and the wireless network characteristics.

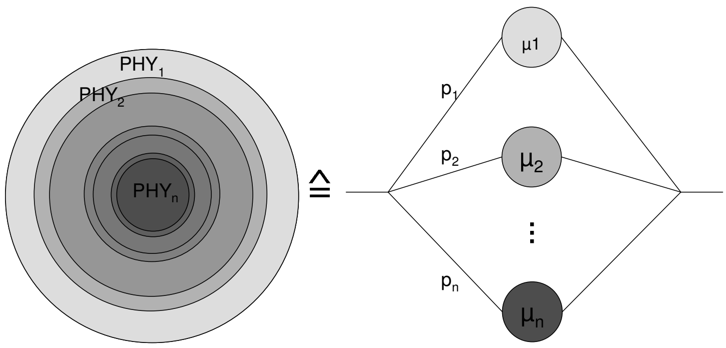

Figure 9. Service Process Modeling.

In order to fully understand the benefit of SC it is important to compare its performance with the legacy case. In order to emulate the ”cut off” behavior of legacy network setups as shown in the upper graph of Figure 2 the service rates μ_i have to be reduced accordingly. Up to now the service process does not cover areas without connectivity. As, naturally, the transmission rate is zero in such regions it is not possible to directly include it in the above model (average service time would go to infinity). If a customer traverses a coverage gap between two concurrent APs the incoming packets of the e.g. video stream are buffered in the SCache. Due to this buffering the end2end delay of the packet is drastically increased. Which influence the gap has on the actual packet delay is depicted in Figure 10. The scene is separated in three periods. In the first, denoted by x (the same as before in Equation 3), the customer leaves a coverage zone and the SCache starts the buffering so that its fill level increases. After reentering a new coverage zone the incoming traffic is still buffered, as there are still earlier arrived packets left in the cache. Since the arrival rate of new packets is smaller than the transmission rate of the air interface the buffer starts to get drained and the fill level decreases. The period until the whole buffer is emptied is called y. And finally the period z just denotes the normal operation of the wireless network (In the derivation of Equation 3 it was assumed that z is equal to zero and y includes the complete period of network coverage). Packets which arrive in period z have to wait only a short period until radio resources get available and they are transmitted.

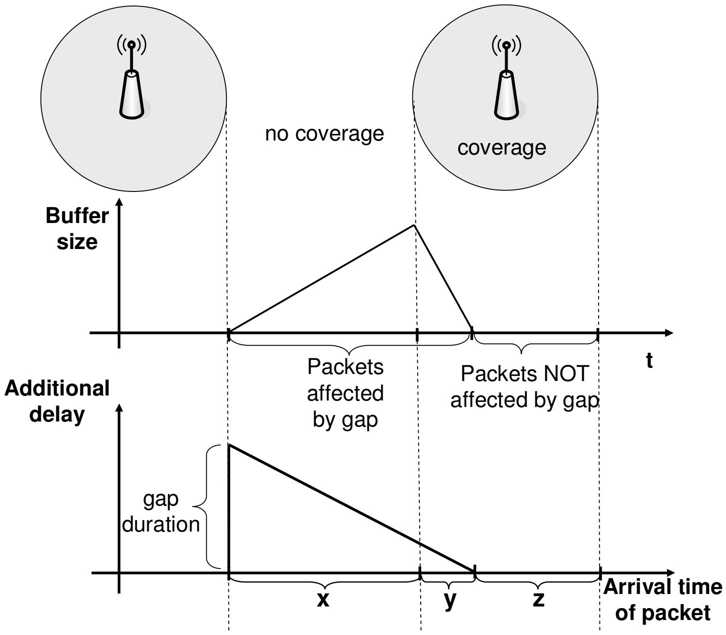

Figure 10. Influence of Coverage Gap on Buffer Size and Packet Delay.

In the lower part of Figure 10 the additional delay depending on the arrival time of the packet is shown. It is caused by the gap in the coverage. The maximum delay is suffered from the first packet which arrives after the user terminal has left the coverage zone of the first AP. It is delayed for the whole period of no coverage and is instantly transmitted when the next coverage zone is reached. Therefore the additional delay vitally depends on the duration of the gap period. The packet delay linearly decreases until the point is reached where the complete SCache is drained and the operation migrates to the normal behavior. For later evaluations it is necessary to determine the ratio of packets which are affected by the gap and the portion which is transferred during the normal operation period. Clearly the first portion is given by $(x+y)/(x+y+z)$. The prior defined Equation 3 clearly still holds so that together with the fact that the coverage ratio is given by the quotient out of x and the overall size $x+y+z$ it can be concluded that

$$\frac{x+y}{x+y+z} = (1 - cov\_ratio)\left(1 + \frac{\rho^{tu}_{Video}}{1 - (\#user - 1)\rho_{Video} - \rho^{tu}_{Video}}\right) \tag{10}$$

The additional delay the packets perceive due to the gaps in the connectivity is equally distributed between gap duration and zero. Since this delay is independent from the waiting time in the queuing model both values can be summed up. For their probability densities this implies a convolution. Since the outcome of the $MMAP/G/1$ queuing system's analysis is the Laplace Stieltjes Transform (LST) of the waiting time distribution it is natural to perform the convolution by a simple multiplication of the LSTs. Therefore it is also necessary to get a closed form expression of the LST of the gap duration. But as packets during the period z out of Figure 10 are not affected by the gap it is necessary to separate the distribution of the additional delay in two parts. With the probability $z/(x+y+z)$ the additional delay is zero and with the probability $(x+y)/(x+y+z)$ an additional delay exists. The first part can be reflected by a Dirac impulse in the origin $\delta(t)$. For the second part the probability density function (pdf) of the additional delay $pdf(t)$ is required. $pdf(t)$ and the resulting pdf for all packets are given in Equation 11.

$$\begin{aligned} pdf(t) &= \int_t^\infty \frac{p(x)}{E[x]}dx = \frac{1}{E[t]}(1 - CDF(t)) \\ pdf_d(delay = t) &= \frac{z}{x+y+z}\delta(t) + \frac{x+y}{x+y+z}pdf(t) \end{aligned} \tag{11}$$

What still is missing for a final solution of the packet waiting time is the waiting time expression of the $MMAP/G/1$ system. The LST of the waiting time distribution per packet type i is given by

$$\begin{aligned} W_k(s) &= \frac{V(s)D_k}{\lambda_k} \\ V(s) &= (1-\rho)s\mathbf{g}(sI + D_0 + D(s))^{-1} \\ D(s) &= \sum_k D_k(s) = \sum_k D_k H_k(s) = \sum_k \int_0^\infty e^{-st} D_k(t)dt \end{aligned} \tag{12}$$

where λ_i is the arrival rate of type i packets and ρ is the overall utilization of the system. The vector $\mathbf{g}$ is the stationary vector of the matrix Q ($\mathbf{g}Q = \mathbf{0}$ and $\mathbf{g1} = 1$) which is given by the recursive formula

$$Q = D_0 + \int_0^\infty D(x)e^{Qx}dx \tag{13}$$

$D(x)$ is similar defined to $D(s)$ but instead of using the LST the probability densities has to be used. The matrix Q can be iteratively calculated by starting with $Q = D_0$ and continuously substituting it in Equation 13 until the differences between concurrent results gets small enough.

With all this preparation it is possible to derive the CDF of the video streaming packet waiting time as depicted in Figure 11. The scenario is still the urban environment as introduced above and three parameters are varied: The user velocity v, SC enabled or backbone limited setup, and the number of supported users per cell. The first (solid-black) curve shows the packet waiting time for a user velocity of 1 m/s, enabled SC support, and just one user per cell. 95% of the packets have a waiting time smaller than 100 s.

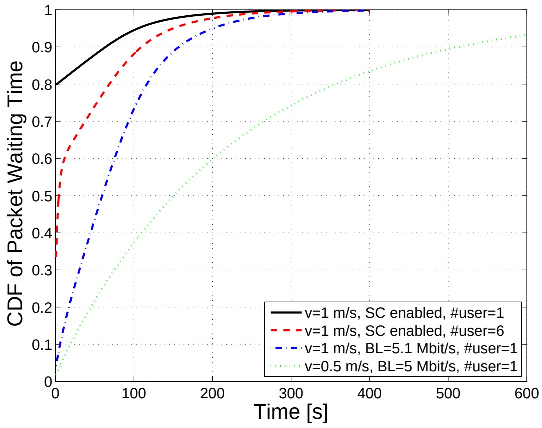

Figure 11. Packet Waiting Time in Smart Caching enabled Urban Scenario.

But still even for the SC enabled network setup the problem exists how the delay can be handled. For applications like video streaming the delay can be compensated by extensively buffering video data in the end device. If data storage is provided in the end device (namely the TB) the play-back of the video can continue when the mobile user leaves the zone of wireless network connectivity - as long as there is still buffered video data at hand. The major questions are, how big has the TB to be, and secondly, what is the required initial fill level before the play-back of the video can start? A reference value for the initial fill level and the TB size could be the 95 percentile of the packet waiting time. It has to be guaranteed that a new packet arrives early enough to continue the video play-back before the SCache is drained. If an amount of video data is stored in the end device, which corresponds to the 95 percentile of the CDF, it means that with 95% probability the next packet arrives within that time limit. This implies that in the first case 100 s of video data have to be accumulated in a reservoir in the TB.

If the maximum number of users is served by the network the 95 percentile further increases so that initially more data has to be buffered. The situation gets even worth if SC is not employed and the BL is set to the minimal possible value of 5.1 $\mathrm{Mbit/s}$. In that case it has to be mentioned that almost all packets have to wait and that the 95 percentile is around 200 s. The biggest impact in the urban scenario has the average user velocity. If the velocity is halved the 95 percentile jumps above 600 s. The BL was set again to the minimum possible value of 5 $\mathrm{Mbit/s}$ for that experiment. The discrepancy between the two minimum values of the scenario with a user velocity of 0.5 $\mathrm{m/s}$ and respectively 1 $\mathrm{m/s}$ results from variations in the mobility simulations and is not inherent to the concept of SC.

The initial fill level of the buffer is a substantial parameter for SC but it has only an indirect impact on the user. It does not matter how much video data has to be stored but of more significance is it how long the download of this amount of data takes. For example if the user starts the video streaming while he is close to an AP, and furthermore alone in the cell, it might be possible that he can use the full network capacity to download the initial fill level within seconds. On the other side the user might be at the edge of the cell and it is crowded with other users, then the download might take a long while. A worst case scenario is that at the point in time the user requests a video stream currently no network coverage at all is provided. Then the download has to be postponed until a connection is established. In Figure 12 the time which is necessary to bring the buffer to its initial fill size is depicted. For the transmission rate the average network rate is taken. If more than one user is supported then the network capacity is shared between the users. But it is taken into account that of x users only x/cov_ratio users are usually within a cell so that the capacity share per user is bigger than $1/x$ of the overall capacity.

For users moving with an average velocity of 2 $\mathrm{m/s}$ and one user per cell the fill time is always below 100 seconds even if no SC is employed and the BL gets to a minimum of 5 $\mathrm{Mbit/s}$. But with SC the fill time is lower 10 seconds which means that a user does not have to wait long between start of the streaming and beginning of the play-back. If the cell is fully loaded with 6 users even with SC a fill time of 70 seconds is necessary.

With lower velocities of 1 $\mathrm{m/s}$ or 0.5 $\mathrm{m/s}$ the situation gets worse but due to SC the fill times can be limited to values around 100 s despite of a fully loaded system (6 user) and low user velocity (0.5 $\mathrm{m/s}$). With a legacy network setup such values get much worse so that the benefit of SC is clearly shown.

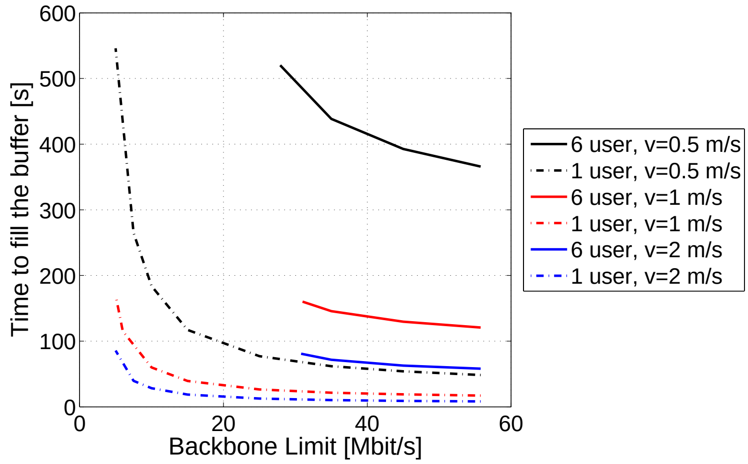

Figure 12. Download Time for Initial Buffer Fill.

7. MOTORWAY APPLICATION SCENARIO

To further illustrate the applicability of SC and its capability to increase performance in intermittent broadband networks a second setup is investigated. In this motorway scenario users go by car and access again a high quality video stream. Due to the massive deployment costs not the complete motorway is covered with broadband wireless access but only in certain intervals APs are mounted (Compare Figure 13). The AP sites are chosen depending on their infrastructure. To provide power supply and backbone connectivity usually places like bridges or traffic gantries which cross the motorway are the best places. While moving from AP site to AP site data is downloaded and stored in the TB when network coverage is provided - hence, close to the node. This data is then consumed during the idle periods between two AP. In such phases the streaming of data is continued between video sever and SCache.

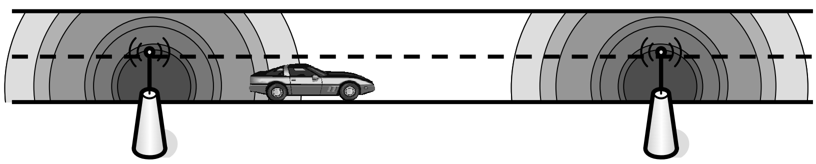

Figure 13. Motorway Scenario.

Table 2. Motorway Scenario Parameters.

Car Velocity	Gap Size
$f_v(v)$	$f_l(l)$
Normally distributed	Normally distributed
$\mu = 104km/h$	$\mu = 5, 10, 15km$
$\sigma = 12.5km/h$	$\sigma = 0.2\mu$
	Coverage=$71, 35, 24\%$

In this scenario setup it can be assumed that the attenuation between sender and receiver is reduced to a minimum. Therefore the path-loss is derived by the *D*1 Line-of-Sight model taken from [10].

$$Path - loss = 21.5 \cdot log_{10}(distance) + 44.6 \tag{14}$$

The average distance between the APs is set to 5, 10, and 15 km. The probability density of the average car velocity is taken from [11] and corresponds to a normal distribution. The density of the gap distance between two concurrent coverage zones is as well normally distributed. All parameters are listed in Table 2.

From the distributions of the car velocity and the gap size the density of the duration of a gap period can be derived by applying the following formula.

$$f_t(t) = \int_0^\infty v f_l(vt)_v(v) dv \tag{15}$$

With all that the number of user which can be supported by one cell can be derived. Since the coverage ratio differs with changing AP distance there are different curves in Figure 14. Important to notice is that the maximum number of users which is determined by Equation 1 is not influenced by the coverage. Thus, by employing SC video streaming with several users can be supported even if the distance between APs increases and the coverage reduces to values of 24%. If no SC is employed the BL has a significant impact on the number of users. With reduced coverage the number of users decreases as well. Furthermore the limit at which no service can be provided anymore increases from 5.8 Mbit/s for 71% coverage to 27.8 Mbit/s for only 24% coverage. At least the later it a value which is hard to achieve in the Internet end2end which means that not even for one single user the video service can be provided if the coverage decreases below 24%.

But similar to the evaluation of the urban scenario setup here as well the packet delay has to be considered. The question is how long it takes until the required initial fill level of the TB is achieved. By using the 95 percentile of the packet waiting time in Figure 15 it is shown what download times are

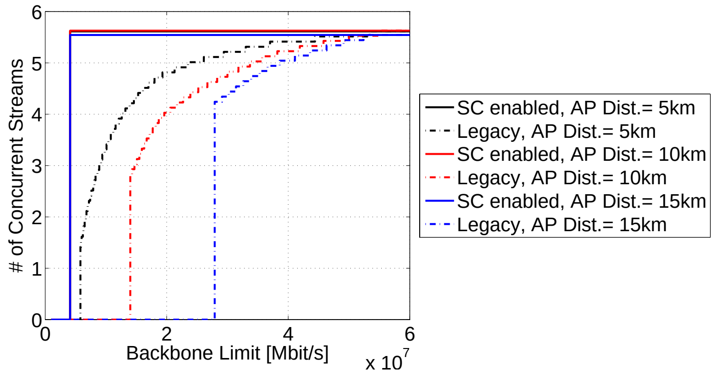

Figure 14. Number of Users in Motorway Scenario.

necessary. The parameters are the BL, the AP distance, and the number of users per cell. For a BL of more than 55 Mbit/s the values correspond to the SC enabled network setup. For the analysis the download rate is set to the average wireless network throughput. An AP distance of 5 km and only one user the initial fill level can be reached in less than 15 seconds. But even for a legacy network setup the time can be kept below 20 seconds up to a limit of 20 Mbit/s. But with a lower BL the duration fast increase to values of more than 60 seconds.

If more users are supported the download time is still low but already for higher BL values the service can no longer supported. If 5 users are using one AP cell simultaneously the BL must be above 20 Mbit/s.

For longer AP distances and therefore lower coverage ratios the durations even double. For 15 km distance and 5 users the fill time is almost constant at 100 seconds and the BL must not go below 40 Mbit/s. Although in this case with SC the delay time cannot substantially reduced the service can be provided even for BLs of around 5 Mbit/s. In legacy networks a BL of 40 Mbit/s would be required, which is very unlikely.

Due to the chosen path-loss model the diameter of the WiMAX cell results to 3.5 km. With an average car velocity of 104 km/h one cell passage lasts around 122 seconds. So in all situations at least one cell passage is enough to reach the initial TB fill level and starting afterwards the play-back of the video.

8. SUMMARY AND CONCLUSION

Smart Caching as a method for supporting intermittent wireless broadband networks and handling connectivity disruptions is introduced. The approach allows the optimization of throughput capacity in such wireless networks. Ad-

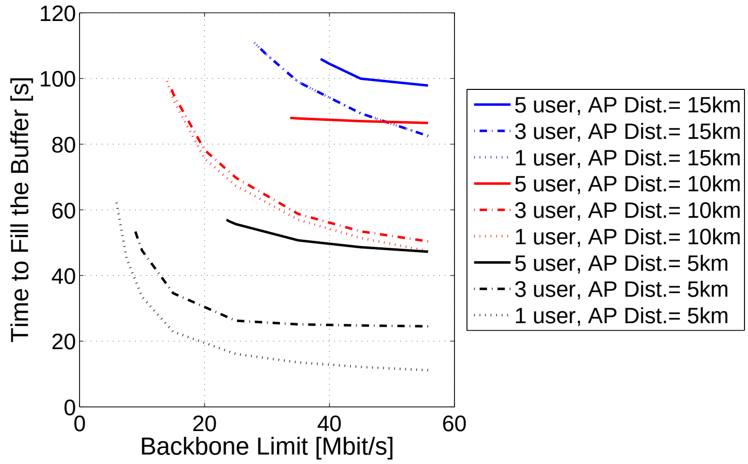

Figure 15. Download Time for Initial Buffer Fill (Motorway Scenario).

ditionally it supports and elaborates the extensive buffering of service data in the end user device. This allows the virtual continuation of broadband services even with disrupted network coverage.

Two application scenarios are presented - a pedestrian user in an urban environment and a vehicular user on a motorway. In both scenarios the applicability of Smart Caching is proven. The analysis with the support of a sophisticated queuing model allows a very accurate dimensioning of the Terminal Buffer size. Furthermore the requirements of an initial fill level for the Terminal Buffer is derived.

The comparison between legacy network setups and the Smart Caching enabled system shows the advantage of the new approach. While in nowadays networks service could not be supported the employment of Smart Caching makes it possible to offer broadband services even under very patchy network coverage. Although also legacy setups might support the same services the requirements in means of Backbone Limit and initial delay are unacceptable for modern radio telecommunication networks. All this proves that Smart Caching is feasible solution to circumvent the problems that occur in wireless networks with patchy coverage.

ACKNOWLEDGMENT

I would like to express my special gratitude to Prof. Walke for the time and effort he spent in his guidance and support of my work. He inspired me in my concept of Smart Caching from the very beginning and made several beneficial

contributions to its enhancements. I highly appreciate to have the opportunity to continue my studies and will hopefully finalize them soon with my PhD thesis.

References

[1] H. Balakrishnan, S. Seshan, E. Amir, and R. Katz, "Improving TCP/IP performance over wireless networks," in *Proceedings of the 1st annual international conference on Mobile computing and networking*, 1995, p. 11.

[2] I. Herwono, S. Göbbels, J. Sachs, and R. Keller, "Evaluation of Mobility-Aware Personalized Services in Wireless Broadband Hotspots," in *Proceedings Communication Networks and Distributed Systems Modeling and Simulation 2004*, vol. 1, SCS. San Diego, California, USA: The Society for Modeling and Simulation International, Jan 2004.

[3] M. Demmer *et al.*, "Implementing Delay Tolerant Networking," Intel Research, Berkeley, Tech. Rep. IRB-TR-04-020, Dec. 2004.

[4] R. Yates *et al.*, "Challenges in low-cost wireless data transmission," *Signal Processing Magazine, IEEE*, vol. 17, no. 3, pp. 93–102, 2000.

[5] C. Hoymann, "Analysis and performance evaluation of the OFDM-based metropolitan area network IEEE 802.16," *Computer Networks*, vol. 49, no. 3, pp. 341–363, 2005.

[6] S. Asmussen *et al.*, "Fitting phase-type distributions via the EM algorithm," *Scandinavian Journal of Statistics*, vol. 23, no. 4, pp. 419–441, 1996.

[7] P. Seeling *et al.*, *Video Traces for Network Performance Evaluation - A Comprehensive Overview and Guide on Video Traces and Their Utilization in Networking Research*. Springer, 2007.

[8] T. Takine, "Queue Length Distribution in a FIFO Single-Server Queue with Multiple Arrival Streams Having Different Service Time Distributions," *Queueing Systems*, vol. 39, no. 4, pp. 349–375, 2001.

[9] B. Maglaris *et al.*, "Performance models of statistical multiplexing in packet video communications," *IEEE Transactions on Communications*, vol. 36, no. 7, pp. 834–844, 1988.

[10] S. Baum *et al.*, "Final Report on Link Level and System Level Channel Models," Winner Project, Tech. Rep. D 5.4, 2005.

[11] C.-H. Rokitansky, Ed., *Validation of the Mobility Model of SIMCO2 with Dutch Motorway Measurements*. Prometheus Workshop on Simulation, 1992.

SELF-X FOR MOBILE NETWORKS: IS THERE A NEED FOR AUTONOMOUS MANAGEMENT?

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Abstract: This paper addresses a new paradigm of network management applying the "Self-X" principle. Being one main enabler for cognitive radio, Self-X has experienced noticeable attention within the recent years. After a general introduction what Self-X is about, its necessity for future network operation is pointed out. One main driver in this process is the operators' need for decreasing costs in general and decreasing OPEX in particular, while at the same time increasing the user's service experience. Explaining the money flow and service chain model, it is pointed out why we need Self-X and who is supposed to be in charge of coming up with appropriate technical solutions.

1. INTRODUCTION

Within the development of mobile radio, the changing of one predominant radio technology to the next one is a smooth process. This entails periods of parallel deployments that challenge the operators in terms of workforce load and technical complexity for the role out, maintenance and operation. Within this paper, it is discussed in how far Self-X based management is a potential solution and whether it is required and applicable.

Section 2 provides an introduction to Self-X which is an integral part of cognitive radio and thus facing lots of attention within recent years. Section 3 discloses operator requirements on Self-X. Section 4 gives an answer to the questions, which stakeholder is required to provide the technical solutions on Self-X related problems. The paper concludes with an overview on ongoing research and standardization activities dealing with Self-X.

2. SELF-X AS VITAL ENABLER WITHIN COGNITIVE RADIO

2.1 The Cognitive Radio Life Cycle

Radio (CR) [1] generally can be seen as another evolutionary step within Software (Defined) Radio (SDR). While SDR mainly focuses on Phy aspects and potential hardware implementations of transceivers, CR is far more sophisticated and is used as new paradigm for wireless communication. Any network entity thereby tries to avoid interference by changing its transmission or reception parameters, hence increasing spectral efficiently.

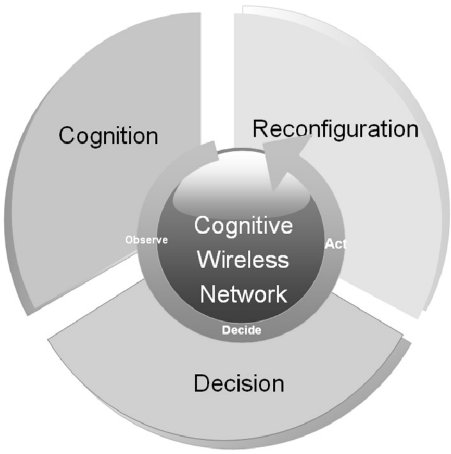

Figure 1. Cognitive Radio Life Cycle [2].

The CR life cycle is illustrated in Figure 1: Initial actions focus on observation of environmental conditions. Since the CR does not exclusively analyse the radio environment but incorporates network- and service environment as well, context awareness is achieved for which this phase is referred to as "cognition" phase. Sophisticated algorithms, in the following referred to as CR inherent intelligence, are put into place to derive mutually dependencies of all acquired observation input parameters. Applying dedicated policies, the CR performs weighting of context information and such is able to take autonomously controlled decisions. Since the CR life cycle is continuously being applied, decisions are non-static but can be adapted within each run in order to ensure dynamic optimization. The last phase finally is characterized by actions to be taken such as transmission mode (re-)selection or resource assignments. Policy enforcement hence finds its execution in terms of reconfiguration actions triggered by the CR.

The interplay of CR entities leads to the notion of a "Cognitive Network". According to [3], a "cognitive network generally addresses the future network being able to sense the radio environment (sensing the radio context, service context, location context and user context), automatic reasoning (interpreting the radio environment), self-actuating (reacting to the changes), self-tuning (tuning the radio and implementation parameters) and self-healing (fault management). The involved functional entities will distribute themselves over the radio subsystem, O&M subsystem and switching subsystem of the future telecommunication network. In order to support scalability, feasibility, integrability and extensibility, a certain amount of open interfaces and P2P signalling can be necessary. The primary goal is to increase the spectrum efficiency and decrease the CAPEX (CAPital EXpenditure) and OPEX (OPerational EXpenditure)."

2.2 Self-X Definition

The previous definition of a cognitive network introduces some aspects of autonomous management (self-actuating, self-tuning, self-healing). In fact, there are many more such as self-planning, self-configuration, self-optimization, self-maintenance, etc. The notation "Self-X" is used as common genus to subsume the diversity of autonomous entity and network operation and their specification. Accordingly, if Self-X is meant to address all possible technical functions enabling the network to manage itself in an autonomous way, the goal is to improve quality and operational efficiency by exactly this autonomous functionality.

The definition of Self-X is closely related to the realization of a cognitive network. While in principle the CR life cycle could also be realized based on human/manual interactions, its actual complexity does not permit non-autonomous implementations. Self-X hence is a substantial inherent part of cognitive radio, serving as one of its key enabling schemes.

2.3 Need for Self-X

The enormous speed with which mobile radio is developing necessitates Self-X enabled automatization. Basically, there are four key challenges that need to be satisfied:

1) Cost pressure
High cost pressure requires improvement of operational efficiency.

2) Complexity & Heterogeneity
Complexity and heterogeneity of radio (access) networks is dramatically increasing and cannot be counterbalanced by manual interception only.
3) Usability
Usability of future wireless access solutions must be improved ("plug & play").
4) Time to Market
Introduction and deployment of new wireless services and systems need to be accelerated (time to market).

Similar to the Cognitive Radio Life Cycle, Self-X, which is seen as one answer to the above challenges, is considered to be an iterative, self-repeating process, see Figure 2. An initial parameter set thereby is constantly monitored with respect to its impact on the network operation. The effectiveness then is checked against the current parameter set in order to derive new optimised settings that can be applied henceforth. The general principle of Self-X based management basically is technique agnostic and hence applicable to any wireless technology.

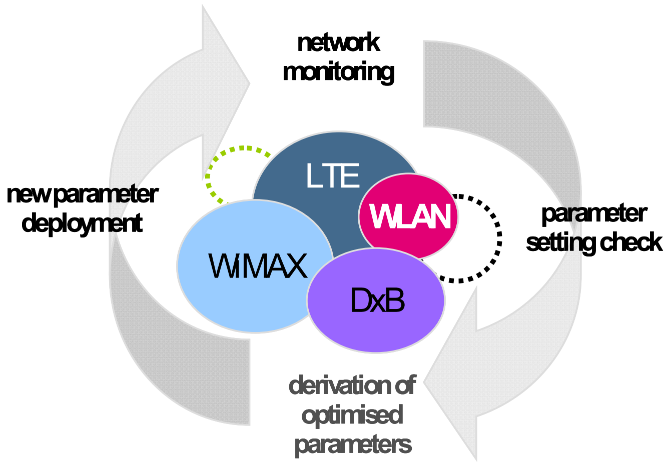

Figure 2. Self-X based parameter optimization.

3. OPERATOR REQUIREMENTS ON SELF-X

With the development of 4G systems, mobile communications will enter another era of radio based information exchange. Wireless systems enter the broadband market and thus become attractive as carrier technology for formerly wireline agnostic applications and services. Operators face two parallel extraordinary challenges due to this development:

First, it is expected that legacy carrier technologies will continue to co-exist. T-Mobile predicts UMTS to be in operation until 2015 [4]. GSM as cheap and robust technology for basic speech services is even expected to operate until 2020 [4] thanks to its worldwide successful launch with more than 3 billion users to date and continuously lasting construction in emerging markets such as China. Apart from 2G/3G mobile communication systems, further wireless connectivity options such as WLANs have been established as user accepted solutions for dedicated scenarios. Operators have had high investments in all these infrastructure compounds that start paying off for which a hastily roll-out denies itself from an economic point of view.

Second, the launch of 4G systems such as LTE results in additional efforts for the operators. Apart from roll-out implications that are hard to predict, it can be expected that operation demands of the new systems will be order of magnitudes higher due to their increased complexity. Considering that current 2G/3G systems in Germany even today offer an enormous amount of control parameters (up to several millions!), it is clear that this number will even be outperformed by LTE installations. Thereby, additional complexity due to potential launch of Home Node B (HNB) installations, also known as femtocells, is not even considered.

It is therefore obvious that both, continuation of legacy system operation as well as launch and operation of highly complex new 4G techniques require automatization support being provided by Self-X. Figure 3 quantifies the OPEX dominated cost structure of wireless communication: Taking financial figures of T-Mobile from 2007 [5], the revenue was reported to 33bn €. Considering a CAPEX of 3.3bn € and a profit of 5bn €, the remaining OPEX amounts to 25.7bn € composing a percental share of 75%. Hence, offering mobile services obviously is an OPEX driven business. Accordingly, operators prioritize optimization strategies considering OPEX reductions.

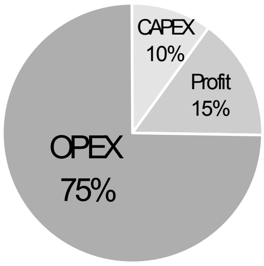

Figure 3. Mobile services provisioning is an OPEX driven business.

The Next Generation Mobile Networks (NGMN) Alliance [5] serves as operator organ that was founded in 2006 in order to provide a coherent vision for the future mobile network technology evolution. Representing 18 mobile network operators including global players such as T-Mobile, Vodafone, China Mobile, France Telecom, NTT DOCOMO, Telecom Italia and Telefonica, NGMN can be seen as the single operators' voice to express requirements and expectations with respect to B3G. The NGMN work programme is executed in working groups, each of which with a well defined scope and dedicated objectives. NGMN Project 12 focuses on "Self-Organizing Networks" (SON), thus the Self-X dogma is well anchored within NGMN.

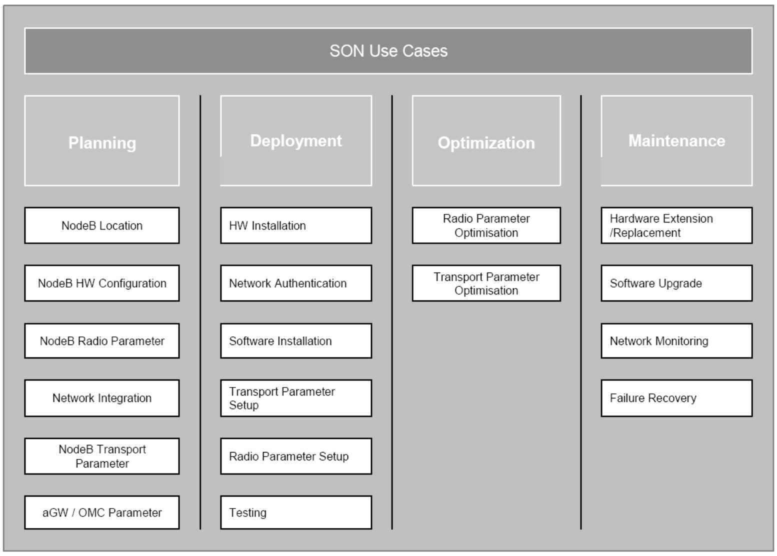

Figure 4. NGMN SON categories and sub-groups [6].

OPEX reduction (among others) is addressed within four different SON clusters which are “Planning”, “Deployment”, “Optimization” and “Maintenance”. The different clusters incorporate sub-groups of SON related use cases, see Figure 4 [6]. In total, NGMN currently proposes 32 concrete use cases serving as examples on how to implement SON functionality.

In detail, the Planning related use cases cover [7]

- [P01] NodeBLocation
- [P02] NodeB Hardware
- [P03] Automatic Generation of Radio Parameters
- [P04] Planning of transport parameters of a new eNodeB
- [P05] Planning of security Node, aGW and OMC

For the Deployment optimization, the following uses cases have been chosen to serve as a reference:

- [D01] Hardware Installation
- [D02] Network authentication
- [D03] Software Installation
- [D04] Transport Parameter Setup
- [D05] Radio Parameter Setup
- [D06] Testing

For the Optimization cluster, NGMN distinguished between use cases related to optimization of radio parameters and optimization of transport parameters:

- [O01] Radio Parameter Optimization: Neighbour cell list optimization
- [O02] Radio Parameter Optimization: Interference Control
- [O03] Radio Parameter Optimization: HO parameterization optimization
- [O04] Radio Parameter Optimization: QoS related parameter optimization
- [O05] Radio Parameter Optimization: Optimization Scenarios with Home BTS/Pico BTS
- [O06] Transport Parameter Optimization: Routing Optimization
- [O07] Transport Parameter Optimization: Optimization Scenarios with Home BTS/Pico BTS
- [O08] Reduction of Energy Consumption

The last major category considers Maintenance use cases

- [Ops01] Hardware / Capacity extension
- [Ops02] Autonomous Inventory
- [Ops03] Automatic SW Download to eNodeB
- [Ops04] Automated NEM upgrade
- [Ops05] Cell outage detection
- [Ops06] Performance Management in real time
- [Ops07] Direct KPI reporting in real time
- [Ops08] Information Correlation for Fault Management
- [Ops09] Subscriber and Equipment trace
- [Ops10] Cell Outage Compensation
- [Ops11] Compensation for Outage of higher level network elements
- [Ops12] Fast recovery on instable NEM system
- [Ops13] Mitigation of outage of units

Since most of the use cases related requirements can be derived due to the self-explaining notations, a detailed explanation is omitted here but can be found in [7].

Analyzing these different use cases and clusters, it is found that the Planning cluster candidates for the highest *relative* OPEX saving potential which denotes to 50% - 60% of the current spending[1]. However, it is to say that this includes a 100% realization of all use cases while [P01] and [P02] currently are not seen as realistic ones. Anyway, further analyses lead to the conclusion that the Optimization cluster should be paid most attention to since realization of its uses cases promises to result in the highest *total* savings[1].

In general, Planning of transport parameters [P04] as well as Radio Parameter Optimizations [O01]-[O05] turned out to be the most promising single use cases.

Considering the overall OPEX reduction potential of all clusters and use cases, it was found that efficiency gains add up to 40% of current OPEX spending for network operations (ceteris paribus). Even for pessimistic assumptions of SON applicability, total savings up to eight-digit amounts[1] are possible in Germany. Worldwide, ideal SON realization potentially covers even a nine-digit amount.

While the above figures are quite impressive, it is to say that the actual amount of savings is expected to be even higher. The 32 use cases generally entail high saving potentials. However, analyses were based on a ceteris

[1] Total numbers are known to the author, but cannot be disclosed in the scope of this contribution.

paribus assumption, hence only *improvements to existing installations* could be analysed to derive saving potential figures. For use cases such as Home Node B deployment [O05/O07], no savings could be determined since there is no comparable solution in place by now. However, though the saving potential cannot be quantified here, it is self-evident that a Self-X based plug and play solution for Home Node Bs obviously is a desirable and highly efficient option.

4. SELF-X SOLUTIONS & SERVICE CHAIN

Section 3 has pointed out one main motivation for Self-X: Operators long for OPEX reductions thanks to SON features. Obviously, this goes hand in hand with further (technical) goals such as complexity handling.

Accordingly, operators push Self-X features to be developed and expect first solutions to be in place for commercial availability of their next generation mobile network in 2010. This directly leads to the question, which party is considered to be responsible to ensure availability of Self-X solutions. From the operator's point of view, this is clearly a task for the vendors.

Having a look at Figure 5, the mobile user is on top of the service chain. He is the end-customer for which dedicated services are designed according to his desire. In most of the cases, one can assume that the end-user is technique agnostic, which means that he is not interested in technical details and problems to be overcome in order to provide the services. Having a contract with an operator, a subscription for dedicated services is launched which ideally covers an "all inclusive package". Since the money flow is from the end-customer to the operator, the operator needs to fulfil all service related tasks for the user's benefit. The reciprocal direction of the money flow therefore is called "service chain".

With respect to the vendor, the operator holds quite the opposite role compared to the previous case. Since operators buy their network equipment, hardware, software and parts of consulting support from manufacturers, the operator now is in the position of being a customer. The money flow carries forward from the operator to the vendor while the service chain follows the opposite direction. The difference here is that the operator obviously is not technique agnostic; in fact quite the opposite attitude applies. Having well defined network roadmaps in mind, operators impose technical requirements to the vendors. Often, frameworks such as joint research or standardization are used as a platform for discussion and mutual information exchange.

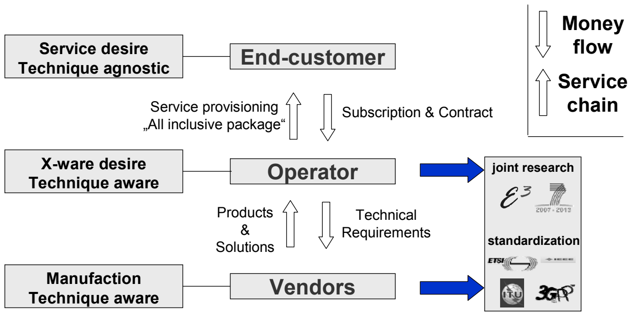

Figure 5. Service chain & money flow for mobile services provisioning.

However, if the end-customer has got the right to expect dedicated services from the operator due to its subscription status and the money payment, the same expectations hold true for the operator towards the vendor. For the Self-X case this means, operators state their needs and expect vendors to come up with adequate solutions. The use cases given in Section 3 are nothing else than scenarios covering the technical requirements operators expect vendors to cope with.

5. RESEARCH & STANDARDIZATION ACTIVITIES

While legacy systems have been designed without Self-X in mind, it is unlikely that much effort and research will be spent on upgrades here. For 4G systems, however, that are just under specification, this mind set has taken place and there is no alternative way forward.

The European Integrated Project (IP) project E3 (End-to-End Efficiency) [8] funded within the 7th Framework Programme of the European Commission [9] has taken up the challenge of specifying autonomic and Self-X related concepts to be incorporated in a System Architecture (SA) framework. Another FP7 project, SOCRATES aims at the development, evaluation and demonstration of methods and algorithms for self-configuration, self-optimization and self-healing, as a promising opportunity to automate radio network planning and optimization. Key gains are a substantial OPEX reduction and an enhancement of network efficiency and QoS [10].

The omnipresent appearance of Self-X and SON is also subject to standardization.

3GPP Release 8 has recently taken up the challenge of SON specification in its TS32.XXX series. Work is carried out by the TSG SA WG5 "Telecom Management" that specifies the management framework and requirements for management. SON Concepts and requirements [11], Self-establishment of eNodeBs [12][13], Automatic Neighbour Relation (ANR) management [14] as well as concepts and requirements for self-optimization and self-healing [15] are currently being specified. In addition, a technical report on "Self-configuring and Self-optimizing Network (SON) Use Cases and Solutions" [16] for E-UTRAN is in preparation by the TSG RAN WG3 that is responsible for the overall architecture development and the specification of protocols.

Within IEEE, the topic of Self-X was firstly introduced to 802.16 during the Working Group meeting #46 in November 2006. Deutsche Telekom made a contribution on "Provision of self-x functionalities in IEEE 802.16 networks" [17] that gained substantial attention and could lead to the launch of a new IEEE 802.16 study group on Self-x mechanisms in the mid-term.

ETSI has recently (March 2008) established a new technical committee on Reconfigurable Radio Systems (RRS) with the scope of improved spectral utilisation and inter-operator coexistence, using flexible usage modes and a variety of technologies. To achieve this goal, Self-X will undoubtedly play a key role in related specifications

There are many other fora dealing with Software Radio and Cognitive Radio, for which Self-X is seen as a vital enabler, cf. Section 2. For a comprehensive overview on these activities please refer to [18].

6. SUMMARY

The development of mobile radio within recent years results in deployments that at least for some period of time necessitate operation and maintenance of several systems in parallel: 2G, 3G and B3G. From an operator perspective, this additional overhead cannot be coped with traditional means, e.g. human workforce is limited and due to the high cost pressure it is not possible to duplicate all internal structures for each newly supported system. As a response to these challenges, Self-X is seen as a highly potential candidate to ensure more automatic operation of the systems, resulting in more flexibility, faster reaction time and thus reduced costs while at the same time increasing the user's service experience. Further on, Self-X is seen as a substantial element of cognitive radio, hence enabling sustainable network development.

While the need for Self-X can easily be retraced, its realization path is probably more critical. According to the money flow and service chain principle presented in this paper, operators consider vendors to be in charge of coming up with sophisticated solutions to this problem. However, the first step was with the operators by defining a detailed framework within which they want Self-X to be deployed. This has happened in terms of use cases representing the scenarios for future SON and thus Self-X application.

References

[1] J. Mitola et al., "Cognitive radio: making software radios more personal", IEEE Personal Communications, Volume 6, Issue 4, Aug 1999 Page(s):13 – 18

[2] Zhiyong Feng, "Joint Radio Resource Management and Cognitive Pilot Channel (CPC) in Cognitive Wireless Networks", European-Chinese Cognitive Radio Systems Workshop, 26-27 May 2008, Beijing, China

[3] E3 Project Glossary, https://ict-e3.eu/glossary/C/c.html

[4] Dr. K.-J. Krath (Senior Vice President Radio Networks Engineering & Quality), T-Mobile International AG, Keynote Plenary, LTE World Summit, May 2008

[5] NGMN Alliance, http://www.ngmn.org/

[6] NGMN Alliance, "Use Cases related to Self Organising Network. Overall Description", Version 2.02, 16th April 2007

[7] NGMN Alliance, "Informative List of SON Use Cases", Version 1.53, 17th April 2007

[8] FP7 project E3 (End-to-End-Efficiency), https://www.ict-e3.eu/

[9] European Seventh Framework Programme (FP7), http://cordis.europa.eu/fp7/home_en.html

[10] FP7 project SOCRATES, http://www.fp7-socrates.org/

[11] 3GPP TS 32.500, "Telecommunication management; Self-Organizing Networks (SON); Concepts and requirements"

[12] 3GPP, TS 32.501 "Telecommunication management; Self-Organizing Networks (SON); Self-establishment of eNodeBs; Concepts and requirements"

[13] 3GPP, TS 32.502 "Telecommunication management; Self-Organizing Networks (SON); Self-establishment of eNodeBs; Stage 2"

[14] 3GPP, TS 32.511 "Telecommunication management; Self-Organizing Networks (SON); Automatic Neighbour Relation (ANR) management; Concepts and requirements"

[15] 3GPP, TS 32.521 "Telecommunication management; Self-Organizing Networks (SON); Self-optimization and self-healing; Concepts and requirements"

[16] 3GPP TR 36.902 V0.0.1 (2008-02), "Evolved Universal Terrestrial Radio Access Network (E-UTRAN); Self-configuring and self-optimizing network (SON) use cases and solutions"

[17] E. Bogenfeld at al., "Provision of self-x functionalities in IEEE 802.16 networks - e.g. self-configuration and self-optimization". Contribution C80216-06_027 in IEEE 802.16 Broadband Wireless Access Working Group, Nov. 2006,
http://wirelessman.org/docs/06/C80216-06_027.pdf
[18] L. Berlemann and S. Mangold, "Cognitive Radio in IEEE Wireless Standardization" in ABMT 1000: ComNets Sonderausgabe anlässlich der Eröffnung des ComNets-Gebäudes, Nov 2008
[19] ETSI Technical Committee RRS,
http://portal.etsi.org/portal_common/home.asp?tbkey1=RRS

THE GPRS ERA AT COMNETS - WORLD LEADERS IN STANDARDISATION, IMPLEMENTATION AND DEPLOYMENT

Peter Stuckmann, Götz Brasche, and Peter Decker

1. INTRODUCTION

In the early 90s, after 10 years of specification work, the first cellular digital mobile radio systems based on the global system for mobile communications (GSM) started operation. Although GSM had been designed as mobile extension of the integrated services digital network (ISDN), the structure of the air interface did not allow support of the complete range of ISDN data services. Already in the late phases of the specification work, Prof. Walke who had been involved in the development and research of GSM, envisioned that the steadily growing number of users and their increasing requirements as well as the tough competition within the rapidly expanding mobile communications market would soon require extended speech and new data services that accommodate both an efficient resource management and flexible quality of service. He decided to establish a research group for the extension of GSM towards new multi-media services.

This paper is an acknowledgement of Prof. Bernhard Walke's visionary mind and his significant reputation and influence in the development of cellular mobile radio networks. It gives a high-level survey of the "GPRS era" at the Department of Communication Networks of RWTH Aachen University (ComNets) – the time period in which Prof. Walke and over time a few tens of ComNets researchers started a successful journey to investigate methods to offer sophisticated services in future releases of the initial GSM-system. This era began in 1991 when Prof. Walke, Peter Decker et al. presented CELLPAC, a packet radio protocol proposed for the GSM mobile radio network at the Mobile Radio Conference [1]. His first paper on GPRS [2] was the first major publication on the topic, which led to wide acceptance of his expertise in the field. This culminated in 2002 with the publication of a text book about the evolution of GSM [3]. In between, numerous publications have been made and 3 PhD-theses have been published [4], [5], and [6].

Starting in 1991, Dr. Peter Decker laid the foundation of the research of possible enhancements of the (GSM) by next generation packet oriented data services and protocols at ComNets. He investigated new concepts and methods to increase the data rate within a single GSM traffic channel and proposed a new medium access control (MAC) and a new radio link control layer (RLC) protocol that enable effective multiplexing of packet switched data sources. Dr. Decker actively participated in the GSM phase 2+ standardisation and was member of the respective standardisation group of the European Telecommunications Standards Institute (ETSI) that was leading the standardization efforts. The results of his research along with the performance evaluation of the proposed concepts significantly influenced the early stages of the specification work for the MAC and Radio Link Control protocols of the future GSM service GPRS (General Packet Radio Service).

In 1995, Dr. Goetz Brasche took over Dr. Decker's research and role as representative of ComNets in the GSM standardization driven by ETSI. With the Variable Rate Reservation Access (VRRA) and Master-Slave Dynamic Rate Access (MSDRA) protocols, he implemented, enhanced and evaluated the performance of the two candidate MAC protocol concepts for the emerging radio interface of the GPRS standard. In particular, Dr. Brasche developed GPRSim, a sophisticated simulation suite to analyze the performance of the candidate protocols with the intention to support the ETSI specification groups in the final specification and selection of the radio interface protocols. For the purpose of simulation, Dr Brasche also applied a new software engineering method. Based upon a combination of formal description methods and object-oriented programming this method allowed the use of integrated Computer-aided Software/System Engineering (CASE) tools encompassing proof of functional correctness and detailed performance evaluation. Thus, his research gave new impulses to advanced software engineering and substantially contributed to the performance analysis of the new packet radio service prior to its introduction.

When Dr. Peter Stuckmann started his research on GSM packet data services three years later in 1998 he took advantage of the comprehensive knowledge on GPRS and significantly evolved the simulation suite and supported the early deployment of the service. After approval of the first edition of the GPRS standards at the end of the 1990s, leading network operators in Europe started to collaborate with Dr. Stuckmann and his research group in feasibility and dimensioning studies necessary for the economic introduction of GPRS and its evolution EGPRS (enhanced GPRS) based on Enhanced Data Rates for GSM Evolution (EDGE). In this framework Dr. Stuckmann developed GPRSim to an emulator comprising the implementation of the GPRS and EDGE protocols and load generators for typical GPRS/EDGE usage. This tool not only allowed to study existing and evolved GSM systems in their natural environments with the appropriate radio coverage, mobility and typical traffic volumes but also to analyze approaches for improvement and introduction of new features. The further development of this simulation tool to a comprehensive capacity planning tool and the achievement of the traffic engineering results were only possible with the support of on average about five Master students and student assistants whom Dr. Stuckmann organized in a research group under his supervision.

2. GSM PACKET RADIO

With the development of the Global System for Mobile communication (GSM) standard for digital cellular mobile radio networks in the late 1980s in Europe and their introduction in the 1990s a new mass market with several million subscribers worldwide has been created. Besides the growth of the subscriber numbers, the technological evolution of GSM is going on. New services and applications have been developed and standardized and are presently integrated into GSM networks. Though its principal use is for mobile telephony, mobile data services are becoming more and more popular. The success of text messaging in Europe and the growth in both subscriber numbers and the usage of the i-mode service in Japan, which enables the delivery of Internet-like content on mobile phones, has led to very high market potential estimations for packet-oriented mobile data services. Licenses for Third-Generation (3G) systems, based on Universal Mobile Telecommunication System (UMTS), which are seen as the successor of GSM-based Second-Generation (2G) systems, were granted in 1999–2002 in Europe to offer the needed radio capacity for data services with higher data rates and for enhanced speech services.

UMTS is aiming to realize peak bit rates of up to 144 kbit/s with wide coverage, up to 384 kbit/s in hotspots, and up to 2Mbit/s in indoor scenarios. A stepwise way in the direction of 3G, however, will already be performed by the extension and the further development of existing cellular systems. The advantage of such an evolution process is the faster availability of such services, since the infrastructure of existing 2G systems can be used. Furthermore, the opportunity is given to prepare the customers for new, so-called 3G services.

First packet-switched services based on the General Packet Radio Service (GPRS) have been available in Europe since 2001 and most of the GSM networks world-wide have introduced GPRS. Due to this service, mobile data applications with peak bit rates of up to 117 kbit/s and typical user data rates of 25–64 kbit/s have been offered and established on the market. To realize higher data rates the European Telecommunications Standards Institute (ETSI) has developed the Enhanced Data rates for GSM Evolution (EDGE) standard. The packet-oriented part Enhanced General Packet Radio Service (EGPRS) offers a peak bit rate of up to 384 kbit/s and typical user data rates of 40–100 kbit/s by means of modified modulation, coding and medium access schemes.

3. GPRS STANDARDISATION

In the early phase of GPRS standardization ComNets focused on design, prototype implementation and performance evaluation of MAC protocols for the GPRS air interface. The master slave dynamic rate access (MSDRA) protocol that was studied within this framework can be considered as a GRPS RLC/MAC prototype implementation since it incorporates the main features of the ETSI standard proposal.

3.1 Channel Concept

When a network operator decides to offer GPRS-based services within a cell one or several physical channels out of the pool of available channels are dedicated to packet mode transfer. One PDCH is mapped onto one physical time slot. According to the requirement for flexible adaptation to different traffic conditions allocation of PDCHs is based on demand. Furthermore, uplinks and downlinks are basically used as independent channel resources, i.e. in one time slot an uplink PDCH may carry data from one MS while data to another MS is transmitted on the downlink PDCH. In order to simplify the logical channel concept, the allocated PDCHs are logically grouped into master MPDCHs and slave channels SPDCHs. The SPDCHs represent the channels on which user data and dedicated signaling is transferred.

MPDCHs accommodate common control channels (CCHs) that carry the signaling information that is required to initiate packet transfer:

- The Packet Random Access Channel (PRACH) is used on uplink exclusively in order to initiate data transfer of the MS
- The Packet Paging Channel (PPCH) is used on downlink exclusively in order to inform MSs about incoming packets
- The Packet Access Grant Channel (PAGCH) is used on downlink only to send channel reservation information to an MS prior to data transfer:
- The Packet Broadcast Control Channel (PBCCH) is used on downlink only to broadcast all GPRS-specific information.

With the 26- and the 51-multiframes there are two multiframes defined in GSM. The 51 multiframe has been chosen for GPRS for synchronization reasons. The first multiframe of a multiframe cycle is divided up into 12 blocks of 4 frames. The first 11 frames are allocated to carry control information, while the remaining 10 blocks are allocated for paging and broadcasting. Within the other seven multiframes 3 frames are dedicated to random access on uplink (PRACH), while all others represent a PDCH.

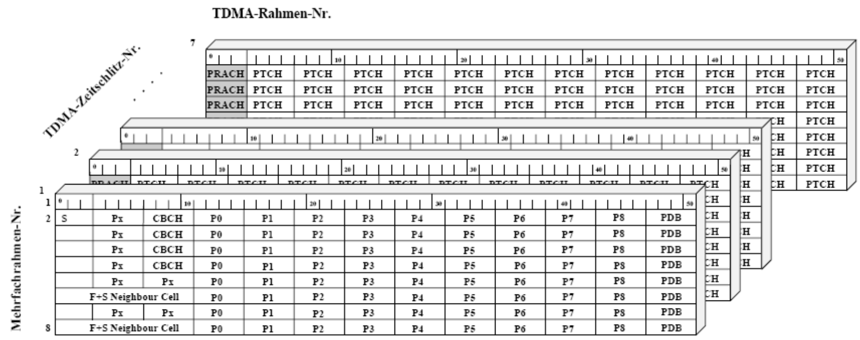

Figure 1. Multiframe Structure.

3.2 Model of Operation

An LLC protocol data unit that is to be transferred over the air interface is segmented into one or more RLC frames that are handed over to the MAC layer. Each MAC frame is transmitted as one block of consecutive TDMA slots. A selective ARQ mechanism controls retransmission of erroneous or missing blocks by use of a temporary frame identity (TFI). The TFI further contains a job identification by means of which multiplexing of multiple jobs onto one PDCH is possible.

Organization of slot assignment to the different MS is done centrally by the BS. The channel reservation includes the time slot number and an uplink status flag (USF) coded by 3 bits. Thus, this USF can be used to multiplex up to 8 different MS onto one slave channel. An MS monitors the USF and according to the USF value it is able to identify PDCHs that are assigned to it and starts transmission. The USF is transmitted at the beginning of each RLC block. On the MPDCH multiplexing of up to three MS is allowed.

Provided that an MS is multi-slot capable blocks of one MAC frame can be sent on different SPDCHs simultaneously. By this multi-slot reservation the packet delay can be reduced and the bandwidth assigned to one MS can be varied dynamically. Thus, the status flags not only result in a highly dynamic reservation but also allow interrupting transmission in favor of pending or high priority messages.

Medium access is based on a slotted ALOHA reservation protocol, i.e. on the uplink there are three phases:

- Contention phase: a slotted ALOHA random access technique is used to transmit reservation requests.
- Notification: the BTS transmits a notification to the MS indicating the channel allocation for a pending uplink transmission
- Transmission: the data transfer occurs without contention.

On the downlink, there are two phases:

- Notification: the BSS transmits a notification to the MS indicating the channel allocation for a pending downlink transmission
- Transfer: the MS monitors the indicated channels and the transfer proceeds without contention.

3.3 Mobile Originated Transfer

Packet transfer is initiated by a random access request on the PRACH that is determined by uplink status flags (USF) sent on the corresponding downlink MPDCH. Together with the access request the MS indicates the number of GPRS slots required.

In case of correct receipt of the access request, a channel reservation command including timing advance and TFI is sent by the BTS in which the reserved slots are marked. Since the capacity of the PAGCH is limited, not all correctly received access requests can be served directly. Nevertheless, to avoid superfluous re-sending, the affected MS are informed about receipt and later channel reservation by use of an access grant notification message that may be concatenated with the channel reservation message to another MS. Furthermore, since the blocks are sent according to descending order the BTS always knows how many blocks are still to be received and may adjust reservation scheduling.

If no response to an access request is received by an MS, a re-transmission procedure is started after a random back-off time up to a maximum number of access attempts. After transmission in the reserved time slots is completed, an acknowledgment is sent by the BTS. In the case of erroneous or missing blocks, a negative ACK (NACK) is sent and only those blocks listed as erroneous are re-transmitted. For that reason, this NACK directly includes an appropriate channel reservation. This implies that a NACK can be retransmitted with a minimal delay only since the BTS directly recognizes missing MS data re-transmission on the first pre-reserved time slots. If the MS does not receive an ACK within a certain time frame transfer recovery is started by a new random access. Within the random access the reservation of one single slot is indicated. After access grant, the first block of the frame is sent. Thus, the BTS knows that the last ACK belonging to this frame transfer was not correctly received and should be retransmitted. In order to enable frame recovery, the frame and associated information about received and acknowledged blocks should be kept up to a minimum value of several seconds.

3.4 Mobile Terminated Transfer

A BTS initiates a packet transfer by sending a page on the PPCH. If the BTS knows the location of the MS, this page may include a direct reservation of uplink slots. Thus, an MS responds by either sending a random access request on the PRACH or immediately starts data transmission on pre-reserved slots. In the case that the location is only known to a certain degree of probability, the BTS does not reserve slots for immediate data transmission but avoids the collision sensitive random access procedure and reserves a single slot for a page response that precedes the channel reservation. If a page without reservation is made, the MS initiates the random access and indicates a reservation of one block to be able to identify itself after access grant is received. As far as multiplexing multi-slot downlink transmission and error handling are concerned, the BTS has the same functional possibilities as an MS. Of course, whether more than one downlink PDCH that has been assigned to GPRS can be used for parallel transmission depends on the MS's capability to monitor these PDCH simultaneously.

3.5 Burst formats and Coding

Four different coding schemes are defined to be able to adaptively react to the current channel quality. The first coding scheme equals the SDCCH coding used in GSM 1/2 rate convolutional coding and a 40 bit Fire code is applied. This scheme is used for all signaling messages. The second and third schemes are punctured versions of the first one with rates of 2/3 and 3/4 respectively. The fourth coding scheme does not apply a convolutional coder. The latter three schemes use a 16 bit frame check sequence for error detection.

In order to speed up decoding of USF, a 12 bit block USF code word can be generated. This is achieved by pre-coding the USF into a 6 bit block word before applying convolutional coding to the whole block without puncturing the first 12 bits. The coding scheme is indicated by the GSM stealing flags of the four consecutive bursts that belong to one block using an 8 bit block code with hamming distance of 5.

For random access and paging the existing GSM random access burst is used while data transmission is done with GSM normal bursts.

4. FROM STANDARDISATION TO DEPLOYMENT AND ENGINEERING

4.1 The Dimensioning Problem

Whereas just after the service introduction minimal cell configurations were chosen supporting only a basic availability of GPRS, with increasing data traffic load during the next years GSM/GPRS cell capacity had to be extended. For these evolution scenarios traffic engineering guidelines are required. They should describe the relationship between the offered traffic and the radio resources that have to be allocated to reach a desired quality of service for the different applications. For traffic engineering in circuit-switched networks the Erlang-B-Formula has been successfully applied over decades, while for packet-switched cellular radio networks such an applicable traffic engineering model was missing. The analytical description of statistical multiplexing and Internet and Multimedia traffic modeling are more complex than for circuit-switched networks. Although results for packet multiplexer systems are available, e.g., for Asynchronous Transfer

Mode (ATM) networks, the specifics of the radio interface had not been included into the models.

Traffic engineering procedures ensure that the network is designed and upgraded in a cost-effective way. They should be based on the traffic-performance relation, linking network capacity, traffic demand and realized performance, and should assure that the network has sufficient capacity to handle the offered traffic. Traffic engineering is fundamental to the design of circuit-switched networks like the telephone network. The traffic-performance relation here is given by the Erlang loss formula which gives the probability of call blocking when a certain volume of traffic is offered to a given number of circuits.

For packet-switched networks, the objective of theoretical or simulation studies is to define simple network engineering procedures like applying the Erlang formula in circuit-switched networks. For mobile packet data services like GPRS the amount of radio resources available for packet data traffic is most critical for the realized performance. The resources can be defined by a number of fixed or on-demand Packet Data Channels (PDCHs) to be allocated for GPRS in the case of sufficient capacity for both speech and packet data traffic or by the number of additional transceivers. These have to be installed in the existing GSM base stations, if the traffic demand for speech and data is exceeding the acceptable traffic that can be carried by the existent hardware. To achieve the goal of defining simple network engineering procedures that are usable in practice, dimensioning graphs for application-specific performance measures are proposed that are valid for the cell and load scenarios of interest. With these graphs, describing the traffic-performance relation, the offered traffic that can be served under a given number of radio resources (see Figure 2.2(a)) or the number of resources, necessary for a given offered traffic (see Figure 2.2(b)), can be estimated.

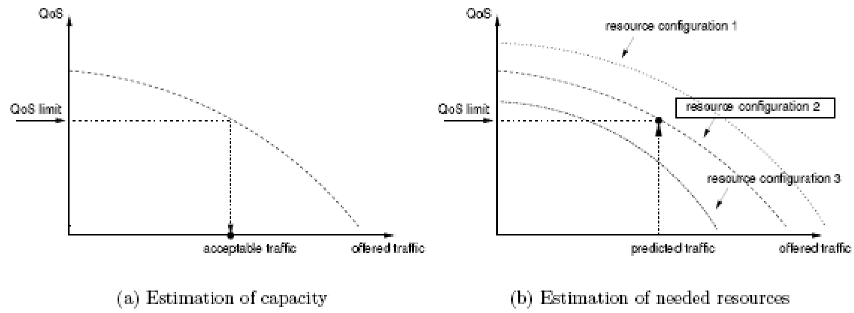

Figure 2. Dimensioning graphs.

For interactive download-oriented applications like World Wide Web (WWW) and e-mail the throughput performance is of interest. For transaction-oriented applications with small objects like WAP the application response time is perceived by the user. For Streaming applications, however, the delay and delay variation is critical. Since in GPRS networks no strict QoS guarantees are supported and no delay-critical applications is introduced, traffic engineering rules are based on mean values first of all. For the network evolution like the introduction of streaming applications, the introduction of radio interface enhancements or for traffic engineering of related packet radio networks, the same methodology has been applied in using stricter QoS measures like throughput or delay quantiles. To ensure the applicability in practice, the traffic engineering rules themselves should be simple and only based on the user number and traffic volume during the busy hour. However, the dimensioning graphs themselves should be taken from accurate models for the protocol stacks, the traffic pattern and the radio channel as close as possible to reality.

4.2 The Tool GPRSim – the Second Generation

The GPRS Simulator GPRSim is a pure software solution based on the programming language C++. For implementation of the simulation model the Communication Networks Class Library (CNCL) is used that is a predecessor to the SDL Performance Evaluation Tool Class Library (SPEETCL). This allows an object-oriented structure of programs and is especially applicable for event-driven simulations. The complex protocols like LLC, RLC/MAC, the Internet traffic load generators and TCP/IP are specified formally with the Specication and Description Language (SDL) and are translated to C++ by means of the Code Generator SDL2CNCL and are finally integrated into the simulator.

Unlike the usual approaches to building a simulator, where abstractions of functions and protocols are being implemented, the approach of the GPRSim is based on the detailed implementation of the standardized protocols. This enables a realistic study of the behavior of EGPRS and GPRS. In fact, the real protocol stacks of (E)GPRS are used during system simulation and statistically analyzed under a well-defined traffic load. The event control is performed by event handlers that are activated by arriving events and that send events to other event handlers after processing. The scheduling is done by a scheduler, which determines the order of processing of the events. Each event has a priority and a defined processing time. The simulation time advances in discrete steps, when all events, the processing time of which coincides with the current simulation time, are processed. Through this contemporaneity in the simulation process the simultaneous reaction of, e.g., several mobile stations to an event is represented realistically.

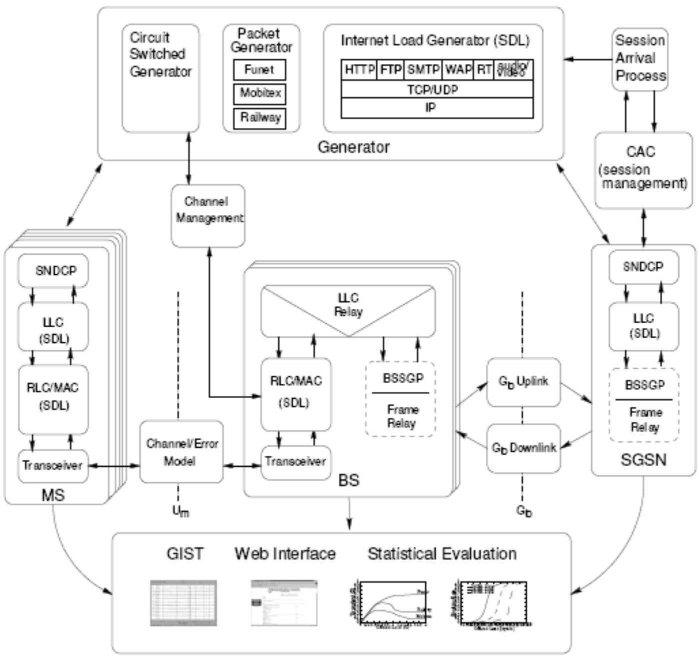

Figure 6.1: Structure of the (E)GPRS simulator

Figure 3. GPRS Simulator GPRSim.

The logical structure of the GPRSim and the information flow between the modules is shown in Figure 6.1. The simulator comprises the modules MS, BS and SGSN, the transmission links, the load generator, session control modules, a graphical user interface for presentation and a module for statistical evaluation. Multiple instances of MS and BS can be generated and studied in a multicellular environment. The formal structure of MS, BS and SGSN is similar. They contain the implementations of the respective protocol stacks. The layers BSSGP and Frame Relay (FR) are not represented in the GPRSim because the focus was set on the radio interface. The respective classes do not provide any functionality and simply forward the service data units to the peer entity. The transmission links are represented by error models. While the Gb interface is regarded as ideal, block errors on the radio interface Um can be simulated based on lookup tables, which map the actual Carrier-to-Interference (C/I) to a Block Error Probability (BLEP) considering mobility.

The load generator comprises generators for both circuit- and packet-switched traffic and includes transport and network protocols implementations of the TCP/IP protocol stack. The module Channel Management supervises the physical GSM channels available in the respective cell and allocates channels for the GPRS resource management entity, if the channels are not used by circuit-switched services or if they are not allocated as dedicated PDCHs for (E)GPRS.

The output of the simulator comprises a graphical presentation of the protocol cycles and the statistical evaluation results of the performance measures. In the following sections the different modules are presented with their functionality and interactions.

4.3 Outcome of this Work

The goal of this work was to develop dimensioning concepts for cellular packet radio networks that will remain valid during network evolution. Two important requirements for these concepts are practical applicability and accuracy. While the traffic engineering rules themselves should be simple and only consider the user number and traffic volume during the busy hour, they should be based on accurate models for the protocol stacks, the traffic pattern and the radio channel that ought to be close to reality. The key tasks to achieve this aim are the identification and development of adequate traffic models for existing and future mobile applications, the prototypical implementation of the GPRS/EDGE protocol stack as well as the integration of optimized methods for QoS support, so that the presented concepts will be inline with the evolution of the radio interface protocols.

For existing and future mobile applications, which have been predicted, traffic models are proposed. For WWW and e-mail applications, which are already popular today in fixed networks, adequate traffic models from literature have been identified and adapted for the applicability in mobile environments. Traffic models for applications that were emerging for mobile networks like WAP and MMS have been derived from measurement. Finally traffic models for Streaming applications have been identified in following the recommendations of standardization bodies that are defining the standards for mobile Audio and Video applications. From these models an integrated Internet and Multimedia load generator was implemented, which enables accurate modeling of all relevant traffic types in configurable traffic mixes.

To be able to study the existing and evolved systems in their natural environments with the appropriate radio coverage, mobility and typical traffic volumes and to analyze own approaches in the improvements and introduction of new features, the emulation tool GPRSim has been developed. The complete GPRS/EDGE protocol stack based on the actual standard has been formally specified. Additionally the load generators and the underlying TCP/IP protocol stacks have been integrated and finally adequate channel and mobility models have been implemented and integrated. To ensure that the results of the GPRSim can be regarded as representative a validation by analysis and measurement for simplified scenarios has been carried out.

In addition to this simulative approach the GPRS system has also been evaluated analytically. With the analytical results a fast estimation of the general system capabilities can be performed. On the other hand the examinations show the limitations of state-of-the-art analytical approaches, when complex scenarios with heavy-tailed traffic sources, the dynamic behavior of TCP and radio interface-specific mechanisms like scheduling and link adaptation have to be considered. Considering these limitations in analytical modeling the work was concentrated on a comprehensive simulative performance evaluation of GPRS and EGPRS for relevant scenarios, protocol options and predicted applications.

Summarizing the contribution of this work, traffic engineering guidelines have been developed that are accurate and usable in practice. The most important effects that occur for different traffic characteristics are covered in the performance analysis. While the simulation results can be used as a basis for accurate GPRS/EDGE capacity planning, the general concepts and effects that were found are valid for all cellular packet radio networks. The concepts have been developed in collaboration with several network operators and have been integrated in the planning process by many leading

GSM operators world-wide. Further, a comprehensive capacity planning tool has been established that has already been applied by several GSM operators and that presently is has been successfully further developed to a commercial planning tool. With this tool additional scenarios can be examined in detail und the results can be further integrated in the planning process.

5. EVOLUTION TO EDGE

Enhanced Data rates for GSM Evolution (EDGE) is a further development of the GSM data services High-Speed Circuit-Switched Data (HSCSD) and GPRS and is suitable for circuit- and packet-switched services. The circuit-oriented part is the Enhanced Circuit- Switched Data (ECSD). The packet-oriented part is the Enhanced General Packet Radio Service (EGPRS). Applying modified modulation and coding schemes EDGE reaches very high raw bit rates of up to 69 kbit/s per GSM physical channel. If a user utilizes all 8 time slots in parallel, the theoretical maximum raw bit rate rises to 554 kbit/s. The maximum bearer bit rate achievable rises to about 384 kbit/s. EDGE was introduced to the ETSI for the first time in 1997 for the evolution of GSM. After a successful feasibility study of the ETSI the standardization process for EDGE was initiated. Although EDGE was introduced for the evolution of GSM, this concept can be applied to increase the data rate in other systems. Since the network architecture of the GSM will remain similar for EDGE, the modifications at the air interface are depicted in this chapter. To support higher data rates, a modulation scheme called 8-Phase-Shift-Keying (8-PSK) is introduced which does not replace the Gaussian Minimum Shift Keying (GMSK) but coexist with it. With 8-PSK it is possible to provide a higher data rate, which is necessary to support bandwidth extensive data applications. The modifications mostly concern the RLC/MAC layer and the physical layer. Since these protocols are implemented in the MS and the Base Station (BS), both have to be modified. In reality, the changes that have to be made comprise a new EDGE transceiver unit and software upgrades to the Base Station Controller (BSC), which then can handle standard GSM or GPRS traffic and will automatically switch to EDGE mode when needed. The core of EDGE is the Link Quality Control (LQC) mechanism that allows the adaptation of the Modulation and Coding Schemes (MCSs) to a changing radio link quality. Although Link Adaptation (LA) is already possible within the GPRS standard, higher CSs are not supported by the actual equipment and will probably only be introduced together with EDGE functionality.

Additionally, a type-II-hybrid ARQ (soft ARQ) scheme is introduced. Soft information is stored during retransmissions to enable Incremental Redundancy (IR). The RLC/MAC protocol structure and retransmission mechanism proposed for EGPRS are based on the GPRS standard.

A throughput performance gain of up to 100% compared to GPRS is reachable with the new Modulation and Coding Schemes (MCSs) in EGPRS. A maximum throughput performance of about 47 kbit/s is reachable for the traffic mix considered. While this gain in user throughput performance is not dramatic, the advantage of EDGE is mainly the higher capacity per PDCH compared to GPRS, since more than 40 active MSs can easily be served with 8 PDCHs maintaining good performance. Additionally Link Adaptation (LA) and Incremental Redundancy (IR) are enabling performance gains in comparison to a fixed chosen MCS. Although the gain through LA, in comparison with a well chosen fixed MCS, is not dramatic, LA should be used in EGPRS networks. The reason is that without LA, a fixed MCS would have to be chosen for every site. This cannot simply be realized in realistic radio coverage planning processes. Further insights were gained by the examination of the most important mobile applications over GPRS and EGPRS. It has been shown that WAP traffic can be multiplexed seamlessly with Internet traffic because of the small and limited WAP deck size, while Internet traffic slightly slows down WAP traffic in situations with high traffic load. Regarding Streaming applications over EGPRS it has been found that the throughput performance remains acceptable for up to 20 stations offering Streaming traffic. In traffic mix scenarios with Streaming traffic and TCP-based applications, EGPRS can easily serve 40 stations. In GPRS scenarios, Streaming applications can only be realized in situations with low traffic load.

6. THE STORY CONTINUES – PACKET RADIO IN 3G AND BEYOND

6.1 From GPRS to 3G

GPRS and EDGE can be seen as a first step towards mobile multimedia services. With penetration of mobile phone services reaching saturation in developed countries, the next big growth opportunity is seen in 3G mobile data services. The first of these 3G networks, are now starting commercial services in Europe. In 2004 there were 75 licensed 3G operators, of which 31 were offering commercial services and another 21 were in the precommercial phase. By the beginning of 2005, it is estimated that there were around 5.4 million Universal Mobile Telecommunication System (UMTS) subscribers in the EU out of 16 million worldwide. This figure is growing rapidly as new operators roll out their 3G networks and worldwide subscriptions to 3G networks. After the UMTS Forum, worldwide subscriptions to 3G networks have exceeded 100 million in June 2006.

Despite the promises of more feature-rich, highly interactive and high bit-rate multimedia services for the end-users and increased revenues for the operators, the research community has perceived the limitations of these systems in terms of user throughput, which is typically not exceeding 64-384 kbit/s, and cost of operation and usage. Consequently, evolved 3G standards have been developed. Possibly the most important improvement is the new series of technologies referred to as High Speed Packet Access (HSPA). These technologies are available as a relatively straightforward upgrade to existing UMTS networks and are offering improved user data rates, typically 1-2 MBit/s, improved network capacity, and improved interactivity for data applications. As the next step, 3G Long Term Evolution (LTE) is currently standardised. Based on the different radio interface transmission technology Orthogonal Frequency-Division Multiplexing (OFDM), 3G LTE is aiming to achieve a downlink data rate of three to five times higher compared to HSPA in the same bandwidth and a significantly lower latency. For most situations, the cost to evolve a UMTS network to a next generation radio interface will be low compared to the cost of deploying a new network. The reason is that most of the existing infrastructure will remain the same, requiring only major upgrades at a base station and on terminals and inter-working with existing systems is easily achievable as the same core network can be used.

6.2 4th Generation Mobile Communication Systems

4G is a concept that is currently subject of intensive research efforts throughout the world. These envisioned advanced mobile communication systems are expected to offer broadband mobile applications with access to high-quality multimedia content and offering machine-to-machine communications and communication with objects and devices [7].

The technological approach driving 4G efforts is however somewhat different from the one that has driven other mobile and wireless technologies such as GSM/2G, UMTS/3G, or even wireless access technologies such as WLAN and WMAN (see previous section). These technologies have been developed with a vertical approach, with a target subset of services and environments supported by one particular radio access scheme complemented with a supporting network infrastructure in the case of 2/3G. 4G is on the other hand researched with a comprehensive system approach that includes a continuum of different access technologies, federated through a core network that ensures the following requirements.

- True broadband, i.e. no user-perceived difference from fixed and mobile broadband access
- Enhanced resource efficiency (in particular spectrum-efficiency) and versatile/reconfigurable technologies, minimising Capital Expenditure (CAPEX) and Operational Expenditure (OPEX).
- Increased service capabilities, in principle enabling design and implementation of context-aware applications
- Full fixed-mobile convergence, i.e. equivalent service capability across a fixed or mobile access
- Service portability and operations across multiple networks/service provider domains

For new radio access schemes that are aimed to be integrated with other existing radio access networks, ITU has set the target of 100 Mbit/s for truly mobile applications, and 1 GBit/s for fixed/portable radio access. These objectives are framing the research and the characteristics of the test beds that are currently being developed in various regions of the world. Although a number of concepts and demonstrators towards this target have been developed in several research projects world-wide, 4G standardisation has not started, yet. This is mainly due to the uncertainty of the identification of frequency bands for such a new radio schemes, which will be addressed in the framework of the upcoming World Radio Conference (WRC) 2007.

6.3 Next Generation Mobile Networks

Recently, a group of operators published a white paper on Next Generation Mobile Networks (NGMN). Most of the concepts and requirements outlined in this paper are similar to the requirements taken into account by current 4G research activities including the need to develop a new radio access scheme, called the NGMN Access, and to satisfy a large range of interoperability requirements. Key concerns are relating to the need to simplify the architecture, which is already considered as too complex and whose complexity will increase as interoperability requirements increase. New standards should be defined with well-defined transparency of Intellectual Property Rights (IPR) from the onset. Emphasis is also put on the need for upwards compatibility, i.e. for standards enabling an evolutionary migration of 2G/3G towards the target NGMN system whilst ensuring transition towards end-to-end support of Internet Protocol (IP) - based applications.

ACKNOWLEDGEMENT

Prof. Walke was one of the first in mobile communications research recognising that the steadily growing number of users and their increasing requirements within the rapidly expanding mobile communications market would soon require extended speech and new data services based on packet radio. His first paper on GPRS [2] was the first major publication on the topic, which led to wide acceptance of his expertise in the field. He was able to generate critical mass around the topic GSM packet radio very early by creating enthusiasm among researchers and students at ComNets. After this great start he continuously followed-up this topic in research, standardisation and deployment. In total he acquired around 10 projects on this topic with industrial partners and numerous government-funded research projects. In addition he empowered his researchers to take the lead when it came to standards contributions, publication of articles and even text books as well as training courses at major conferences and in industry. In his generosity and confidence he let his research assistants stand in the front rank and get a lot of recognition in the research community and in the industrial environment. It is also because of his commitment and continuous fatherlike support that the authors of this paper were able to lay the foundation for their successful careers.

DISCLAIMER

The content of this article has been extracted from previous publications of the authors. The comments and statements made in this article are from the authors and do not necessarily reflect the official position of their employers.

References

[1] Walke, B. and Mende, W. and Decker, P. and Crumbach, J.:Performance of CELLPAC, a packet radio protocol proposed for the GSM mobile radio network. In Proceedings of Mobile Radio Conference 1991, p.p. 57-63, Nizza, France, 11/1991

[2] Brasche, G. and Walke, B. Concepts, Services and Protocols of the New GSM Phase 2+ General Packet Radio Service. IEEE Communications Magazine, Volume 35, Issue 8, Aug. 1997 Page(s):94 - 104

[3] Stuckmann, P.: The GSM Evolution - Mobile Packet Data Services. Wiley & Sons, September 2002, ISBN: 978-0-470-84855-5

[4] Decker, P.:Entwurf und Leistungsbewertung hybrider Fehlersicherungsprotokolle für paketierte Sprach- und Datendienste im GSM-Mobilfunksystem. Ph. D. Dissertation, , p. 208, 1997, ISBN: 3-86073-387-8

[5] Brasche, G.:Prototypische Bewertung und Implementierung von neuen Paket-Datendiensten für das GSM-Mobilfunksystem. Ph. D. Dissertation, Aachen University, p. 272, 1999, ISBN: 3-86073-628-0

[6] Stuckmann, P.: Traffic Engineering Concepts for Cellular Packet Radio Networks with Quality of Service Support. Ph. D. Dissertation, Aachen University, p. 270, 06/2003, ISBN: 3-86073-984-0

[7] Stuckmann, P. and Zimmermann, R.: Toward Ubiquitous and Unlimited-Capacity Communication Networks: European Research in Framework Programme 7, IEEE Communications Magazine, Volume 45, Issue 5, May 2007 Page(s):148 – 157

DESIGN AND PERFORMANCE EVALUATION OF A WIRELESS AD-HOC EMERGENCY RESPONSE MANAGEMENT SYSTEM

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Abstract In this paper, the architecture and middleware services of the wireless-enabled emergency response management system developed on the research project MobileEmerGIS are introduced. MobileEmerGIS aims to gather and to distribute multimedia information to and from mobile nodes (either human users or sensors/actors). Thereby, field forces and eye witnesses can transfer data collected in the field, such as pictures from emergency sites, immediately to centrally located decision makers as well as other field forces (using PDA platforms and even their own private mobile phones). By referencing real-life scenarios (such as a large-scale fire emergency), concrete application services enabled by MobileEmerGIS are presented. One particular challenge for the information sharing is that in emergency situations, multi-media capable public networks are not sufficiently reliable. Hence field forces, who want to share multimedia mission information, have to build up their own multi-hop wireless network. To build up these ad hoc networks, battery driven wireless nodes with meshing capability, called MobileEmerGIS Dropped Units (MDUs), are introduced and have been evaluated within the project. While a demonstrator proved the feasibility of the concept, a dedicated simulation environment allows realistic prediction of the required number of MDUs to achieve a sufficient coverage.

1. INTRODUCTION

In order to aggregate and make available all data necessary to manage large-scale emergency incidents of any kind in an efficient way, a central software platform (called deNIS – German Emergency Prevention &

Response Information System) has been established by the German Ministry of Interior Affairs. The deNIS system contains static data (such as floor plans of critical infrastructure) as well as a constantly growing amount of dynamic data, such as online data collected by sensors to measure for example toxic gas intensity or water levels. One key challenge is to make the vast amount of data stored in the deNIS system available to mobile field forces using wireless technology. MobileEmerGIS is a pilot project within Germany and is based on the collaboration of the developers of the central emergency management software platform deNIS IIplus, the fire brigade of one of the largest cities in Germany and a research partner to provide the innovative, IP-based wireless-enabled communication and data filtering/fusion technology. Based on an in-depth-user requirements analysis, relevant features of a wireless-enabled technology platform have been derived to support field forces of different organizations. In the following we will present selected capabilities of the system, which are envisaged to complement existing "voice-only" communication between field forces. By its multimedia capabilities, MobileEmerGIS provides for example group communication services for simultaneous transmission of graphical data such as annotated maps accompanied by voice explanations to allow for "to-the-point" instructions to field forces. Based on real-life examples, such as a large-scale fire in a storage area, the concrete benefit of MobileEmerGIS will be highlighted.

2. MOBILEEMERGIS SYSTEM ARCHITECTURE AND SERVICES

The usage of standard technologies allows for a design of a technology platform, which is affordable to rescue organizations and civil institutions. The essential contribution on top of the standard components is a dedicated middleware software, which is able to hide the heterogeneity of different embedded platforms. At the same time, the MobileEmerGIS middleware needs to be flexible enough to implement applications owned by different rescue organizations involved in the process.

To fulfil the above listed requirements, the MobileEmerGIS system architecture builds upon the following key standard technology components:

- Usage of off-the-shelf mobile telephone and personal computer devices, optionally in ruggedized versions.
- Leveraging of available wireless multi-access network capabilities: at least WLAN and cellular networks including relay functionalities [1,2], WiMAX availability optional.

- Leveraging of integrated localization technologies (GPS, in future Galileo, the European satellite positioning system).
- Usage of Internet Protocols (IP) for data exchange and Web-based principles for service control (Service-oriented architecture).

In order to allow flexible application development by various organizations involved in rescue actions and disaster mitigation, the principles of service–oriented architectures (SOA) have been adopted for the design of the MobileEmerGIS middleware [3]. For an overview of the basic architecture and the implemented services, see Figure 2. In the following specific system characteristics are highlighted.

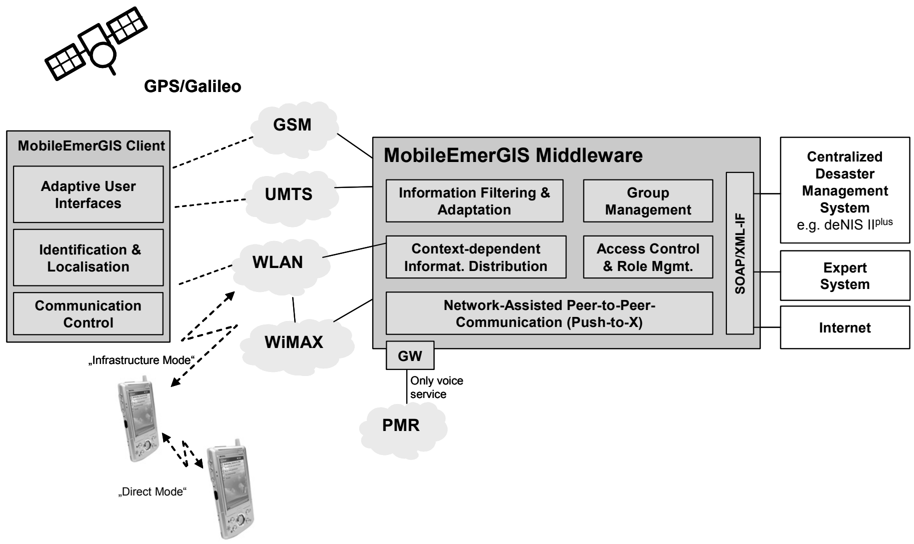


Figure 1. MobileEmerGIS System Architecture.

2.1 PTX Communication and Information Sharing

The MobileEmerGIS system relies on the IP-based Push-to-X system, which allows to share text, voice and video information across various wireless links in groups (see [4]).
The following key characteristics can be highlighted:

- The PTX approach relies on a network-assisted Peer-to-Peer communication paradigm. While a network server is involved to support the service and group management, once a communication group is established, the communication within the group will be performed

independently of the network server. Thereby potential outages of the network server during a rescue action will not lead to a breakdown of the communication as long as the physical communication links between the communicating peers are available.
- The PTX system relies on the Internet Protocol for interoperability with standard routers, but implements a highly efficient dedicated protocol for the signalling and payload transfer within the groups.
- The system is designed and has been experimentally proven to work across very heterogenous wireless links ranging from cellular networks (GSM/UMTS) across Wireless LAN to WiMAX-enabled data links.

2.2 Dynamic Group Management and Geo-Casting

For the coordination of rescue actions, it is often required to share information in larger groups of users or services because a one-to-one communication would be inefficient. In traditional analogue walkie-talkie or Push-to-Talk systems, the grouping of communication devices is realized by the use of different analogue channels: a user joins a communication group by switching his device to a certain channel, which is commonly shared by the members of the group. While this mechanism is simple and efficient, it has also some severe drawbacks, which can be overcome by a digitally enabled, IP-based system. With an IP-based system, it is first of all possible to share information not only by voice messages, but by text, images and other forms of multi-media content. Furthermore, the definition of logical groups can be performed much more differentiated and linked to the role model, i.e. the membership within a group can be restricted to certain roles within the organization.

The MobileEmerGIS group management enables to define logical groups of communicating entities according to various criteria. These groupings can be performed on a long-term basis (e.g. if they are linked to the hierarchy of an organization) as well as on an ad-hoc basis. Examples for grouping criteria supported by MobileEmerGIS are:
- Role, e.g. all members of a rescue team form one group TEAM-ABC
- Services, e.g. all services capable of measuring gas concentration levels are joined in one group TOXIC_GAS_SENSORS.
- Location, e.g. all rescue forces located around a potential explosion source form an ad-hoc group DANGER_ZONE_50m_RANGE.

Figure 2 shows an example how an ad-hoc group would be formed to convey an alarm message to rescue forces endangered by explosives.

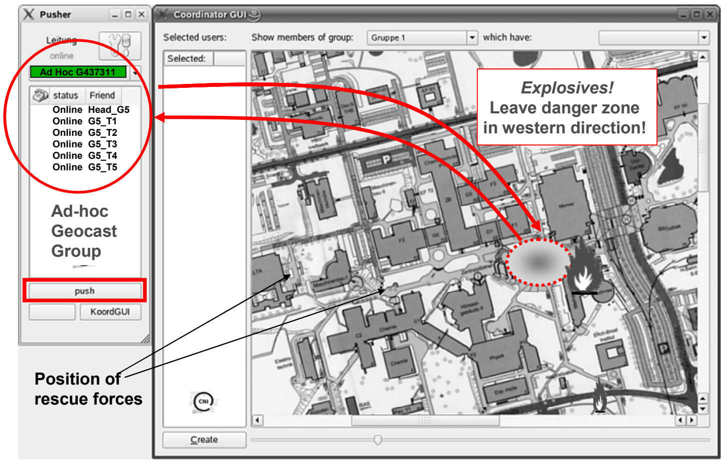


Figure 2. Example Geo-casting Service.

3. DROPPED UNITS CONCEPT

Within MobileEmerGIS, so-called Dropped Units allow for the creation of a highly reliable network, which enables rescue and security personnel to share multimedia data. They are carried by a scout, who has the special task to circle around the area, which is affected by an incident. During his walk he places Dropped Units to extend the range of the base access point, which is connected to the MEG server.

There are various possibilities to place Dropped Units. One simple algorithm would be to spread as many Dropped Units as possible allover a scenario and to locate them at random chosen positions. This procedure is inefficient and leads to undesired side effects such as interference and hidden station problems. Therefore adequate dropping rules are required, which lead to an optimal number of Dropped Units for a given scenario.

3.1 Coverage Analysis for ad-hoc wireless networks for emergency scenarios

The coverage of ad hoc wireless networks strongly depends on the topology of the analyzed scenario. The radio propagation characteristic of WLAN line-of-sight connections is completely different compared to the radio propagation within buildings. In situations, which require surveillance monitoring because of an emergency, often complex topologies occur. These heterogeneous environments include indoor and outdoor radio propagation, which also depends on the materials of the structures located within the scenario. For the performance analysis of ad hoc wireless networks, the knowledge about radio propagation and coverage of each WLAN access point is mandatory. To consider multiple influencing factors, a scenario is analyzed, which consists of heterogeneous materials within an indoor and outdoor environment.

In this paper, the campus of the Dortmund University of Technology serves as a use case scenario: An incident occurs at the chemical lab of the department of chemistry. This incident would threaten several gas cylinders full of highly explosive material. Several hundred people would have to be evacuated. To analyze the coverage of ad hoc wireless networks, a 3D-model of this scenario was created using a ray tracing tool, which computes the radio propagation of WLAN transceivers (see Figure 3). The base area of the chemical lab is 10m x 10m and the height is 5m. Its exterior walls are made of concrete (10 cm). Three boxes made of 20cm chipboards represent the chemical and equipment within the lab.

First of all, only one fix WLAN access point (AP) is included in the model. This AP is typically located at the command vehicle in front of the building. The WLAN antenna is normally mounted to the roof of the vehicle, which increases the coverage. The emitted radio power of the WLAN antenna is 20 dBm. The resulting network coverage is shown in Figure 3, top right. Only one third of the scenario is covered with sufficient signal strength for transmission of multimedia data (see darker areas in Figure 3).

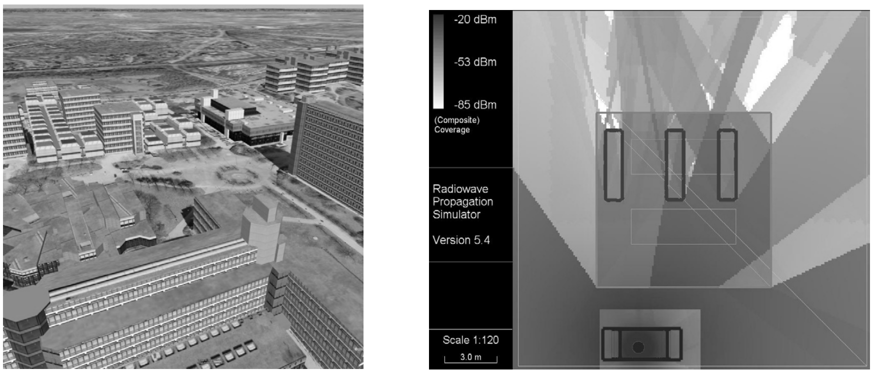


Figure 3. Coverage analysis using 3D ray tracing (left-3D scenario modeled with Google Earth/Sketch-up, right: receive power distribution around the chemical lab).

The excellent indoor penetration is due to a huge open gate at the front side. Areas with coverage below -85 dBm, which is the minimum signal strength defined in IEEE 802.11, are shown in white color. If a mobile node enters such an area, the connection will be terminated. For the transmission of surveillance data, high bandwidth and therefore signal strengths above -40 dBm are required. The signal strength falls below this threshold at the corners of the building which is called corner effect. The corner effect leads to the sudden loss of multimedia services of mobile nodes turning around corners.

3.2 Dropped Units System architecture

Our approach for a solution of the problem described above is the usage of Dropped Units or dynamically deployed wireless relays [5]. Dropped Units route the communication traffic to the server or other mobile nodes.

Due to the mobility requirements, Dropped Units are lightweight, battery driven devices. To perform their task as a wireless relay, two IEEE 802.11 b/g transceivers are integrated. One transceiver operates at 5 GHz creating the backbone of the network by meshing of the Dropped Units and the initial access point. The second transceiver operates at 2.4 GHz for the communication with moving or movable wireless clients. There are two types of movable clients: Surveillance devices like web cams, which upload their video streams to the officer-in-charge's client and first responder devices like PDAs and mobile phones which are used to share information between users.

The received signal strength and therefore the coverage gain depends on the height of the position a dropped unit is placed. The term Dropped Units seems to imply that the Units are dropped to the ground. Because of moving persons and vehicles within a scenario, a position on the ground leads to bad communication results. The best position for dropped units is on a windowsill or at least at the height of the windows within the scenario. Tripods can be used to place them in a sufficient way.

More important than the height of the drop point is the location within the scenario. Typically the topology of the scenario is unknown, prior to the incident. The position of the initial access point, which is mounted on the command vehicle, depends on the parking position. Furthermore environmental conditions influence the radio propagation. Therefore the prediction of an optimal dropping location is very complex. Without the knowledge about the optimal locations, the dropping points have to be selected by dynamic parameters. Examples for possible parameters are:

- Distance
- Special locations (doors, windows, building edges)
- Signal strength thresholds

In the following we use this simple rule: *Drop a unit if a well defined threshold of the signal strength is reached.*

4. MULTI-SCALE NETWORK SIMULATION FOR QUANTITATIVE PERFORMANCE EVALUATION

To map the complexity of the Dropped Units concept to different scenarios, a new kind of simulation environment is required. Focusing on single aspects may lead to an incorrect interpretation of the data. One way to avoid this difficulty is to include as many aspects as possible into one simulation has to be taken into account. Hence a wide range of parameter sets to control the behavior of the simulation. For a better control of the simulation, its complex structure is broken down using a divide-and-conquer algorithm, named the multi-scale approach.

In this approach the problem is divided into different aspects and each aspect is simulated by a dedicated best-in-class tool. The multi-scale network simulation environment (MNSE) integrates the whole tool-chain into one single environment (see Figure 4). The key advantage of the multi-scale approach is a wide range of network types which can be simulated. The scope of the simulation includes mobile sensor networks [7] as well as satellite networks [6]. To ease the scenario modeling, CNI has integrated Google Earth/Sketchup into a tool chain.

The simulation of the network protocols is the core of each simulated scenario. The behavior of the **network protocols** are influenced by the radio propagation model as well as the mobility model. A low SNR can lead to higher bit error rates and may cause multiple retransmissions of data packets. As a result to these dependencies, the main simulation control is integrated as a central event broker into the network simulation. It is implemented as an additional module to the OMNeT++ discrete event driven simulation environment, which is the basis of the MNSE. Additionally, the INET framework is used for the simulation of the standard protocols the network is based on.

The surveillance and communication network as described in this paper is implemented as an 802.11 network with a full TCP/IP stack. This allows for standard hardware for real test environments and verification of the simulated results.

The channel model is derived from the 3D raytracing, which is dynamically linked through socket communication with the protocol and mobility model.

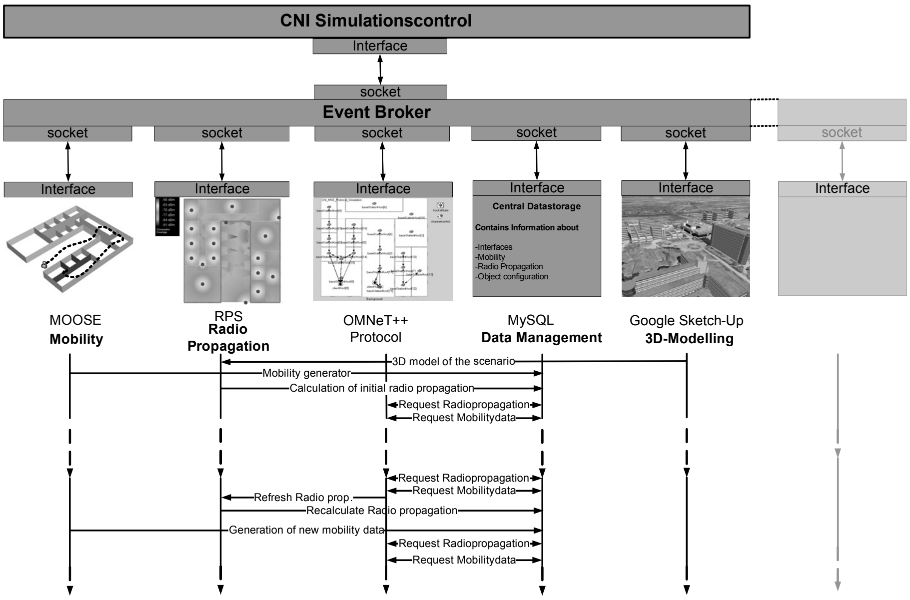


Figure 4. Multi-Scale Simulation Approach.

To get more realistic results, a specific **communication traffic model** of emergency communication is integrated as an extension in the application layer. It integrates a traffic generator for voice communication and for multimedia data, which is one precondition for the operation of surveillance networks.

The **mobility models** used for the simulations are based on real tactics of firefighters as well as on random mobility models. To elaborate a suitable mobility model, typical movement patterns have to be identified by analyzing emergency response teams' tactics. For this simulation, two different mobility patterns are used (see Figure 5):

- In the beginning of a rescue mission, a scout walks around the operational area to identify risks and to build up a reliable communication network if it is necessary. This is modeled by a fixed path in the scenario to ensure that the movement is not influenced by random elements. After finishing the first circle, the scout changes his behavior to a random model. This behavior is included in the simulation for at least one crew member.
- All other crew members do random movements to model the scenario specific operations.

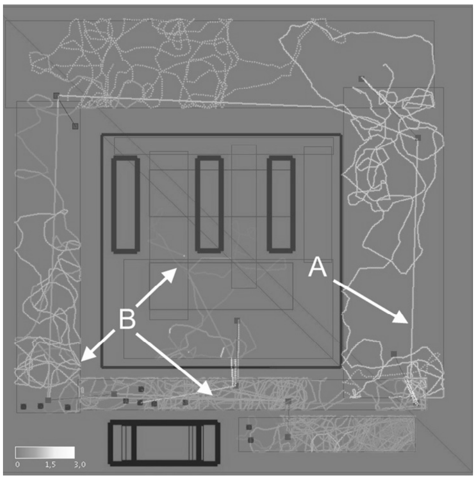


Figure 5. Emergency Response specific mobility model: Traces of scout (A) and crew members (B).

Depending on the exact movements, handovers between different Dropped Units are necessary to establish communications between the members.

5. RESULTS OF PERFORMANCE EVALUATION

To get the results, the following parameters were used as input for the multi-scale simulation: radio propagation, movement and protocol stack. For the radio wave propagation model, a single laboratory building without neighboring buildings was modeled. The movement of firefighters was divided into three heterogeneous classes: scout, crew and officer-in-charge. IP over IEEE 802.11 b/g at 2.4 GHz was used as the protocol stack.

The scout has the task to explore the situation and therefore moves around the building: he places MobileEmerGIS Dropped Units (MDUs) if a certain threshold of the wireless network signal strength indicator is reached. After circling the building, the scout created a network which covers the whole area. The remaining firefighters are not responsible for creating the network, but simply share multimedia data via the wireless ad-hoc network.

Table 1. Simulation parameters.

Scenario size	20m x 20m
Channel model resolution sending	1 Sender per m²
Channel model resolution receiving	4 Receiver per m²
Number of fire fighters	9
Transmission Power	20 dBm
Antennas	Dipole with 2 dB gain

Four variants of this scenario with zero to three MDUs in use were simulated. All simulations have been repeated 10 times for each version with one hour duration. The statistical analysis of the availability of the network for the scout results in the graph presented in Figure 6 (left): the connection outage time decreases with an increasing number of MDUs from 377 sec. (no MDU, 89.53% availability) to 3 sec. (three MDUs, 99.92% availability) within a simulation period of 3600 seconds.

The network availability for one crew member is presented in Figure 6 (right). One additional MDU causes only a small increase of the availability. If all units are dropped the network availability for one crew member compared to the scout is the same. The lack of direct correlation of the scout's results compared to the crewman's results is reasonable, because of the fact that the scout carries the Dropped Units and therefore directly gets an improvement of the network availability.

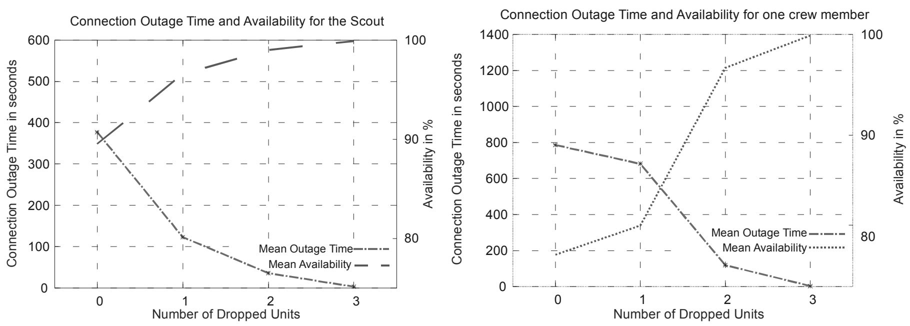


Figure 6. Simulation results (left-scout, right-crew member).

6. CONCLUSIONS AND OUTLOOK

The MobileEmerGIS project has developed and validated new concepts to support field forces in protecting critical infrastructure. In this paper we have therefore presented the architecture and exemplary services of a wireless-enabled system, which allows to aggregate and to distribute information collected by eye witnesses and field forces in a highly efficient and intuitive way. The system leverages off-the-shelf hardware and integrates multimedia and geo-positioning information provided by mobile devices. We have presented the Dropped Units and their abilities to support the dynamic set-up of wireless local networks. The simulation results lead to the conclusion that with a limited number of Dropped Units a practically 100% network availability can be achieved. For the investigated scenario three dropped units are sufficient to cover the whole scenario.

The investigated scenario provides a first indication of the expected abilities of Dropped Units. Larger scale scenarios will be analyzed in the future as well as the incorporation of unmanned arial vehicles (UAVs) to serve as low-altitude relay points.

ACKNOWLEDGEMENTS

The author Christian Wietfeld would like to sincerely thank Prof. Dr.-Ing. Bernhard Walke for being a source of great inspiration and for lending enduring and commendable support and guidance. It is indeed a great honor to be a part of ComNets community.

References

[1] B. Walke, Mobile Radio Networks, Networking and Perfromance, Wiley, 2001.
[2] R. Pabst, B. Walke et al., Relay-based Deployment Concepts for Wireless and Mobile Broadband Radio, IEEE Communications Magazine, September 2004.
[3] C. Wietfeld, A. Wolff, Uli Bieker, MobileEmerGIS: a wireless-enabled technology platform to efficiently support field forces in protecting critical infrastructure, 2007 IEEE Conference on Technologies for Homeland Security, Woburn, USA, May 2007.
[4] J. Seger, A. Wolff, C. Wietfeld, Analysis of IP-based Real-time Multimedia Group Communication in Heterogenous Wireless Networks, International Symposium on Wireless Communication Systems, Valencia, Spain, September 2006.
[5] Michael R. Souryal et al: Real-time deployment of multihop relays for range extension, MobiSys '07: IEEE 5th international conference on Mobile systems, applications and services, San Juan, 2007
[6] A. Lewandowski, R. Burda, C. Wietfeld: A Multiscale Real-Time Communication and Navigation Satellite Simulation Model for OMNeT++, 1st International OMNeT++ Workshop on the SIMUTools Conference, Marseille, March 2008
[7] R. Burda, A. Lewandowski, C. Wietfeld: A Hybrid Indoor Localization Approach combining Meshing and Time of Arrival in IEEE 802.15.4 Networks, IEEE Vehicular Technology Conference, Singapore, May 2008.
[8] C. Wietfeld, A. Wolff, S. Subik: Performance analysis of highly available ad hoc Surveillance Networks Based on Dropped Units, 2008 IEEE Technologies for Homeland Security Conference, Boston, MA, USA, May 2008.

ScaleNet - KONVERGENTE NETZE DER ZUKUNFT

Bangnan Xu

Abstract Dieser Artikel gibt einen Überblick über DTAG Teilvorhaben des BMBF Projektes ScaleNet. Eine neue konvergente Netzarchitektur und neue Netzfunktionen, wie Mobilitätskonzepte, werden vorgestellt. Des Weitern wurden Konzepte für die Carrier-Grade Mesh-Netze präsentiert. Schließlich wird das in Zusammenarbeit mit Prof. Walke entwickelte neue Verfahren zur Spektrumsbedarfsschätzung und flexiblen Spektrumsnutzung beschrieben. Meine Danksagung beendet diesen Artikel.

1. EINLEITUNG

Im DTAG Teilvorhaben des BMBF Projektes ScaleNet wurden die Netzkonvergenz und Dienstkonvergenz untersucht. Unter Netzkonvergenz versteht man die Verwendung von zukünftigen, flexiblen Zugangsnetzen sowohl von Mobilfunknetzbetreibern als auch von Festnetzbetreibern. Dabei erfolgt die Aggregation auf einer einheitlichen, IP-basierten Struktur mit entsprechenden Netzknoten an deren Rand. Diese Netzknoten ermöglichen die Integration verschiedenster „First Mile"-Zugangstechniken. Dazu sind neue Steuerungsmechanismen notwendig, um die heterogenen Zugangstechnologien flexibel zu integrieren. Die neuen Steuerungsmechanismen sollten übergreifende Netzfunktionalitäten zur Verfügung stellen, um QoS, Mobilität, AAAC (Authentication, Authorization, Accounting and Charging) und Sicherheit für den Endnutzer zu ermöglichen. Zur Realisierung von Dienstkonvergenz wird IMS (IP Multimedia Subsystem) als eine Kernkomponente betrachtet.

Mobilität ist eine der wichtigsten Eigenschaften der zukünftigen Netze. In ScaleNet sollten vier Mobilitätstypen, d.h. „Terminal Mobility", „Session

Mobility“, „Service Mobility“ und „Personal Mobility“ unterstützt werden. IMS-gesteuerte Session-Mobilität bietet eine neue Dimension für die Mobilitätsunterstützung und bildet deshalb einen Untersuchungsschwerpunkt in ScaleNet.

Als eine der bedeutenden zukünftigen mobilen/drahtlosen breitbandigen Zugangstechniken werden in ScaleNet Multihop Mesh-Netze intensiv untersucht.

Da Spektrum eine bedeutende und begrenzte Ressource für einen Netzbetreiber ist, wurde eine Zusammenarbeit zwischen DTAG und dem Lehrstuhl für Kommunikationsnetze, Prof. Dr.-Ing. Walke, initiert, um den Einfluss neuer Entwicklungen wie Netzkonvergenz, neue Dienste und neue Zugangstechnologien auf den Spektrumsbedarf abzuschätzen und neue Methoden zur Einschätzung des Spektrumsbedarfs und zur effizienten Nutzung des Funkfrequenzspektrums zu entwickeln.

2. DIE KONVERGENTE NETZARCHITEKTUR VON ScaleNet

Die in Abbildung 1 dargestellte ScaleNet-Netzarchitektur besteht aus den Hauptelementen “Converged Access Aggregation Network“ (CAAN), “Universal Access Node“ (UAN), “Access Border Controller” (ABC) und “Converged Control Layer” (CCL). Der UAN realisiert die generischen Funktionen, die für die Adaption heterogener Zugangstechniken an das CAAN erforderlich sind. Der ABC ist eine lokale, generische Network-Edge-Funktion, die, in Zusammenarbeit mit dem UAN, die ScaleNet-Ende-zu-Ende-Konzepte für QoS (Quality of Service), AAA (Authentication, Authorization and Accounting), Mobilität (Mobility) und Sicherheit (Security) umsetzt. Das CCL ist eine generische Steuerungsebene, die in wesentlichen Teilen an die 3GPP-IMS Spezifikationen angelehnt ist. Sie ermöglicht Funktionen zur konvergenten Steuerung von IP-Diensten und stellt standardisierte Schnittstellen (APIs: Application Programming Interfaces) zur Anwendungsplattform bereit. Das CAAN ist ein Transportnetz, das den UAN und Access Edge Router verbindet. Das CAAN soll kosteneffiziente Aggregation und Netzkonvergenz durch einheitliche Methoden für Verkehrsweiterleitung (traffic forwarding), QoS handling und AAA ermöglichen.

Der “Access Border Controller” (ABC) ist ein wichtiges Netzelement im CAAN, das Mobilität, QoS, AAA und Sicherheit in der konvergenten Netzarchitektur gewährleisten soll. Der ABC verfügt über folgende Funktionen:

- Steuerung der Signalisierung und des Datenstroms

- IP Mobility Management
- Ressource and Admission Control (RAC)
- Interaktion mit der Converged Service Control Plane - ETSI TISPAN/3GPP IMS

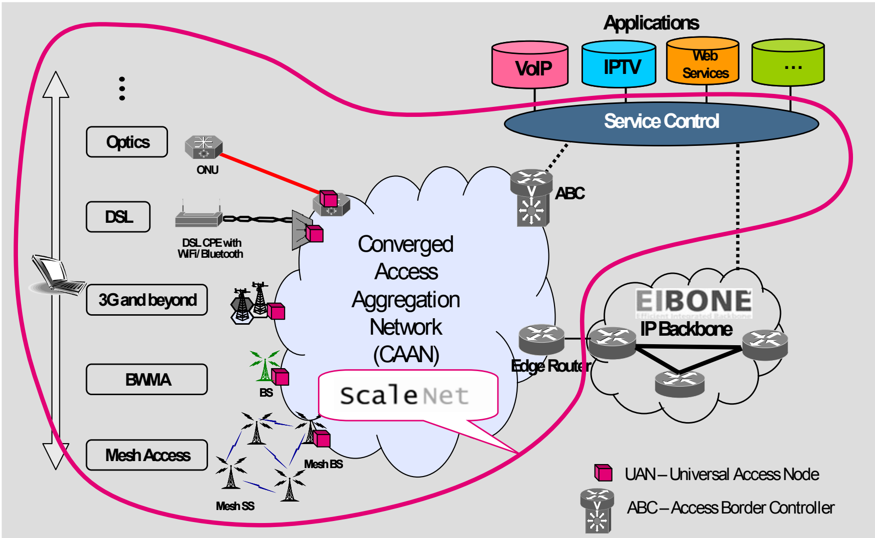


Abbildung 1. Die konvergente Netzarchitektur von ScaleNet.

Die Funktionalität eines Converged Service Control Layer wird wesentlich zur Service-Konvergenz beitragen. Eine Converged Service Control Plane erlaubt die schnelle Einführung neuer Dienste. Das Converged Service Control Layer bietet generische Funktionen, wie z. B. Session Control, AAA, Sicherheit, QoS etc., die für viele existierende und zukünftige Anwendungen von großer Bedeutung sind.

3. INTEGRATION DER MOBILFUNKNETZE DER NÄCHSTEN GENERATION (NGMN, 3GPP SAE/LTE)

Mobilfunknetze der nächsten Generation (NGMN, Next Generation Mobile Network), wie sie von 3GPP im Rahmen der Architektur und Technologiekonzepte für Release 8 standardisiert werden (SAE/LTE: System Architecture Evolution/Long Term Evolution) sollen sich vor allem durch geringere Bereitstellungskosten pro übertragener Dateneinheit und hochwertige Dienstangebote auszeichnen. Verringerte Betriebskosten und

Investitionen werden durch eine einfachere Architektur, flexiblere Nutzung der Funkfrequenzen, offene Schnittstellen und eine – auf paketvermittelte Datenübertragung optimierte - effizientere Funktechnik erreicht.

Das Prinzip einer auf zwei Elemente reduzierten Architektur ('Two node architecture': the evolved NodeB and the access Gateway, eNodeB+aGW) erlaubt kürzere Verzögerungen, vereinfachten Netzaufbau, bessere Skalierbarkeit und unkompliziertere Nutzung des Transportnetzes.

Dieser Ansatz wird durch die ScaleNet-Netzarchitektur mit dem konvergenten Aggregationsnetz (CAAN) unterstützt, in dem das UAN eine Schnittstelle zum eNodeB besitzt und der aGW an den ER (Edge Router) angeschlossen ist. Langfristig sind eNodeB und aGW in UAN bzw. ER integriert und die von 3GPP spezifizierte Schnittstelle S1 wird im auf GPON basierenden CAAN effizient übertragen (siehe Abbildung 2).

Sowohl für die Schnittstelle S1 als auch für die zwischen einzelnen eNodeBs vorgesehene X2-Verbindung würde eine Verzögerung des Transportnetzes CAAN von etwa 1 ms innerhalb der spezifizierten Werte liegen. Das CAAN sowie das IP/MPLS-basierte Backbone werden sowohl von NGN als auch von NGMN gemeinsam genutzt, womit weitere Kosteneinsparungen ermöglicht werden. Für den Transport der Nutz- und Steuerungsdaten (user and control plane) zwischen eNodeB und aGW werden bei 3GPP derzeit verschiedene Versionen des Internet Protokolls (IPv4 und IPv6) diskutiert. Entsprechend der endgültigen Festlegung wird eine Unterstützung für diese Funktionalitäten von den CAAN-Knoten bereitgestellt werden.

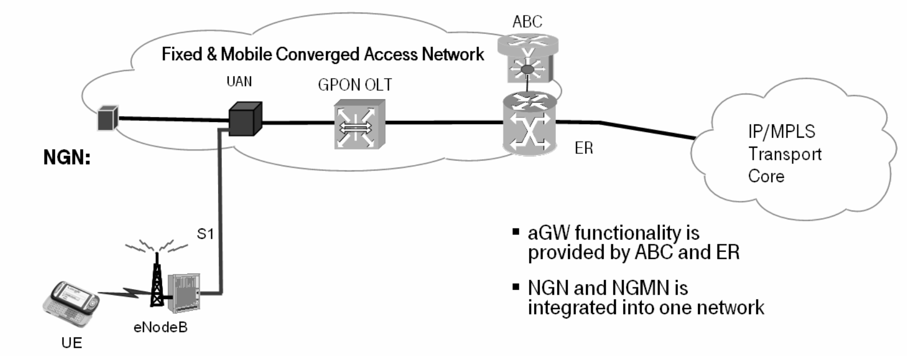


Abbildung 2. Integration der Mobilfunknetze der nächsten Generation in die ScaleNet Architektur.

4. MOBILITÄTSKONZEPTE FÜR DIE KONVERGENTE NETZARCHITEKTUR

In den konvergenten Netzen sind übergreifende Netzdienste zur Unterstützung von Mobilität, zur Bereitstellung von Ende-zu-Ende QoS, AAAC und Sicherheits-Mechanismen erforderlich. Im Projekt ScaleNet werden vier Mobilitätstypen – Terminal/Session/Personal/Service Mobility - untersucht. Konzepte für IMS-basierte Session- und Terminal-Mobilität wurden entwickelt. Ein Schwerpunkt der IMS-basierten Session-Mobilität liegt darin, dass die laufenden Sessions von Multimedia Triple Play-Diensten individuell und nahtlos zwischen Endgeräten umgeschaltet werden können. Um die IMS-basierte nahtlose Terminal-Mobilität zwischen verschiedenen Zugangsnetzen zu realisieren, ist es notwendig die PDF (Policy Decision Function) zu erweitern.

Ein wesentlicher Bestandteil der „Systemübergreifenden Netzdienste" zur IMS-gesteuerten Dienstekonvergenz (Service Convergence) ist Mobilitätsmanagement. Ein neues Verfahren zur Unterstützung der Terminal-Mobilität für Ethernet-basierte Infrastruktur innerhalb des konvergenten Zugangsnetzes wurde entwickelt. Das auf dynamische Zuordnung zwischen IP und MAC Adresse basierte patentierte Verfahren [6] ermöglicht die Verwendung einer einzigen IP-Adresse für alle Netzschnittstellen eines mobilen Knoten, die an das gleiche konvergente Zugangsnetz angebunden sind. Bei einem Handover zwischen zwei heterogenen Zugangspunkten muss somit die IP-Adresse des mobilen Knotens nicht gewechselt werden, was die Verwendung zeitaufwändiger Mobilitätsprotokolle der Netz- bzw. höherer Schichten erspart. Aufgrund der schnelleren Bearbeitung der Mobilität auf Layer 2 kann ein nahtloser Handover-Prozess realisiert werden, was für die multimedialen Netzdienste besonders wichtig ist. Außerdem ist die optimierte Auswahl eines Zugangspunktes für mobile Endgeräte mit mehreren Netzschnittstellen sehr wichtig, um die angeforderte Dienstgüte (QoS) für die Applikationen zu erfüllen. Der auf Layer 2 basierte Routingmechanismus ermöglicht gleichzeitige Übertragung der Datenpakete verschiedener Datenverbindungen über verschiedene Netzschnittstellen. Dazu wurde ein neues, QoS-bewusstes Mobilitätssystem entwickelt, das die Auswahl der jeweils besten Zugangspunkte für Datenverbindungen mit verschiedenen QoS-Anforderungen ermöglicht. Das entwickelte Mobilitätssystem ermittelt QoS-Ressourcen, die für einen spezifischen mobilen Knoten an verschiedenen Zugangspunkten zur Verfügung gestellt werden können. Aufgrund dieser Information trifft das Mobilitätssystem Entscheidungen, welche Netzschnittstellen für welche Datenverbindungen genutzt werden müssen, um das angeforderte QoS-Niveau für Applikationen bereitzustellen. Die vorhandenen Transportressourcen werden nach dem Aufbau einer Session überwacht. Somit kann das QoS-bewusste Mobilitätssystem Handover initiieren und neue Zugangspunkte für Datenverbindungen, deren Qualität sich verändert, selektieren. Die Untersuchungen der Skalierbarkeit und der Performanz des neuen Systems haben gezeigt, dass mit deren Hilfe eine deutliche Steigerung der Dienstgüte erzielt werden kann.

5. CARRIER GRADE MESH-NETZE

Als eine der bedeutenden zukünftigen mobilen/drahtlosen breitbandigen Zugangstechniken werden in ScaleNet Multihop Mesh-Netze intensiv untersucht.

Durch Untersuchung der vorhandenen Mesh-Standards bezüglich deren „Carrier Grade“ Fähigkeiten wurde herausgefunden, dass existierende Lösungen für QoS, AAA sowie bestehende Sicherheitsmaßnahmen nicht ausreichen, um die Anforderungen eines Carrier Grade Mesh-Netzes zu erfüllen. Deshalb wurden in ScaleNet QoS-Mechanismen entwickelt, um die Qualität von Echtzeitanwendungen in Mesh-Netzen (MN) zu verbessern. Hier kann zwischen Mechanismen für IEEE 802.11-basierten and IEEE 802.16-basierten MNs unterschieden werden. Für IEEE 802.11 MNs wurden Schicht-3 basierte Mechanismen entwickelt, da Änderungen in der MAC-Schicht aufgrund der weiten Verbreitung der WLAN-Technologie nicht sinnvoll erscheinen. Der hier verfolgte Ansatz basiert im Wesentlichen auf zwei Aspekten: (1) Qualitätsbewertung aller Echtzeitdienste auf jedem Mesh-Knoten und (2) gezielte Bearbeitung von Datenströmen, um eine Qualitätsanforderung der Echtzeitdienste zu erfüllen.

Für IEEE 802.16 MNs wurden Schicht-2 basierte Scheduling-Mechanismen untersucht und entwickelt, die in der Lage sind, Delay, Jitter und Paketfehlerrate niedrig zu halten, um eine „gute“ Sprachqualität zu gewährleisten. Auch im Bereich der TDMA-basierten QoS-Mechanismen wurden Scheduling-Verfahren erarbeitet, die in der Lage sind, die Qualität von VoIP-Verkehr in MNs auf einem hohen Niveau zu halten. Dank der entwickelten QoS-Mechanismen erreicht die VoIP-Qualität bei einer großen Anzahl der Hops zwischen den Endgeräten und der Basisstation noch sehr gut [4]. Weiterhin wird auch der negative Einfluss von breitbandigem Datenverkehr auf die VoIP-Qualität behoben.

6. SPEKTRUMSBEDARFSSCHÄTZUNG UND FLEXIBLE SPEKTRUMSNUTZUNG

Eine Voraussetzung für den Netzbetreiber zur Realisierung von „Broadband Everywhere“ ist die Verfügbarkeit von Spektrumsressourcen. In Zusammenarbeit mit dem Lehrstuhl von Prof. Walke wurde eine neue Methode zur Einschätzung des Spektrumsbedarfs für ScaleNet entwickelt, die zur Ermittlung des Spektrumsbedarfs von zukünftigen konvergenten Netzen eingesetzt werden kann. Da die Verfügbarkeit des Spektrums sehr begrenzt ist, sind neue Konzepte zur effizienteren Nutzung der Spektrumsressourcen von großer Bedeutung. Dazu werden im Rahmen des Vorhabens neue Verfahren zum „Spectrum-Sharing“ untersucht.

Zur flexiblen Spektrumsnutzung ergeben sich gerade bei OFDM-basierten Funktechniken (802.11g, 802.16, 802.16e, zukünftig UMTS-LTE) mehrere Möglichkeiten mit einer zwischen den Basisstationen abgestimmten Nutzung des Spektrums, die Signalrauschabstände und damit die nutzbaren Datenraten und QoS-Eigenschaften zu verbessern. Zur Leistungsbewertung wurden die Simulationen durchgeführt. Dabei wurden 7 Basisstationen, die jeweils mit 3 Relay-Stationen ausgerüstet sind sowie eine Reihe zugehöriger Endgeräte simuliert. Es kam der Systemsimulator openWNS vom Lehrstuhl Kommunikationsnetze (Prof. Dr.-Ing. Bernhard Walke) auf einem Computer-Grid basierend auf Sun-Grid zum Einsatz. Die Verfahren wirken vor allem in Situationen mit niederer oder mittlerer Last bzw. mit unterschiedlichen Lastverteilungen zwischen den Zellen. Der Gewinn liegt dabei weniger im erhöhten Datendurchsatz als vielmehr in einer verminderten Paketverzögerung (um bis zu 100 ms) oder im günstigeren Signal-Rausch-Abstand.

Durch den verbesserten Signal-Rausch-Abstand können die Sendeleistungen bei den Endgeräten und Basisstationen reduziert werden. Die Zeitspannen des aktiven Betriebs werden kürzer. Dies führt zu längeren Akku-Laufzeiten bei den Endgeräten und Leistungseinsparungen bei der Basisstation. Die Nutzungszeiten des Spektrums verkürzen sich und ermöglichen einen vorübergehenden Transfer der Frequenzen auf andere Systeme oder Zellen. Für SLS (Spectrum Load Smoothing) ist die Kenntnis über das Sendeverhalten der anderen Basisstationen nötig. Innerhalb von ScaleNet liegt der Schwerpunkt dabei auf Verfahren, die auf einer gegenseitigen Bebachtung der Basisstationen beruhen, ohne Nachrichten zwischen den Basisstationen auszutauschen.

7. DANKSAGUNG

Das Projekt ScaleNet DTAG Teilvorhaben wird in Zusammenarbeit mit Universitäten und Hochschulen durchgeführt. Ich bedanke mich deshalb bei allen Kollegen, Doktoranten, Diplomanden und Praktikanten, die im Projekt ScaleNet hervorragende Leistungen erbracht haben.

Meine Dank gilt insbesondere Herrn Prof. Walke für seine außergewöhnliche Beiträge auf dem Gebiet der Mobilfunknetze. In Zusammenarbeit mit Lehrstuhl Kommunikationsnetze wurde in Projekt ScaleNet das neue Verfahren zur Spektrums-bedarfsschätzung und flexiblen Spektrumsnutzung entwickelt, das eine große Bedeutung für Netzbetreiber hat. Bei dieser Gelegenheit bedanke ich mich auch für seine freundliche und wertvolle Betreuung meiner Promotionszeit am Lehrstuhl, die es mir ermöglicht hat, ein komplexes Forschungsprojekt wie ScaleNet erfolgreich zu leiten.

References

[1] B. Walke, Mobile Radio Networks: Networking, Protocols and Performance, Wiley, 2001.

[2] Nico Bayer, Dmitry Sivchenko, Bangnan Xu, Joachim Habermann, Veselin Rakocevic. „Transmission timing of signaling messages in IEEE 802.16 based Mesh Networks“, European Wireless 2006, Athen, Greece, 04/2006

[3] M. Siebert, B. Xu, E. Weis, B. Bayer, D. Sivchenko, T. Banniza, K. Wünstel, S. Wahl, R. Sigle, R. Keller, A. Dekorsy, M. Bauer, M. Soellner, J. Eichinger, C. Gan, F. Pittmann, R. Kühne, M. Schläger, I. Baumgart, R. Bless and S. Stefanov, "ScaleNet - Converged Networks of the Future", Information Technology. vol. 48, pp. 5, October 2006.

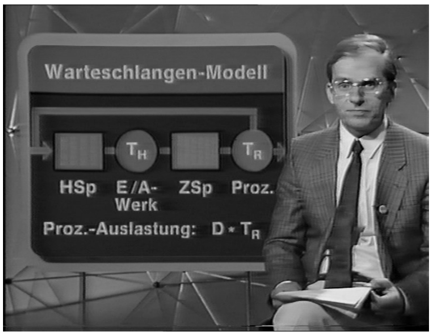
[4] Nico Bayer, Andreas Roos, Roger P. Karrer, Bangnan Xu, Christian Esteve, “Towards Carrier Grade Wireless Mesh Networks for Broadband Access”. “First IEEE International Workshop on Operator-Assisted (Wireless Mesh) Community Networks 2006”, Berlin, Germany. 09/2006.

[5] Nico Bayer, Bangnan Xu, Joachim Habermann, Veselin Rakocevic, “Improving the Performance of the Distributed Scheduler in IEEE 802.16 Mesh Networks”, 65th IEEE Vehicular Technology Conference 2007 Spring (VTC ’07-Spring). Dublin, Ireland. April 2007.

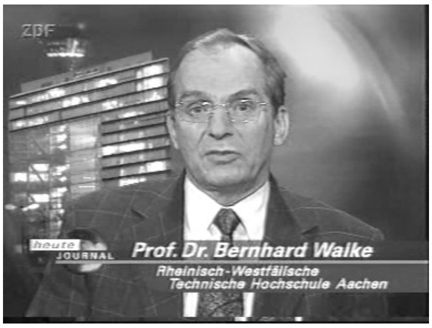
[6] Eugen Mikoczy, Dmitry Sivchenko, Bangnan Xu, Jose Ignacio Moreno, “IPTV services over IMS - Architecture and standardization”, IEEE Communications Magazine on “IPTV Systems, Standards and Architectures“, May 2008, Vol. 46, No. 5, pp. 128-135.

[7] Dmitry Sivchenko, Bangnan Xu “A new flow based Layer 2 handover mechanism for mobile node with multi network interfaces” Patent application EU07106360.6, US 12/104,803.

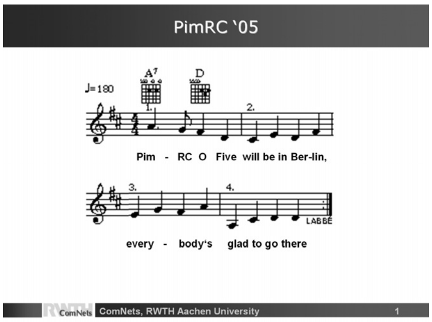
BILDER



Zu seiner Zeit an der Fernuniversität Hagen: Professor Walke in einer Videovorlesung über „Rechnerorganisation und Systemleistung“



Im heute Journal erklärt Professor Walke ganz Deutschland die Vorzüge von UMTS



Nicht mit Worten sondern mit einem Kanon wirbt Professor Walke für die PIMRC 2005 in Berlin

Auch sportlich erfolgreich: Zum dritten Mal gewinnt ComNets in 2005 das Fakultäts Fußball-turnier. Es sollte nicht das letzte Mal sein.

Besser spät als nie: Bei seiner Emeritierungsfeier erhält Professor Walke endlich seinen Doktorhut.

Wieder einer fertig: Weit über 50 Mitarbeiter führte Professor Walke erfolgreich zur Promotion. Viele werden hof-fentlich noch folgen

PROJEKTLISTE

Im Folgenden eine Auflistung vergangener und gegenwärtiger Projekte bei ComNets. Trotz beachtlicher Länge erhebt diese Auflistung keinen Anspruch auf Vollständigkeit.

Kürzel	Projekttitel	Zeitraum
Mobilise	Mobilise	01/1992-12/1994
Metwerk	Verbundprojekt Metwerk: Entwicklung und Erprobung neuer Methoden und Werkzeuge zur simulativen Analyse und Synthese komplexer Kommunikationssysteme	05/1995-04/1998
THESEUS	Terminal at High Speed for European Stock Exchange Users	09/1995-08/1998
RTT	Road Transport and Telematics RTTT	01/1996-07/1996
VaSCo	Validation of Dedicated Short-Range Communication	01/1996-05/1998
ATMmobil	Breitbandige Mobilkommunikation für Multimedia auf ATM-Basis	03/1996-03/2000
DCA	Kooperation von Mannesmann Mobilfunk und ComNets im Bereich DCA/HCA für GSM	06/1996-06/1999
Schnelle Netze	Nutzung und Entwicklung der Datenübertragung mit Hoch- und Höchstgeschwindigkeit (Schnelle Netze)	06/1996-06/1999
ATM	Zellulares ATM-Netz "Spezifikation mit prototypischer Implementierung von Protokollen für ein mobiles Endgerät"	07/1996-05/2000

SAMBA	System for Advanced Mobile Broadband Application	07/1996-10/1999
Mobilkom	Signalisierungssoftware entsprechend der Implementierungsvereinbarung des "Public User Network Inerface" (UNI-3.1) des ATM-Forums	08/1996-08/1997
InSUR	Integrated Satellite UMTS Real Environment Demonstrator	09/1996-07/1998
Motiv	Motiv: Sichere Straße / Fahrzeug-Kommunikation	01/1997-12/1998
Commerce I & II	Evaluation of Electronic Commerce Schemes in Mobile Internet	03/1997-03/1999
DKM	Dynamische Kanal- und Mobilitätsverwaltung paketorientierter Mobilnetze	08/1997-07/1999
Top-Nets	Redundanter Topologieentwurf	11/1997-10/1999
CAMELEON	ACTS Communication Agents for Mobility Enhancements in a logical Environment of Open Networks	03/1998-02/2000
Accord	ACTS/ AC348 / ACCORD ACTS Braodband Communication joint Trials and Demonstrations	04/1998-01/2000
A1	Interoperability of EU EFC Systems Based on DSRC	05/1998-04/2000
TDMA	Entwicklung von Konzepten für zentral gesteuerte TDMA-Vielfach-Zugriffsverfahren für ATM-basierte, drahtlose Breitband-Teilnehmeranschlußnetze und ATM-basierte Kommunikationsnetze auf Starkstromleitungen	05/1998-11/1998
CDMA-Schutz	Schutzanforderungen zwischen CDMA-Systemen und im Spektrum benachbarter Systeme	06/1998-12/1998

UMTS-L23	Untersuchung eines schnellen Signalisierungsmechanismus für UMTS Datendienste und Entwicklung einer Softwarearchitektur für die UMTS L23-Implementierung	07/1998-12/1999
UMTS Freq.	Entwicklung von Vorschlägen zur effizienten Nutzung des UMTS-Frequenzbandes durch UTRA-FDD und -TDD Systeme	08/1998-03/2002
Adapt. Term.	Layer 2 u.3 Protokolle für adaptive Terminals	09/1998-09/2000
GPRS-GSM	GPRS/GSM Kapazität und Dienstgüte des GPRS-Dienstes bei Koexistenz mit anderen GSM-Diensten	01/1999-09/2000
STAR	Standardization of Inter-Operable Road Tolling Systems based on DSRC	01/1999-03/2000
TETRA	Vergleich der Bündelfunksysteme Tetrapol und Tetra 25 in technich-wissenschaftlicher Hinsicht unter Berücksichtigung der taktischen und betrieblichen Anforderungen der BOS	03/1999-10/1999
Wireless Shop	Development and Evalution of Wireless Shopping Applications and Scenarios using Bluetooth and RFID Technology	03/1999-12/1999
ComCar	Protocols for the Coordination and Usage of Different Mobile Systems in Distributed Frequency Bands: Protocols, Algorithms and Simulation of System Components	04/1999-03/2002
Sorbas	Funkverträglichkeit, Protokolle, Leistungsbewertung von Systemkonzepten	04/1999-03/2002
UTRA/DCA	Entwicklung von Dynamischen Kanalvergabeverfahren für Utra/TDD	08/1999-03/2000

Powerline	Protokollentwicklung und formale Spezifikation in SDL der Medienzugriffs- und Sicherungsschicht für die Vermittlung von IP und VoIP in Punkt- zu Mehrpunkt-Systemen auf Niederspannungs-Energieversorgungsleitungen	09/1999-08/2000
Anette	Netzinfrastruktur und Software-Tools für Teleteaching und HomeLearning	10/1999-09/2001
Multifunk	Hypernet/Multifunk "Untersuchung des multivalenten Nutzbarkeit des Rundfunkbereiches durch dynamische Frequenz- und Bandbreitenzuweisung"	11/1999-12/2002
NetCologne	Wissenschaftliche Begleitung des Betriebsversuchs Powerline Communication	11/1999-02/2000
HSCoVerifikation	Hardware-Software-CoVerifikation komplexer Telekommunikationssysteme	11/1999-10/2001
Brain-Mind	Entwicklung eines System Level Simulators und von Vorschlägen zur effizienten Nutzung des Hiperlan Type 2 Standards für ein breitbandiges zellulares Mobilfunksystem	12/1999-09/2001
GPRS	Radio Network Dimensioning for GPRS in Coexistence with Circuit Switched Services in GSM	12/1999-11/2000
@dwise	Advanced Wireless Services	01/2000-12/2000
DELTA	DSRC Electronics Implementaion for Transportation and Automotive Applications	01/2000-07/2002
Ford	Untersuchung von Systemlösungen für Parkingmanagement	01/2000-04/2000
Smartshopping	Development and Enhancement of the Ericsson SmartShopping Demonstrator System	01/2000-06/2000

VirtUouS	Virtual Home UMTS on Satellite	01/2000-06/2002
GPRS	Dienstgüte-Verwaltung in GPRS-Netzen	02/2000-01/2001
Brain	Optimierung der Linkadaption für HIPERLAN/2	04/2000-03/2001
DRIVE	Dynamic Radio for IP-Services in Vehicular Environments	04/2000-03/2002
Multihop	Förderschwerpunkt hyperNET (universelle Nutzung von Kommunikationsnetzen für künftige Mobilfunkgenerationen)	05/2000-05/2003
WSI	The Wireless Strategic Initiative	05/2000-12/2002
SGOOSE	Entwicklung eines Werkzeugs zur kombinierten Simulation von Funkfeldausbreitung und Mobilität von Funkterminals in wählbaren Szenarien sowie von Protokollabläufen der Funkschnittstellen GSM/GPRS bzw. EDGE	06/2000-12/2000
Coverage-Multihop	COVERAGE-Cellular OFDM Systems with Extension Points for Increased Transmission Range	07/2000-06/2003
Multimedia-Mobil	Continuous Media Use with Discontinuous Wireless Data Transmission for Mobile Terminals	07/2000-09/2000
CoCoNet I-III	Adaptivität in heterogenen Kommunikationsnetzen mit drahtlosem Zugang I-III	08/2000-07/2006
Irma	Entwurf effizienter Verfahren zur systemübergreifenden Resourcenverwaltung für UMTS/HIPERLAN/2	08/2000-07/2001

Samo	Entwicklung eines Werkzeugs zur kombinierten Simulation von Funkfeldausbreitung und Interferenz von Funkterminals in wählbaren Szenarien zur Beurteilung von Funkschnittstellen wie GSM/GPRS/EDGE, UMTS usw.	11/2000-12/2000
Future	Functional UMTS Real Emulator	01/2001-06/2003
TETRA	TETRA Pilotversuch Tetra-Tetrapol	03/2001-02/2003
Utran	UMTS Radio Network Dimensioning	05/2001-06/2002
MIND	Mobile IP based Network Developments	06/2001-11/2002
IPonAIR	Media Point in City Areas: Provision of Personalized High-Volume Data via Media Points in the Public Cellular Infrastructure	07/2001-12/2003
IPonAIR-T-Nova	Mobilität und Billing in heterogenen IP-basierten Netzarchitekturen	10/2001-09/2004
GPRS-QoS	Dimensioning Rules for GPRS Networks Considering Quality of Service Management Technique	01/2002-12/2002
miniWatt	Alternative Funksysteme mit minimaler Strahlungsdichte im digitalen Rundfunk, Mobilfunk, drahtlosen LANs	01/2002-12/2002
OverDrive	Spectrum Efficient Uni- and Multicast Services over Dynamic Multi-Radio Networks in Vehicular Environments	04/2002-03/2004
VMTL	Evaluation of High-Level Protocols in a Distributed Application Environment	04/2002-05/2005
GPRS-Video	Dimensionierungsregeln für GPRS bei Einführung von Video Streaming	05/2002-11/2002
WWRI	The Wireless World Research Initiative	06/2002-01/2003
ANWIRE	Academic Network on Wireless Internet Research in Europe	09/2002-08/2004

NEXWAY	Network of Excellence in Wireless Applications and Technology	09/2002-12/2004
SAILOR	Satellite Integrated UMTS Emulator	09/2002-05/2005
STRIKE	Spectrally Efficient Fixed Wireless Network based on Dual Standards	09/2002-08/2004
IRAT	Untersuchung und Optimierung des Systemwechsels zwischen GSM und UMTS Mobilfunknetzen	11/2002-10/2003
Mobil-Funk	Untersuchung von Mobilitätskonzepten in heterogenen Funknetzen	01/2003-09/2003
SorgN	Selbstorganisierendes Drahtlos-Netz mit Multihop-Fähigkeit	05/2003-09/2003
MobCon I-V	Future Mobility Concepts in Heterogeneous Networks I-V	10/2003-12/2007
WIGWAM	Leitinnovation Mobile Internet - Wireless Gigabit with Advanced Multimedia Support	10/2003-03/2007
AMBIENT NETWORKS	WWI Ambient Networks: Create a pervasive, reliable communication environment hiding the heterogeneous infrastructures, supporting the ever-changing needs of users and services	01/2004-12/2005
IPonAIR II	Enhanced Integration of UMTS and Media Points Systems Towards Fast Session Handover	01/2004-06/2004
SatNEx	Satellite Communications Network of Excellence	01/2004-12/2004
TakeOFDM	Integration von COFDM in Mehrantennensysteme und Entwicklung adaptiver Mediumzugriffsprotokolle	01/2004-12/2005
WINNER I & II	Wireless World Initiative New Radio	01/2004-12/2007
URMEL	UMTS Radio Resource Management Evaluation	03/2004-12/2004

SVC I-III	Sicherheitsrelevante Fahrzeug-Kommunikation	05/2004-04/2007
MYCAREVENT	MobillitY and CollAboRative work in European Vehicle Emergency NeTworks	10/2004-09/2007
4G-Spektrum	Spektrumbedarf für zukünftige paketvermittelnde Funksysteme	04/2005-03/2008
Dmotion	Methodologie für eine modellgetriebene Spezifikation von Komponenten für Systeme zum Straßenverkehrsmanagement und zur Straßenverkehrssteuerung und ihre Unterstützung durch marktverfügbare Spezifikationswerkzeuge	04/2005-04/2008
ScaleNet	ScaleNet-Flexible Spektrumsnutzung, Koexistenz und Spektrumsbedarfsschätzung	07/2005-09/2008
ScaleNet	Entwicklung von skalierbaren MAC und RLC-Protokollen für ein hochratiges, teilweise mehrstufiges, mobiles Zugangsnetz	10/2005-09/2008
AMBIENT NETWORKS II	Development of innovative mobile network solutions for increased competition and cooperation in an environment with a multitude of access technologies, network operators and business actors	01/2006-12/2007
Fireworks	Flexible Relay Wireless OFDM-based networks	01/2006-03/2008
TakeOFDM II	Schwerpunktprogramm "Techniken, Algorithmen und Konzepte für zukünftige COFDM Entwicklungen"	01/2006-12/2007
Huawei	Wireless Relay Technology Study	09/2006-12/2007
DISTEL	Dienste in ÖPNV-Verbundnetzen mit standardisierten Liniennetzdaten	11/2006-12/2008
UMIC	Ultra High-Speed Mobile Information and Communication	11/2006-10/2007

CoCar	Cooperative Cars-Systematische Analyse der Eignung zellularer 3G Mobilfunktechnologien für die Datenkommunikation in kooperativen, fahrzeugbasierten Telematikanwendungen	02/2007-12/2008
COOLWAVE	Work in the area of: 1. Basic interference management for flexible radio architecture, 2. Coexistence and inter-working in future IMT-Advanced systems, 3. Spectrum sharing techniques	04/2007-02/2008
Mesh-networks	Advanced Cognitive and High Throughput MAC Protocolls for Wireless Networks	04/2007-12/2008
Huawei-NextMesh	NextMesh Technology Study	07/2007-06/2008
PoSSuM	Koexistenz und Kooperation vermaschter, im Spektrum konkurrierender Funknetze mit Dienstgüte-Unterstützung	10/2007-09/2009
simoKIM	Verbundprojekt simoKIM - Sicheres und mobiles, kommunales Infrastrukturmanagement am Beispiel Straße	10/2007-09/2010
OMEGA	Collaborative Project Omega - Home Gigabit Access	01/2008-12/2010
ROCKET	Collaborative Project Rocket - Reconfigurable OFDMA-based Cooperative NetworKs Enabled by Agile SpecTrum Use	01/2008-12/2009
OTS2	Verbundprojekt OTS 2 - Kommunikation in Verbundsystemen für Verkehrsinformation und Verkehrsmanagement: Rahmenwerk zur Erweiterung, Anwendung und Test des offenen OTS-Standards	04/2008-03/2010

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