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Abstract

This deliverable represents the final report on the METIS radio link research. It provides a comprehensive and self-contained summary of all investigated technology components (TeCs), including evaluation results and conclusions on their potential for 5G. The METIS radio link research covers three main areas, which are considered key aspects for developing a self-contained air interface design for 5G: 1) Flexible air interface, 2) Waveforms, coding & modulation and transceiver design and 3) Multiple access, Medium Access Control (MAC) and Radio Resource Management (RRM). TeCs with similar research context and objectives have been grouped into clusters, whereof five have been selected as the most promising for 5G systems: From research area 1, TeC clusters providing key enablers for an air interface for ultra-dense networks (UDN) and for moving networks; from research area 2, multi-carrier transmission schemes with filtering; and from research area 3, novel access schemes for massive machine access as well as for non-orthogonal access.

Keywords

air interface, availability, dense deployment, coding, cost, coverage, energy efficiency, fading, faster than Nyquist, filter-bank multicarrier, flexibility, full-duplex, latency, link adaptation, machine-to-machine, medium access control, mobility, modulation, multiple access, non-orthogonal multiple access, orthogonal frequency division multiple access, overhead, power-domain multiplexing, radio resource management, reliability, research topic, signalling, spectrum efficiency, waveform

Executive summary

This deliverable D2.4 represents the final report on the METIS radio link research. It provides a comprehensive and self-contained summary of all investigated technology components (TeCs), including evaluation results and conclusions on their potential for 5G. The METIS radio link research covers three main areas, which are considered key aspects for developing a self-contained air interface design for 5G:

1. Flexible air interface
2. Waveforms, coding & modulation and transceiver design
3. Multiple access, MAC and RRM

For each of these areas, a multitude of novel TeCs have been proposed. These have been investigated in detail, including an assessment of their individual contribution to the METIS overall goals. Moreover, each of them has been connected to the METIS Horizontal Topics as well as to the test cases defining the METIS scenarios for 5G in Deliverable D1.1 [MET13-D11]. The connections have served as a basis for the selecting potential enablers for the METIS overall system concept [MET15-D66]. This has been facilitated by selection of TeCs for the individual HT specific concepts. In the overall test case evaluations presented in [MET15-D65], those TeCs promising significant improvements towards the METIS overall goals have been considered and evaluated with respect to their performance on system level.

The document presents a high level summary of all TeCs considered in each of the three above research areas, highlighting the key aspects of each TeC and assessing their maturity for practical application and their potential for 5G. In the main chapter, the document provides a detailed description of each TeC, describing the concept, emphasizing most important results and drawing an overall conclusion. For enabling a quick overview on each TeC's key features and their connections inside METIS (to HTs, test cases and other TeCs), profile tables are presented in the Annex. These tables also summarize the TeCs individual contributions to the METIS overall goals; they further provide a judgement on whether the TeC can be considered an evolution of current systems or if it represents a revolution in the sense that it introduces a completely novel system approach. The Annex further contains detailed evaluation results for a number of TeCs, complementing those published in conference papers and Deliverable D2.3, as referenced in the main part of the document. .

METIS radio link research has been structured into so called TeC clusters (TeCCs), which group different TeCs with similar research context and objectives. Five of these TeCCs have been identified as "most promising" for future mobile radio systems, as they address key challenges of the 5G system design such as providing enablers for new services like V2V and MMC or increasing the flexibility of the air interface to address the diverse requirements of future radio services. These are briefly summarized as follows (a more comprehensive summary can be found in the conclusions):

1. **Unified air interface for ultra-dense deployments:** Concept of an air interface targeting mobile broadband services characterized by high data rate and reduced latency.
2. **Air interface for moving networks:** This cluster addresses aspects of an air interface design for moving networks and provides radio link enablers for future V2V services.
3. **Filtered and filterbank based multi-carrier:** New waveform concepts enabling flexible spectrum sharing of different radio services with highly diverse requirements.
4. **Non- And quasi orthogonal multiple access:** Novel multiple access schemes allowing for overloading the spectrum by multiplexing users in the power and the code domain.
5. **Contention based massive access:** Novel MAC schemes for contention or reservation based access of a large number of devices with low overhead.

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List of Abbreviations, Acronyms, and Definitions

2G	2 nd Generation
3G	3 rd Generation
3GPP	Third Generation Partnership Project
4G	4 th Generation
AC	Access Class
ACK	Acknowledgement
AD	Analogue to Digital
AI	Air Interface
AMC	Adaptive Modulation and Coding
AP	Access Point
APDP	Average Power Delay Profile
ARES	Adaptive distributed Network Synchronization
ARQ	Automatic Repeat reQuest
AV	Availability
AWGN	Additive White Gaussian Noise
B4G	Beyond 4G
BB	Baseband
BER	Bit Error Rate
BH	Backhaul
BLER	Block Error Rate
BLT	Balian-Low theorem
BOM	Bill-of-Materials
BPSK	Binary Phase Shift Keying
BR	Bandwidth Request
BS	Base Station
BW	Bandwidth
C2R	Complex to Real
CA	Collision Avoidance
CAPEX	Capital Expenditure
CCRS	Coordinated and Cooperative Relay Systems
CDF	Cumulative Distribution Function
CDMA	Code Division Multiple Access
CFO	Carrier Frequency Offset
CFR	Carrier Frequency Range
C-ITS	Cooperative Intelligent Transport Systems
cmW	Centimetre Wave
COV	Coverage
C-plane	Control plane
CP	Cyclic Prefix
CP-OFDM	Cyclic-Prefix Orthogonal Frequency Division Multiplexing
CPE	Common Phase Error
CQI	Channel Quality Indicator
CPM	Constant Phase Modulation
CR	Cognitive Radio
CRAD	Cell Radius
CRC	Cyclic Redundancy Check

CRS	Common Reference Signal
CS	Compressed Sensing
CSA	Coded Slotted Aloha
CSI	Channel State Information
CSMA	Carrier Sense Multiple Access
C-TP	Cell Throughput
C-UE	Cellular UE
CW	Contention Window
D2D	Device-to-Device
DA	Digital to Analogue
DCC	De-centralized Control Congestion
DCF	Distributed Coordination Function
DeNB	Donor eNB
DFT	Discrete Fourier Transform
DFT-S-OFDM	DFT-Spread OFDM
DI	Diffuse Scatterer
DIFS	Distributed Coordination Function Inter-Frame Spacing
DL	Downlink
DMRS	Demodulation Reference Signal
DRX	Discontinuous Reception
DS	Distribution System
DTX	Discontinuous Transmission
DVB	Digital Video Broadcast
DwPTS	DL Pilot Time Slot
ECON-AU	Energy consumption per area unit
ECON-B	Energy consumption per bit
EDCA	Enhanced Distributed Channel Access
EDGE	Enhanced Data Rates for GSM Evolution
EE	Energy Efficiency
EVM	Error Vector Magnitude
eIMTA	enhanced Interference Management and Traffic Adaptation
eNB	Enhanced node B
EPDCCH	Enhanced PDCCH
ESS	Extended Service Set
ETSI	European Telecommunications Standards Institute
FAI	Flexible Air Interface
FAIR	Fairness
FBMC	Filter-Bank Multi-Carrier
FD	Full Duplex
FDR	Full Duplex Radio
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access



FEC	Forward Error Correction
FeICIC	Further enhancement of inter-cell interference coordination
FER	Frame Error Rate
FFT	Fast Fourier Transform
FIR	Finite Impulse Response
FMT	Filtered Multi-Tone
FOM	Figure Of Merit
FSPA	Full Search Power Allocation
FST	Fast Session Transfer
FTN	Faster Than Nyquist
FTP	Fractional TPA
GBR	Guaranteed Bit Rate
GFDM	Generalized Frequency Division Multiple Access
GGMA	GR/GRM Mapping Approach
GI	Guard Interval
GLB	Green Link Budget
GOMP	Group Orthogonal Matching Pursuit
GP	Guard Period
GPRS	General Packet Radio Service
GPS	Global Positioning Service
GR	General Requirement
GRM	General Requirement Metric
GRT	General Requirement Tag
GSM	Global System for Mobile Communications
HARQ	Hybrid ARQ
HETNET	Heterogeneous Networks
HPA	High Power Amplifier
HSD	Hybrid Sparse/Diffuse
HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
HT	Horizontal Topic
HW	Hardware
ICI	Inter Carrier Interference
ID	Identifier
IEEE	Institute of Electrical and Electronics Engineers
IF	Intermediate Frequency
IFDMA	Interleaved FDMA
IFFT	Inverse Fast Fourier Transform
IM	Inter-modulation
INTG	Integrity
IOTA	Isotropic Orthogonal Transform Algorithm
IP	Internet Protocol
IR	Incremental Redundancy
IRC	Interference Rejection Combining
IS	Interference Suppression
ISI	Inter Symbol Interference

ITS	Intelligent Transport Systems
ITU	International Telecommunication Union
KPI	Key Performance Indicator
L1	Layer 1
LA	Local Area
LAN	Local Area Network
LAT	Latency
LDPC	Low Density Parity Check
LDS	Low Density Signature
LFDMA	Localized FDMA
LLC	Logical Link Control
LNA	Low Noise Amplifier
LOLA	Low-Latency
LOS	Line of Sight
LS	Least Squares
LSD	List Sphere Decoding
LSE	Link Spectral Efficiency
LTE	Long Term Evolution
LTE-A	LTE Advanced
M2M	Machine to Machine
MA	Multiple Access
MAC	Medium Access Control
MAI	Multiple Access Interference
MAP	Maximum A-Posteriori
MASP	Modified Automatic Selftime-correcting Procedure
MAX	Maximum
MBB	Mobile Broad Band
xMBB	Extreme MBB
MBSFN	Multicast Broadcast Single Frequency Network
MC	Multi Carrier
MCC	Mission Critical Communications
MCM	MultiCarrier Modulations
MCM	Multi-path Channel Margin
MCS	Modulation Coding Scheme
MD	Moving Discrete Scatterer
MDFT	Modified Discrete Fourier Transform
MIM	Multiple Interface Management
MIMO	Multiple Input Multiple Output
MLD	Maximum Likelihood Detection
MMC	Massive Machine Communication
MMSE	Minimum Mean Square Error
mmW	Millimetre Wave
MN	Moving Networks
MPA	Message Passing Algorithm
MPC	Multipath Component
MRC	Maximum Ratio Combining
MRE	Multi-rate equalizer



MSE	Mean-Square Error
MTC	Machine Type Communication
mMTC	Massive MTC
uMTC	Ultra-reliable MTC
MTD	Machine Type Device
MTU	Maximum Transmission Unit
MU	Multi User
MUD	Multi User Detection
MU-MIMO	Multi User MIMO
MUPA	Multi User Power Allocation
NACK	Negative ACK
NLOS	Non Line of Sight
NMSE	Normalized Mean Square Error
NOMA	Non-Orthogonal Multiple Access
NW	Network
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OFDP	Optimal Finite Duration Pulse
OMA	Orthogonal Multiple Access
OPEX	Operational expenditure
OQAM	Offset QAM
PAL	Protocol Adaptation Layers
PAM	Pulse Amplitude Modulation
PAPR	Peak-to-Average Power Ratio
PAR	Project Authorization Request
PBCH	Physical Broadcast Channel
PDCCH	Physical Downlink Control Channel
PDCP	Packet Data Convergence Protocol
PDU	Protocol Data Unit
PER	Packet Error Rate
PHY	Physical
PLM	Path-Loss Margin
PMI	Precoder Matrix Indicator
PMP	Point to Multipoint
PNP	Phase Noise Pilot
PRACH	Physical random access channel
PRB	Physical Resource Block
ProSe	Proximity Service
PSD	Power Spectral Density
PSS	Primary Synchronization Signal
PUCCH	Physical Uplink Control Channel
PUSCH	Physical Uplink Shared Channel
QAM	Quadrature Amplitude Modulation

QoE	Quality of Experience
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RA	Random Access
RAN	Radio Access Network
RAST	Random Access Schemes Throughput
RAT	Radio Access Technology
RBDS	Random Broadcast based Distributed consensus clock Synchronization
REL	Reliability
RF	Radio Frequency
RLC	Radio Link Control
RMS	Root Mean Square
RN	Relay Node
RRC	Root Raised Cosine
RRH	Remote Radio Head
RRM	Radio Resource Management
RS	Reference Signal
RT	Research Topic
RTG	RX/TX Transition Gap
RTT	Round Trip Time
Rx	Receiver
SA2	System Aspect, Work Group 2: 3GPP Architecture
SAW	Surface Acoustic Wave
SBW	Supported Bandwidth
SC	Subcarrier
SCMA	Sparse Code Multiple Access
SD	Static Discrete Scatterer
SDM	Spatial Division Multiplex
SE	Spectral Efficiency
SFO	Sampling Frequency Offset
SG	Scheduling Grant
SI	Study Item
SIC	Successive Interference Cancellation
SIFS	Short Interframe Spacing
SINR	Signal to Interference plus Noise Ratio
SISO	Single-Input Single Output
SLNR	Signal-to-Leakage and Noise Ratio
SMT	Staggered Multi-Tone
SNR	Signal to Noise Ratio
SotA	State of the Art
SR	Scheduling Request
SRS	Sounding Reference Signal
SSID	Service Set Identifier
SSS	Secondary Synchronization Signal
STA	Station
STBC	Space Time Block Code
STDMA	Self-organizing Time



	Division Multiple Access
SU	Single User
TA	Timing Alignment
TC	Test Case
TCCA	TETRA + Critical Communications Association
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
TETRA	Terrestrial Trunked Radio
TFL	Time Frequency Localization
TGad	Task Group IEEE802.11ad
TM	Transmission Mode
TP	Throughput
TPA	Transmit Power Allocation
TPC	Transmission Power Control
TSF	Time Synchronization Function
TTG	Tx/Rx Transition Gap
TTI	Transmission Time Interval
Tx	Transmitter
UDN	Ultra Dense Network
UE	User Equipment
UFMC	Universal Filtered MultiCarrier
UF-OFDM	Universal Filtered OFDM
UHF	Ultra High Frequency
UL	Uplink

UMa	Urban Macro
UMi	Urban Micro
UMTS	Universal Mobile Telecommunications System
U-plane	User plane
URC	Ultra Reliable Communications
V2D	Vehicle-to-Device
V2I	Vehicle-to-Infrastructure
V2V	Vehicle-to-Vehicle
V2X	Vehicle-to-X
VANET	Vehicular Ad Hoc Network
VHT	Very High Throughput
VoIP	Voice over IP
WA	Wide Area
WAVE	Wireless Access in Vehicular Environments
W-CDMA	Wideband CDMA
WI	Work Item
WiGig	Wireless Gigabit Alliance
WiMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Network
WP	Work Package
WPAN	Wireless Personal Area Network
WSS-US	Wide-Sense Stationary Uncorrelated Scattering
XOR	Exclusive OR
ZF	Zero Forcing



1 Introduction

METIS radio link research investigates radio link solutions for the 5G mobile radio system, addressing the diverse requirements of the services expected to be provided by future radio. The research has been organized into 14 main technology component clusters (abbreviated as TeCCs), each one defining a research topic focusing on particular aspects of the radio link design and addressing selected challenges and use cases of the future radio research. These main TeCCs are identical with the research topics (RTs) introduced in deliverable D2.1 [MET13-D21] and used in Deliverable D2.2 [MET13-D22]. In Deliverable D2.3 [MET14-D23], further sub-clusters have been introduced, grouping solutions with similar research context and objectives and thus allow for a more detailed overview on the particular tracks of the METIS radio link research. For a complete overview on the TeCCs, see the introduction of D2.3. Each technology component cluster covers several technology components (abbreviated TeCs), which represent the concrete technical solution to a research problem on the radio link layer, as described in deliverable D2.1. Deliverable D2.2 described the state of the art for each of the research topics and elaborated on the particular solutions studied in METIS. Further, it has been identified there how the TeCs can serve as enablers for the METIS Horizontal Topics (HTs). An illustrative overview on the HTs and research challenges addressed by the TeCCs has been presented in the introduction of Deliverable D2.3. That document further provided concrete descriptions of the TeCs together with initial evaluation results.

This deliverable D2.4 represents the final report on the METIS radio link research. It provides a comprehensive and self-contained summary of all investigated TeCs, including evaluation results and conclusions on their maturity for practical application and their potential for 5G. The METIS radio link research covers three main areas, which are considered key aspects for developing a self-contained air interface design for 5G:

1. Flexible air interface (TeCC#1 – TeCC#6)
2. Waveforms, coding & modulation and transceiver design (TeCC#7 – TeCC#10)
3. Multiple access, MAC and RRM (TeCC#11 – TeCC#14)

For each of these areas, a multitude of novel TeCs have been proposed. These have been investigated in detail, including an assessment of their individual contribution to the METIS overall goals. Moreover, each of them has been connected to the METIS Horizontal Topics as well as to the test cases defining the METIS scenarios for 5G in Deliverable D1.1 [MET13-D11]. The connections have served as a basis for selecting potential enablers for the METIS overall system concept [MET15-D66]. This has been facilitated by selection of TeCs for the individual HT specific concepts, which continuously triggered additional evaluations throughout the project. In the overall test case evaluations presented in [MET15-D65], those TeCs promising significant improvements towards the METIS overall goals have been considered and evaluated with respect to their performance on system level.

The following table provides an overview on the individual TeCs investigated in METIS radio link research and their targeted scenarios and test cases from [MET13-D11]. TeCs are grouped and numbered according to the TeC clusters, which are specified in the grey rows.



Table 1-1: List of Technology Components investigated METIS radio link research

No.	TeC	Scenario(s) targeted
1	Unified air interface design for dense deployments	
1.1	Harmonized OFDM	Virtual reality office Dense urban inform. society Massive machine access
1.2	METIS UDN optimized frame structure	
1.3	Dynamic TDD	
2	Optimized signalling for low-cost MMC devices	
2.1	MMC type D2D links	Massive machine access
2.2	Quasi-orthogonal random access	
2.3	Hybrid Access Scheme for reduced signalling overhead	
3	RF architecture for dynamic spectrum access	Blind spots Emergency communications
4	Multiple air interface management	
4.1	Multiple Interface Management in MT-HetNets	Service in a crowd Traffic efficiency and safety
4.2	Software configurable air interface	Service in a crowd Massive machine access
5	Advanced signalling concepts	(obsolete)
6	Air interface for moving networks	
6.1	Framework for URC	Traffic efficiency and safety
6.2	Modelling & predicting the reliability of a link	
6.3	Channel estimation for V2V links	
6.4	Channel prediction	Real-time remote computing
6.5	Ad-hoc MAC for V2V	
7	Faster than Nyquist	Virtual reality office Dense urban inform. society
8	Filtered and filterbank based multi-carrier	
8.1	FBMC based waveform & transceiver design	Service in a crowd Massive machine access
8.2	Universal filtered multi-carrier (UFMC)	Traffic efficiency and safety
9	Modulation & coding and new channel coding concepts	
9.1	Constrained envelope coded modulation	Massive machine access
9.2.1	Adaptive complexity flexible baseband	Blind spots
9.2.2	Practical lattice codes	Real time remote computing
10	Advanced transceiver design	
10.1	Full Duplex communications	Service in a crowd
10.2	Multi-rate equalizers for single-carrier communications	Blind spots Massive machine access
11	Multiple access	
11.1	Non- and quasi-orthogonal access allowing spectrum overload	
11.1.1	Non-orthogonal multiple access (NOMA)	Service in a crowd Traffic efficiency and safety Real-time remote computing Massive machine access
11.1.2	Sparse code multiple access (SCMA)	
11.2	FBMC based multiple access and Cognitive Radio	
11.2.1	Multiple access for MIMO FBMC systems	Service in a crowd

11.2.2	MA using Cognitive Radio	Blind spots Emergency communications
12	Medium access control	
12.1	Contention based massive access	
12.1.1	Coded random access	Massive machine access Teleprotection in smart grid network
12.1.2	Coded access reservation	
12.1.3	Advanced PHY processing for enh. MAC	
12.2	Distributed network synchronization	Traffic efficiency and safety (networks out of coverage of BS)
12.3	MAC for UDN and mmW	Service in a crowd Virtual reality office
13	Hybrid Automatic Repeat Request (HARQ)	
13.1	Backtrack retransmission with ternary feedback	Virtual reality office Dense urban inform. society Service in a crowd
13.2	Reliability-based HARQ	
14	Radio link enablers for RRM	Service in a crowd

1.1 Structure of the document

A high level summary of all TeCs considered in each of the three above research areas is presented in chapter 2, highlighting the key aspects of each TeC and assessing their maturity for practical application and their potential for 5G. Chapter 3 then presents a detailed description of each TeC, describing the concept, emphasizing most important results and drawing an overall conclusion. In this chapter, each TeC is devoted its own sub-section, where the sub-section number is aligned with the corresponding TeC number¹. Each of these sub-sections starts with a short description of the TeC and its key objective, written in italic letters, and ends with a brief conclusion on key achievements and the TeC's relevance for application in future systems. Overall conclusions on the METIS radio link research are drawn in chapter 4. In Annex A (chapter 6), profile tables are given for each TeC, capturing the evaluated KPIs in terms of the General Requirement Metrics (GRMs) and General Requirement Tags (GRTs) as identified in Deliverable D2.1 and the achieved gains. Moreover, information on the connection to METIS HTs and test cases as well as the TeC's individual contribution to the METIS overall goals is given. The table further contains a judgement on whether the TeC can be considered an evolution of current systems or if it represents a revolution in the sense that it introduces a completely novel system approach. The tables allow the reader to get a quick overview on each TeC's key features, their promised gains and their connections inside METIS (to HTs, test cases and other TeCs). The Annex further contains detailed evaluation results for a number of TeCs, complementing those published in conference papers and Deliverable D2.3, as referenced in the main part in chapter 3.

¹ only exception to that rule is for TeC#1.2 and TeC#1.3, which are described jointly in a single section 3.1.2.

2 Candidate technologies for the 5G system

2.1 Flexible Air-Interface Design

Flexibility and scalability are important design targets in addressing the needs of the diverse services envisaged for 5G, such as e.g. high-definition video streams, low-cost massive machine communications (MMC) and the numerous V2x services raised in the context of moving networks. All these services impose different requirements in terms of data rates, latency, coverage, reliability and mobility support together with the demands of reduced cost and improved energy efficiency [MET13-D21], [MET13-D22]. Within the research work conducted in this task, several air interface concepts addressing these different connection types with their diverse requirements and related signalling concepts have been elaborated. Flexibility in terms of new connection types and spectrum usage has been investigated, together with concepts considering multi-RAT aspects and allowing different adaptive air interface configurations. Combining the different proposed schemes to form a fully harmonized and mature air interface solution fulfilling the various requirements is still left for future research.

Flexibility in terms of new connection types

In order to provide an efficient air interface solution for mobile broadband scenarios addressing the target of ultra high data rates and reduced latency, the concept of dense deployment optimized orthogonal frequency division multiplexing (OFDM)-based air interface has been introduced (TeC#1.1-TeC#1.3). This approach utilizes dynamic time division duplexing (TDD) in order to enable efficient spectrum usage. The enablers provided by short link distances, such as possibility for symmetric design with respect to uplink (UL) / downlink (DL) and the utilization of higher cmW and mmW frequency bands, are also exploited. In order to minimize cost and complexity, the design is targeting for a unified solution suitable for diverse link types, such as access and self-backhauling links. The properties of small cell environment together with the expected improvements in the future component technology are utilized to optimize ultra-dense networks (UDN) physical layer numerology (i.e. subcarrier spacing, cyclic prefix and guard period times). The UDN numerology is then exploited to design a TDD optimized physical frame structure. In this design, the target is to reduce latency and to keep processing simple. The proposed frame structure enables a simple hybrid automatic repeat request scheme with noticeably smaller round trip time compared to LTE-A. User plane latency analysis shows ~5 times reduction w.r.t. LTE-A (on cmW). Latency reduction leads also to lower buffering cost and to noticeably reduced battery consumption. In addition, noticeable throughput gains are provided e.g. via the full UL/DL interval switching flexibility and multi-antenna and beamforming solutions. To summarize, the dense deployment optimized air interface can be seen as an important physical layer enabler for 5G mobile broadband, while some of the aspects, such as detailed control channel and signalling design, are still left for further study.

Investigation of air interface components for moving networks, including traffic safety applications and the wireless broadband access demands of the vehicular users, has been widely covered in the research of this task. In order to enable the provision of services requiring ultra-high reliability in the future 5G air interface, a framework for ultra-reliable communications (URC) within the 5G has been investigated (TeC#6.1). The framework incorporates the indication of the availability of a link with a desired reliability as a measure to allow opportunistic access of ultra-reliable services when the link conditions are fair enough. A theoretical model for modelling and predicting the reliability of a link is created (TeC#6.2), which can well complement the framework for URC. URC framework related system level simulation results show that the METIS target reliability can be achieved in the case of perfect channel estimation; otherwise the target reliability is achieved at the expense of availability. It could also be concluded that simple channel prediction methods cannot fulfil the ultra-reliable

requirement especially at high Doppler spreads. To address this issue, a novel scheme exploiting predictor antennas on vehicles has been developed and evaluated (TeC#6.4). With the proposed scheme, dedicated antennas in the front row of an antenna array are used as a 'spy' to predict the radio channel that will be experienced in the near future by the other antennas of the array used for data transmission. This scheme enables more efficient utilization of channel state information also at vehicular velocities. With the investigated approach, the useful prediction horizon can be increased by an order of magnitude compared to conventional Kalman / Wiener prediction. The scheme could further enable massive multiple input multiple output (MIMO) and beamforming solutions to be used also for high speed links.

Since conventional channel estimation scheme have shown to perform poorly in V2V channels, more accurate preamble type of channel estimation algorithm exploiting the mixed sparse-diffuse structure in the delay-Doppler domain of such V2V channel has been proposed (TeC#6.3). The algorithm is shown to deliver 10-15 dB SNR gain of the normalized mean-squared error compared to a traditional least squares estimator. Additionally, since D2D transceivers in moving networks may require to communicate in ad-hoc mode, a novel time-division based V2V multiple access scheme has been introduced and investigated (TeC#6.5). The scheme based on coded slotted ALOHA approach is shown to support more than 125% increase in the number of users over CSMA with reasonable packet error rate. To conclude, the investigated air interface aspects related to moving networks show noticeable progress in enabling ultra-reliable and V2X applications in 5G or for the future releases of LTE evolution. However, various practical aspects, such as including utilization of realistic frame formats and receivers as well as considerations on computational complexity are still left for further study.

In addition to addressing the challenges of mobile broadband, the design of a 5G air interface should consider the proper integration of low-cost MMC and D2D devices. In order to improve the coverage of MMC devices and to provide enablers for access load relief in case of massive machine type communication (MTC) access, the approach of relaying MTC traffic via a cellular UE has been investigated (TeC#2.1), making use of different successive interference cancellation schemes. Theoretical zero-outage rate bounds for the cellular link have been defined both for joint and single-user decoding settings, allowing tradeoffs between the cellular link zero-outage data rates and the amount and data rates of the interfering MTC links. In order to address the challenge of multiple access of MTC, a new multiple access scheme based on a machine-to-machine access manager exploiting the periodicity of the traffic caused by machine type devices has been introduced (TeC#2.3). The proposed scheme is shown to outperform the slotted ALOHA model with respect to throughput and collisions, but comes with a trade-off of increased delay. Also, quasi-orthogonal random access with minimum mean square error (MMSE)-based multiuser detection and narrow-band design has been investigated (TeC#2.2). The proposed approaches are promising candidates for 5G MMC and D2D realizations, while leaving the practical realization aspects still for further study.

Flexibility in terms of spectrum usage

One important design goal for a flexible air interface is the capability to operate in new spectrum, together with the support of dynamic spectrum usage. The cost and the complexity of the baseband design would be minimized if the same baseband design principles could be applied to a large range of carrier frequencies. To address this, the concept of "harmonized OFDM" has been introduced with the idea of scaling the physical layer numerology as a function of carrier frequency, while keeping fast Fourier transform size within a small set of quantized values (TeC#1.1). Results of the related preliminary OFDM numerology analysis indicate that phase noise seems not to be a show stopper even at mmW frequency range with relatively small subcarrier spacing, further indicating that OFDM may still be a feasible waveform candidate even at high carrier frequencies. Further, multiband processing and switching between different RF bands has been examined (TeC#4.1). In order to support cost-

efficient dynamic usage of spectrum from hardware point of view, frequency agile front ends have been investigated (TeC#3). A novel high-IF front end architecture has been proposed for wideband operation. The performance evaluation comparing the new high-IF architecture to traditional homodyne receiver shows significant performance improvement with respect to out-of-band linearity together with gains in terms of noise figure and sensitivity. The proof of concept work is ongoing via validation of a TDD prototype. The proposed high-IF scheme can be seen as a promising enabler e.g. for multiple access for Cognitive Radio applications.

Configuration and multi-RAT aspects

Since future 5G system may eventually consist of several air interface solutions, also the combination and switching between multiple independent air interface designs is seen as an important research topic. For this purpose, a channel quality indicator based green link budget metric has been introduced (TeC#4.1). This metric is used to select the air interface and associated transmission mode guaranteeing targeted radio coverage, data rate and QoS. Evaluations are performed in the context of hot spots and large indoor areas utilizing different transmission modes of IEEE802.11 ac/11n (5 GHz) and UWB-MB-OFDM (60 GHz) technology. Noticeable gains e.g. in terms of transmit power reductions can be identified via appropriate air interface selection based on this metric. Further, comparisons of increasing the amount of MIMO spatial streams versus bandwidth increase have been conducted by utilizing IEEE802.11ac technology. While the proposed metric and the related selection algorithm are already on quite mature state, combining them with the novel 5G air interface solutions is still under research and thus left for further study.

On even higher level, in order to provide an overall solution for adapting the 5G system to the diverse applications and scenarios, the concept of software configurable air interface has been introduced (TeC#4.2). This concept allows for an arbitrary composition of different air interface building blocks, e.g. waveform, frame structure, multiple access scheme, coding and modulation and protocols, according to the needs of a use case or application. Since the eventual selection of the 5G air interface components has not yet been conducted, the identification of the exact set of the 5G air interface configurations and the development of concrete algorithms used for selecting those configurations is still left for further study.

2.2 New waveform candidates, modulation & coding and transceiver design

This task includes four research topics, namely waveform candidates, FTN signalling, advanced transceiver and new modulation and channel coding. Among them, the research on waveform candidates provides the most appealing and mature results. Here we provide a general summary of the outcome from each topic.

Waveform candidates

The LTE waveform, i.e. CP-OFDM, was designed for mobile broadband service. Towards 5G networking context, where the support of a multitude of services will be a must, multiple services cannot easily coexist, in the same band, in CP-OFDM based systems without causing inter-service interference due to the poor frequency sub-band isolation inherent to CP-OFDM. To solve this problem, in this research task, there are two waveform candidates considered as promising alternatives to LTE waveform, which rely on additional filtering techniques to enable proper frequency isolation. The first candidate is named UPMC, or UF-OFDM (TeC#8.2), and the other one is FBMC (TeC#8.1). The main difference between these two is that UF-OFDM/UPMC adopts a sub-band filtering concept, while for FBMC a subcarrier-wise filterbank concept is used. The other difference is the time domain symbol isolation. UF-OFDM/UPMC separates the symbols like CP-OFDM, but this comes at the price of overhead for the filter tails, which is equivalent to the overhead of CP in CP-OFDM. For FBMC, the symbols are overlapping in time domain, so that the filter tails do not directly translate to an

overhead as in UF-OFDM/UFMC, but the symbols are no longer clearly separated in time domain.

Both candidates have very similar advantages compared to CP-OFDM. We briefly list them below

- Good spectrum containment: due to the introduction of subcarrier/subband filtering, the waveform candidates possess better spectrum containment than CP-OFDM. This enables partitioning the transmission bandwidth into isolated sub-bands that can be individually configured according to the requirements of any service. It facilitates simple and interference-free coexistence of multiple services in a contiguous frequency band.
- Increased spectral efficiency: this spectral efficiency gain is indeed given from the first advantage: Due to good spectrum containment, smaller guard bands are needed to properly isolate a signal from another one adjacent in frequency, thus increasing the system effective bandwidth while still respecting the maximum out-of-band power radiation, resulting in a higher spectral efficiency. The FBMC signal can in addition be transmitted without using CP. This further increases the spectral efficiency.
- Relaxed synchronization requirements: This advantage is very helpful for applications that may suffer from the lavish LTE-like synchronization procedure, e.g., MTC access. The waveform candidates provide an enhanced robustness against the synchronization error.
- Compatible with LTE pilot pattern and MIMO modes: Since UF-OFDM/UFMC maintains the signal structure of CP-OFDM, UF-OFDM/UFMC is claimed to be compatible with all LTE transmission modes as well as pilot pattern and channel estimation. FBMC has shown by simulation results that it can use the same pilot pattern and channel estimation method as LTE system without any performance loss. Moreover, the simulation results show that FBMC can use all LTE MIMO transmission modes; notably it is indicated clearly that for MIMO transmission all the LTE linear precoders can be reused for FBMC without any performance loss.

Towards 5G scenarios, there are a lot of advantages that FBMC and UF-OFDM/UFMC can bring to the operators. For instance, 5G network is supposed to support a multitude of services ranging from best effort MBB to sporadic MTC traffic; from static sensors to high speed cars/trains; and from normal access to massive access. The waveform candidates can easily define different service zones within the 5G spectrum for different services without introducing inter-service interference. Moreover, relaxed-synchronism is another important advantage of the waveform candidates. For the MMC applications, strict closed-loop synchronization may no longer be required. This will save the traditional signalling overhead for timing advance up to 15% compared to closed-loop synchronization (as shown in the TC evaluations performed in [MET15-D65]) and reduce latency caused by a connection setup procedure to a few milliseconds.

Faster than Nyquist technology

Faster than Nyquist (FTN) signalling can, in theory, provide up to twice the spectral efficiency compared to regular signalling (TeC#7). Naturally going beyond the Nyquist limit leads to an interference problem. The objectives in the METIS project were to provide an overview of the FTN concept, to examine its single and multicarrier variants and to determine the 5G scenarios that should be preferably targeted. Several results are obtained from this project in different research aspects. From theoretical analysis, a frame theory based analysis was conducted for exploiting a duality property between oversampled OFDM and FTN-Multi-Carrier Modulation (MCM) to find the lower bound of interference for any packing factor. From power loading aspect, a waterfilling approach was proposed in order to find the theoretical limits of the information rate for spectrally shaping OFDM. Also, from practical implementation

aspects, the receiver designs for FTN signalling on top of either OFDM or FBMC were investigated.

The advantage of FTN signalling is to boost the throughput on the link that somehow cannot (or not easily) take advantage of MIMO transmission. For example in the D2D case where the MIMO channel correlation could be very high. The research on FTN is still in its infant stage which calls for further investigations.

Advanced modulation and coding

We investigated the suitability of constrained envelope coded modulation for the various METIS scenarios and test cases, in which we identified a low PAPR multi-carrier scheme, denoted Continuous Phase Modulation Single-Carrier Frequency Division Multiple Access (CPM-SC-FDMA), for MMC in low-mobility scenarios (TeC#9.1). The advanced codec topic involves defining an adaptive complexity flexible baseband, which adapts its iterative processing under different channel conditions, resulting in an efficient complexity consummation for future 5G multitude of service support (TeC#9.2.1). The topic on shaping for lattice codes investigates their capacity achieving ability on the Gaussian and Rayleigh fading channels and their properties which make them suitable to be used in the multiple access relay channel (TeC#9.2.2). The research outcome shows that for low spectral efficiencies, a modification of the receiver is required to take into account the boundaries of the shaping region. On the Rayleigh fading channel, however, shaping is not mandatory. One advantage of lattice codes is their linear structure, which renders them suitable for ML joint code-modulation decoding in the case of low frame lengths. This outcome can be beneficial to the use case where short packages are transmitted, such as in MTC context. Finally, we also investigated novel schemes for fast error rate estimation, which could enable faster link adaptation procedures. However, the contributions have not proven to greatly improve the prediction horizon and the accuracy with respect to the state of the art.

Advanced transceiver design

There are two subjects tackled in this topic. These are Full Duplex (FD) transmission in FDD mode and single carrier transmission with Multi-Rate Equalization (SC-MRE). The advantages of FD (TeC#10.1) are two-fold: On one hand, it can double the spectral efficiency and on the other hand, compared to TDD mode, it can reduce the latency. The research on FD is mature enough for small power transmission applications such as D2D. The simulation results show that the self-interference cancellation can achieve up to 110 dB isolation, which can effectively put FD into consideration for the practical use. The SC-MRE (TeC#10.2) has claimed several advantages. At first, it can naturally lead to a very low PAPR, making it well-suited for MMC type services. With a linear MMSE equalizer at the receiver, it proves that a quasi-Shannon-capacity can also be achieved. The complexity increase compared with multi-carrier system (e.g. OFDM) has proved to be moderate.

2.3 Novel multiple access schemes, MAC and RRM

In addition to the physical layer, also the multiple access and medium access control are an integral part of the radio link design. Current mobile communication systems like LTE were designed with a clear focus on end users that expect high throughput data traffic in addition to voice service and targeting deployment at traditional cellular carrier frequencies. In the meantime these requirements have changed significantly towards a much broader service scope, as described in [MET13-D11]. The targeted new application scenarios also require a fundamental change of the multiple access and medium access. These aspects were the focus of this task, and in this section some results of the research conducted in this area within METIS will be highlighted.

Future cellular networks will probably consist of macro cells delivering the basic coverage and additional layers of small cells delivering high throughput, also called UDN, for which a resource allocation mechanism including Interference Aware Wireless Backhaul Routing and Resource Allocation has been developed and evaluated (cf. [MET14-D33] section 4.3.2 for details). New frequency bands, which are difficult to use in macro cells, like mmW, will likely be used in small cells, whereas macro cells may reuse current technology to some extent. Nevertheless, the macro cell layer has to be improved significantly as well to fulfil the increasing demand of traffic and provision of new services. One promising research direction to increase the spectral efficiency significantly compared to current LTE-A is the relaxation of the orthogonality constraint of OFDMA - not only from the waveform, but also from the multiple access perspective (TeCC#11.1). While MU-MIMO has already extended the multiplexing space into the spatial domain, two new schemes investigated in METIS use even more dimensions, namely the power domain and the code domain.

The first scheme called NOMA can significantly enhance existing macro cell technology by multiplexing users in the power domain with only moderate increase of scheduler and receiver complexity (TeC#11.1.1). This idea exploits the channel gain difference among different users in a cellular system, a feature which has not been directly exploited in the past. NOMA can be easily applied on top of OFDMA and be combined with SU-MIMO and MU-MIMO, and even the LTE procedures and signalling mechanisms can be reused in a quite straightforward manner. With NOMA, the average throughput as well as the cell-edge throughput in a cell can be increased significantly at a moderate increase of the complexity.

An alternative approach called MU-SCMA also relies on the basic idea of power domain multiplexing, but combines this with code multiplexing based on optimized multi-dimensional sparse codewords (TeC#11.1.2). Those codewords additionally achieve a multi-dimensional modulation gain leading to an even higher average and cell-edge throughput gain. Furthermore, SCMA is very flexible as it can be used for macro cell high throughput traffic as well as for efficient contention based random access. The further increased detection complexity due to the multi-dimensional sparse codewords can be kept on a moderate level thanks to the sparseness of SCMA codewords and low projection codebooks. Both approaches are very promising ideas to significantly improve the current LTE-A technology.

To complement macro coverage with a small cell layer which may be ultra-dense, new solutions for medium access control are needed to exploit the benefits of the smaller cell size, to enable the use of new spectrum up to mmW as well as to efficiently support self-backhauling. A new MAC for mmW utilizing layered resource management based on a template frame is proposed, where each layer operates at a different time scale and different granularities (TeC#12.3). The resulting opportunistic scheduling, obtained by utilizing shared resources provided by the template frame, can achieve a significant gain of up to four times the total system throughput that can be achieved by fixed scheduling, which is observed by simulations. In addition, a performance evaluation by system level simulations showed that the MAC approach with contention control signalling works well as part of the layered resource management, and the METIS TC3 throughput and file delay transfer KPIs are met.

Beyond the challenge to improve the throughput per area in the range of 10X compared to current systems, completely new services require innovative solutions to be included in the system design. One of the biggest and most relevant challenges for 5G is MTC, leading to new requirements. The goal is to achieve low cost, especially at the UE side, and low overhead to improve the efficiency for sporadic and short message traffic significantly compared to LTE-A, which is not well suited for this kind of traffic. One direction taken in METIS to achieve this is the idea of unscheduled instead of scheduled access, which is usually used for high throughput traffic (TeCC#12.1). Unscheduled traffic can be served either through direct random access, where the devices transmit data to the network without scheduling their transmission, and access reservation, where the devices perform a reservation request to the network prior to data transmission, using an underlying random

access mechanism. Several new ideas have been investigated in detail with practical assumptions and are already in a very mature state.

Random access with repeated transmissions can be interpreted as a random LDPC code in an erasure channel which is known to get very close to the theoretical limit. A new solution for coded random access proposed in METIS was inspired by this analogy and enables a very efficient medium access control for unscheduled access, as the parameters of the access schemes can be adapted on-the-fly to maximize the expected throughput (TeC#12.1.1). Another very promising proposal is a multi-user detection (MUD) based on compressed sensing (TeC#12.1.3). To increase the efficiency of MMC, a joint activity and data estimation is proposed which allows sensors to transmit whenever they have data to transmit, keeping sensors simple. The aggregation point uses advanced physical layer detection exploiting the sporadic nature of the transmission by so-called compressed sensing based multi-user detection. These two schemes, one on PHY, one on MAC layer, can be beneficially combined, yielding an efficient solution for massive machine access. Detailed evaluations have been carried out under realistic assumptions, and it was shown that up to 10 times more devices can be supported compared to current LTE-A capabilities. In addition to the improvements of random access schemes, also a new adaptive access reservation scheme has been investigated using either orthogonal, random, or unique coded reservation token (TeC#12.1.2). By this adaptation, high throughput can be achieved over a wide range of access loads.

One essential aspect important for many current and future services is reliability of the data transmission, which is a big challenge especially in wireless communication. HARQ is a powerful technique to achieve reliable transmission in fading scenarios with high efficiency. Nevertheless, the HARQ schemes currently implemented in LTE may not satisfy the challenging new requirements like significantly improved spectral efficiency and high reliability communication even in scenarios with limited or delayed feedback.

Two novel HARQ schemes that take the reliability of the earlier transmissions into account to optimize the efficiency of the retransmissions were proposed. One scheme uses a fixed channel code and adapts the fraction of new data and further redundancy of earlier packets in each transmission based on delayed feedback of the channel state (TeC#13.1). By this rather simple protocol the ergodic capacity of the fading channel can be achieved asymptotically. Although this idea is in rather conceptual state, the results are very promising. Another more mature idea that considers reliability to optimize the efficiency was proposed and evaluated with respect to efficiency, delay and feedback improvements (TeC#13.2). The receiver signals back a quantized value of the amount of required additional information, and based on this input the amount of additionally transmitted bits is determined. Together with an aggressive rate adaptation, a gain of approximately 30% in throughput was shown for a wide range of user velocities up to 250km/h with only a minor increase in delay compared to LTE HARQ.

In addition to the topics already mentioned, very promising research has been done in this task for other important enablers for 5G solutions: In the area of multiple access for FBMC, which takes the specific properties of this new waveform into account (TeC#11.2.1), promising ideas were presented as well as for cognitive radio as an important enabler for the co-existence of different systems in the same frequency band (TeC#11.2.2). New and improved distributed synchronization algorithms were proposed and analyzed for networks without a central controller or with only limited contact to a central controller (TeC#12.2). These proposed synchronization algorithms are an enabler for many other techniques and solutions investigated for 5G like D2D, V2V, MMC or ad-hoc communication. Enablers for faster D2D connection setup and transmission times were also investigated. A smart centralized D2D RRM scheme is proposed with a small cost in additional signalling overhead by exploiting partial CSI and user positions, so that the D2D links can contribute to traffic offloading and have very limited influence on cellular links (TeC#14).

3 Radio link technology components

This chapter gives detailed descriptions for the individual TeCs in the TeC clusters. Profile tables for each TeC as well as further detailed evaluation results are found in Annex A, which is structured similarly as this chapter.

3.1 Unified air interface design for dense deployments

The technology component cluster provides flexible air interface for ultra-dense network targeting gigabit mobile broadband communication with optimized numerology and frame structure facilitating wide spectrum ranges up to cmW and mmW bands.

To realise the vision of ubiquitous mobile broadband with extreme data rate demands in the order of many Gbps where needed with excellent user experience, we need a very dense deployment with access to very large bandwidths. Densification enables highly energy efficient high data rate transmission due to short and often line-of-sight radio links, lower output powers and thus access to new spectrum. High gain beamforming with large number of antenna elements provides additional energy efficiency and compensates for higher path loss at higher frequencies at the same time largely reducing interference from other links using the same time and frequency resources. Unified air interface and common spectrum for access and self-backhauling with multi-hop routing enable low deployment cost, deployment flexibility to meet traffic demand, low HW cost and efficient spectrum usage and resource management.

With consideration of these aspects, a time division duplexing (TDD)-based air interface concept for UDN mobile broadband is designed and evaluated as described in more detail in the following two sub-sections and in Annex A, section 6.1.

3.1.1 Harmonized OFDM

Main principle

OFDM based waveforms are currently used in several contemporary systems, as they are key component in 4G. For 5G mobile broadband (MBB) services, they can be seen as well-suited candidates due to the provided good time-localization properties enabling low latency and low cost receiver with good MIMO/beamforming performance. Further, the OFDM out-of-band emission problem becomes negligible in dense deployment environment due to the usage of low power levels with very wide channel bandwidths. Co-channel interference can be seen as the major interference contribution instead of interference due to adjacent channels. Different OFDM variants, such as DFT-S-OFDM as well as zero-tail DFT-S-OFDM [BTS+13] can be obtained as a straightforward add-on over basic OFDM without adding significant complexity in order to achieve flexibility for diverse use cases. The application of OFDM to higher frequencies has been investigated and detailed as follows below. Note that multi-carrier waveform approaches with filtering, such as UF-OFDM/UFMC (subband-wise filtered OFDM) and FBMC (subcarrier-wise filtered OFDM with OQAM signalling) are studied in section 3.8, aiming at a more flexible use and configuration of the spectrum in mixed service scenarios.

For OFDM, when increasing carrier frequency from conventional cellular spectrum towards mmW range, it is proposed simultaneously to increase the used bandwidth and subcarrier (SC) spacing, while keeping the FFT size within a small set of quantized values. Smaller cell sizes lead to decreased propagation losses between the transmission node and the reception node, while delay spread may also be expected to decrease when moving towards higher carrier frequencies and increased utilization of beamforming. Thus, similar scaling can be done in time domain numerology, meaning that e.g. the delay spread and the related numerology, such as cyclic prefix (CP) length, may further be variable according to the carrier frequency. The principle of scalable radio numerology is referred here as harmonized OFDM and is illustrated in Fig. 3-1 with some preliminary numerology values, with the considerations mentioned above.

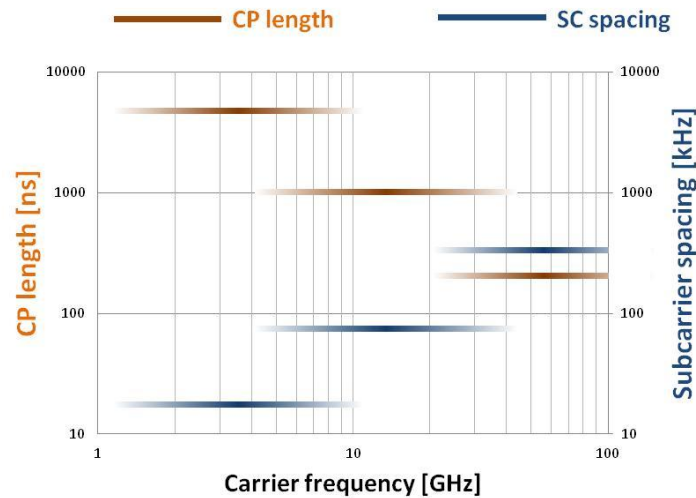


Figure 3-1 Scaling of time and frequency numerology according to harmonized OFDM principle

CP length analysis

For more detailed analysis of the values of different time domain steps in Figure 3-1, we first consider the impact of CP. UDN air interface developed in TeCC#1 was originally optimized for indoor small cells and assuming similar Tx Power at NodeB and UE, meaning that the CP and GP times were originally estimated based on cell size of 100 m (no timing alignment (TA) assumed) and RMS delay spread of 200 ns [MET14-D23]. In [MET14-D23], it was estimated that a CP length of $\sim 1 \mu\text{s}$ would be feasible for 5G dense deployments with cmW carrier frequency. We now extended this analysis by investigating the OFDM spectral efficiency performance for different channel models, such as indoor hotspot and outdoor urban micro and macro channels. See Annex A6.1 for more details of the investigation assumptions and results. Based on the results of this analysis it can be stated that to overcome the channel delay spread, constant $1 \mu\text{s}$ CP length, causing less than 6% overhead with 60 kHz SC spacing, seems to be sufficient for all investigated channel models in cmW frequency range. Correspondingly for mmW, our analysis shows the optimal CP length to be noticeably smaller. For larger cell sizes, TA needs to be applied.

Phase noise impacts to air interface design in higher frequencies

In the physical layer design for UDN at mmW bands, beamforming with large number of antenna elements is a basic component which leads to consideration of in-chip antenna and integrated RF HW. With such solutions, phase noise with current available technologies will set a cap on the maximum received SNR, which potentially improves either by technology progress over time (which though is out of our control and cannot be known in the time frame of METIS), or by the use of large subcarrier spacing (IEEE 802.11ad, $\sim 5\text{MHz}$ at 60GHz band) which leads to large overhead ($\sim 25\%$ extra, due to very short time domain symbols with desired CP length), or by phase noise estimation and compensation at the receiver which motivates this study.

In the study, measured channels provided kindly by Aalto University (cf. [HTWM11], [GHWT14] and [GTHM11]) are used. Our preliminary results show that phase noise seems not to be a show-stopper even with a subcarrier bandwidth 360 kHz which is much smaller than the one using in 802.11ad, at least not for a maximum of 16QAM and 2-stream transmission. These results are quite well aligned with the preliminary results presented in [LPVT14], covering the impact of phase noise and Doppler spread in order to find jointly optimal values for both the CP length and subcarrier spacing. For more challenging transmission schemes, phase-noise compensation algorithms can successfully remove error

floors that appear, at least under the assumption of ideal channel estimation. See Annex A, section 6.1 for details.

Based on the conducted preliminary analysis on the OFDM numerology, it can be evaluated that the estimated subcarrier spacing values even for mmW frequency area are still reasonably low and together with much shorter required CP length the overhead can be kept in feasible limits. As a conclusion, we can estimate that OFDM based modulation may still be a feasible waveform candidate even at high carrier frequencies. In order to take CPE (Common Phase Error) and different RF technologies into account, increasing the SC spacing slightly and adding some pilots for ICI estimation and compensation could further be considered, leading to a small additional increase in overhead.

3.1.2 Frame structure and dynamic TDD

Subframe structure

A TDD optimized physical subframe structure for a UDN system is illustrated in Figure 3-2² [MET14-D23], following these main design principles:

- A bi-directional (including both DL and UL resources) control part is embedded to the beginning of each subframe and time-separated from data plane.
- Data part in one subframe contains data symbols for either transmission or reception. Demodulation reference signal (DMRS) symbols, used to estimate the channel and its covariance matrix, are located e.g. in the first OFDM symbol in the dynamic data part and can be precoded with the same vector/matrix as data.
- Short subframe lengths, such as e.g. 0.25 ms on cmW frequencies when assuming 60 kHz SC spacing, are feasible. By following the principles of harmonized OFDM concept, the frame numerology is further scaled when moving to mmW, leading to even shorter frame length, e.g. in the order of 50 μ s.
- In frequency direction, the spectrum can be divided to separate allocable frequency resources.

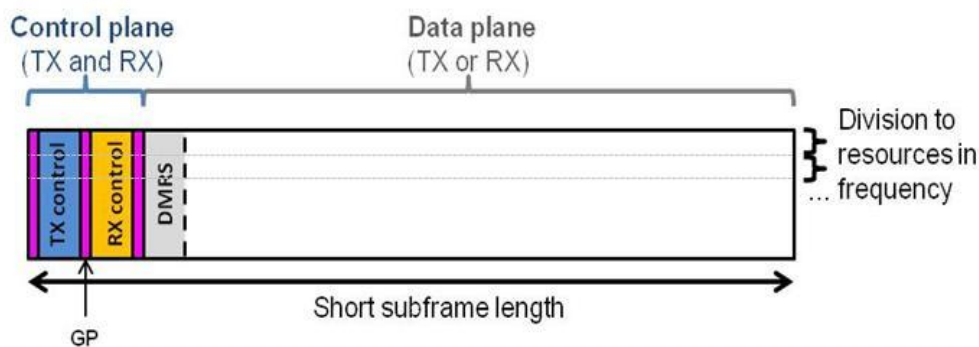


Figure 3-2 TDD optimized METIS subframe structure for UDN.

For comparison, see D2.2 [MET13-D22] for illustration and description of corresponding TDD LTE-A physical frame structure.

Control channel design principles

The bi-directional control part of the subframe allows the devices in the network to both receive and send control signals, such as scheduling requests (SRs) and scheduling grants

² Only one TDD type is presented here, in the other TDD mode the transmission (TX) and reception (RX) phases are in the opposite order

(SGs), in every subframe. In addition to the scheduling related control information, control plane may also contain reference signals (RS) and synchronization signals used for cell detection and selection, scheduling in frequency domain, precoder selection and channel estimation.

Multi-cell transmission and massive MIMO or a combination of them are essential enablers to reach the 5G capacity targets and therefore also need to be supported by control signal and RS structures. Thus, in addition to broadcast / PDCCH³ type of control, support is also needed for EPDCCH⁴ type of control signalling with device specific precoding and support for frequency domain scheduling. Both of these control types require own RS signals. Like illustrated in Figure 3-2, control symbols (including both PDCCH and EPDCCH type of control), are time separated from data plane and located before the data symbols. This is to allow fast and cost efficient pipeline processing in the receiver and also to allow devices to skip the rest of the subframe (DRX) if e.g. no data is scheduled for that user. Consequently, processing and energy gains can be achieved w.r.t LTE-A, where EPDCCH is frequency division multiplexed with data and the design is thus significantly breaking the processing pipeline (it is required to wait for the EPDCCH to be received, demodulated, decoded and checked before the related data processing can start).

Investigations related to whether the control information for different users should be coded separately or also jointly to some respect is left for further studies. Separate coding would provide support for spatial reuse and beamforming whereas a combination of joint and separate coding may provide some gains in the case of non-orthogonal user pairing utilizing advanced receivers.

Symmetrical design between DL and UL control parts is proposed. This is enabled by the properties of small cells, such as easier utilization of channel reciprocity due to the higher similarity of UL and DL transceivers, and further enables fluent support for e.g. self-backhauling and D2D.

Benefits and identified gains

The UDN optimized physical frame structure enables e.g. the following properties [MET14-D23]:

- Fully flexible UL/DL ratio switching for data transmission: Together with short TTI, this also enables efficient utilization of dynamic TDD, enabling improved total throughputs compared to LTE-A.
- Really fast network synchronization (to this DL synchronization signal) by the network devices.
- A clean TDD HARQ scheme with HARQ timing not dependent on the UL/DL ratio. HARQ timing can be fixed and counted in frames and short HARQ RTTs can be supported. [LPT+13].
- Shorter TTI length and 1ms HARQ RTT together with estimated shorter eNB and UE processing times (due to enabled pipeline processing) can be utilized to enable reduced user plane latency. DL user plane latency characterization for METIS UDN concept compared to [3GPP-36.912] is presented in Table 3-1 for cmW. See Annex A6.1 for corresponding analysis for UL. It can be estimated that ~5x user plane latency reduction can be gained w.r.t LTE-A.

³ 3GPP Release 8: PDCCH uses cell-specific reference signals (CRSs) for demodulation, not allowing e.g. UE specific beamforming.

⁴ 3GPP Release 11: EPDCCH was introduced in order e.g. to support increased control channel capacity and frequency domain ICIC, to achieve improved spatial reuse of control channel resources and to support beamforming and/or diversity. Demodulation is based on user specific DMRS (blind decoding is applied).

Table 3-1 User plane latency analysis for TDD in DL with 10% BLER

Delay component	METIS UDN (cmW)	LTE-A TDD
eNB Processing	0.25 ms	1 ms
Frame Alignment	0.125 ms	0.6-1.7 ms
TTI Duration	0.25 ms	1 ms
UE Processing	0.375 ms	1.5 ms
HARQ Re-transmission (10% x HARQ RTT of 1ms)	0.1 ms	0.98-1.24 ms
Total Delay	~1 ms	~5-6 ms*

* depends on TDD configuration

- Enabling reduced total energy consumption: ~7-40 times lower energy consumption w.r.t. MTC (machine type communication) optimized LTE-A, like illustrated in Annex A 6.1 and [MET14-D23].
- Inbuilt support for cross-link interference mitigation, further enabling usage of advanced receivers (such as IRC). [MET14-D23]
- By utilizing the enabled fast network synchronization time, flexible OFDM-based waveforms enabling link direction switching on OFDM symbol level [BTS+13] and advanced TX/RX patterns enabling virtual FDD [TRL+13], the frame structure can be extended to support very short discovery activity time for high number of devices.

Some of the properties enabled by the UDN frame structure may also be beneficial for more advanced connection types and for other METIS scenarios functioning in dense deployment environment. For example, fast network synchronization and the enabled reduced battery consumption may be seen useful for MMC devices. Also, the efficient device discovery may be seen as an enabler for D2D type of communications with improved discovery reliability.

Multi-antenna techniques

As mentioned in [MET14-D23], we assume MIMO with spatial multiplexing or beamforming for the conventional lower frequencies. For UDNs to be deployed in higher frequency bands, where radio wavelengths are substantially smaller than those in conventional cellular networks, a considerable number of antenna elements can be implemented even in nodes with small physical dimensions. For deployment in mmW bands, high gain beamforming is required to fulfil link budget to reach a certain desired throughput. Using polarization diversity, two layers can be multiplexed even for LOS links. Using differently reflected links, higher-layer spatial multiplexing can be utilised. In addition, high gain beamforming is an effective means of spatial isolation of links to obtain frequency reuse-one with simple and low cost receivers to obtain high spectral efficiency. The hypothesis was verified by simulations of a small open office with 9 Tx-Rx links which showed that a channel capacity of 14-22 Gbits/s/link can be reached with 2 GHz BW @ 60GHz with use of 64 Tx antenna elements and 16 Rx antenna elements and one layer with MRC receiver (cf. Annex A 6.1 for details). Note that the capacities are raw bits, i.e. all overheads like guards, cyclic prefix periods, control and reference signals etc. need to be taken into account to get the data throughput. Moreover, the transmitter and receiver impairments like practical signal processing algorithms and implementations, non-ideal channel codec etc. need also to be taken into account to get the actual data throughput. These aspects will be for further studies.

Self-backhauling support alternatives

With self-backhauling and multi-hop, the control information needs to be signalled among access nodes. With half-duplex communication, transmission and reception cannot be done at the same time period which would cause the problem that two nodes with the same parity, i.e. the same TX/RX timing, cannot communicate with each other with the proposed frame structure as depicted in Figure 3-3(left). One alternative to solve the problem is to use an additional control field at the end of a subframe (cf. the middle figure). Another alternative is to aggregate control signals when needed, to transmit every second subframe (cf. the right figure) which would increase latency of control signalling which might not be critical in some cases.

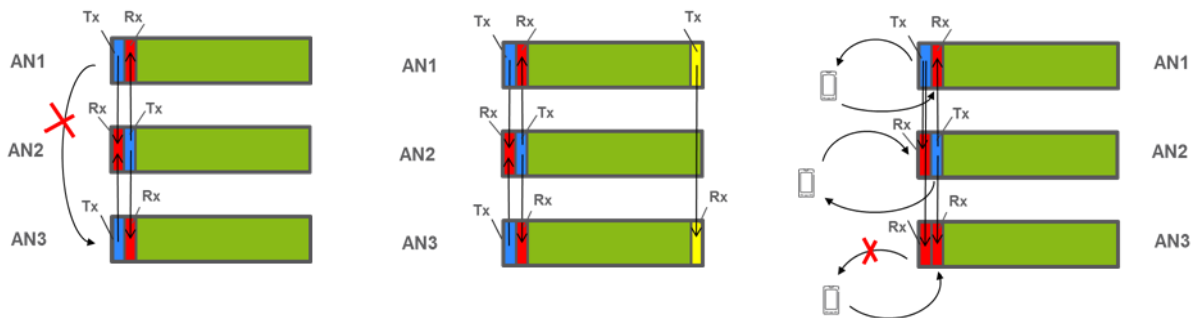


Figure 3-3 Frame structure supporting self-backhauling control signalling for half-duplex communication

Scalable latency

Like described before, TDD optimized frame structure and related numerology set hard limits for minimum achievable HARQ RTT and latency in UDN context. Scaling of numerology according to the harmonized OFDM principle leads also the total achievable latency and minimum HARQ RTT to be scalable as a function of carrier frequency, e.g. HARQ RTT is ~1ms in cmW and can be noticeably lower at mmW. In addition, scalability of the achievable latency with respect to other service requirements, such as coverage, was considered. The main idea behind this approach was to extend the proposed TDD-based METIS UDN air interface scheme to cover also somewhat larger coverage areas with the trade-off of latency. An initial concept related to this and a proposed approach based on TDD-FDD aggregation are described in Annex A, section 6.1. Performance evaluations will be for further studies.

3.1.3 Connection to TeC#12.3

The UDN PHY layer and MAC layer are considered as a combined design. TeCC#1 provides physical layer enablers for MAC solutions developed in TeC#12.3. TeC#12.3 is a complementary technical component providing MAC solution for TeCC#1. The connection of these two technical components and the connection to the technical components developed in other METIS WPs are briefly described as follows and illustrated in Figure 3-4.

Spectrum controller (WP5) determines spectrum resources a UDN may use. The UDN Resource Coordination functionality (WP3, WP4 and WP5) performs a partial resource assignment following centralized or decentralized manner depending on the deployment complexity and determines communication routes for wireless self-backhaul, the MAC determines the actually used resources on which the PHY operates (WP2).

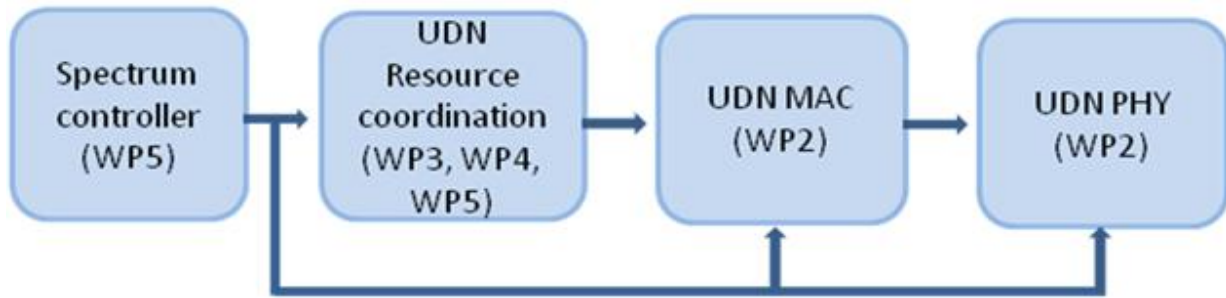


Figure 3-4 TeCC#1 connections to other METIS research

Conclusions

Targeting high data rate mobile broadband for ultra-dense deployment with efficient HW and energy cost, a flexible air-interface concept is conducted. The numerology of the unified frame structure with dynamic TDD and harmonised OFDM for spectrum ranges up to cmW and mmW bands is optimised with respect to spectral efficiency in the UDN context and user plane latency is analysed showing ~5 times reduction w.r.t. LTE-A. Phase noise impact is evaluated with measured data providing insight that OFDM based modulation may still be a feasible waveform candidate even at high carrier frequencies. Multi-antenna techniques applying to spatial multiplexing and high gain beamforming are studied and evaluated for mmW frequency with ray-tracing channel model showing high channel capacity can be reached with a simple MRC receiver and very low transmission power. Further research is needed for application in the future system, e.g. impact of practical algorithms, reference signal design, channel coding and retransmission, control signalling etc. To provide mechanisms for switching on/off UDN nodes on demand and providing seamless mobility, a concept for initial beam-finding and beam tracking has been part of the research, though further research is needed to finalize the concept and the performance evaluation.

3.2 Optimized signalling for low-cost MMC devices

3.2.1 MMC Type D2D Links

Proposal of MMC type low power and low rate D2D links that can underlay the cellular downlink transmission while providing link reliability assurances to both MMC and cellular links.

In this TeC, the challenge to provide optimized signalling for low-cost MMC devices is accomplished through cellular Machine-Type D2D connections between low-cost cellular Machine Type Devices (MTD) and cellular users. This enables communications between low-power devices and cellular users, as well as offering access load relief in case of massive access, as illustrated in Figure 3-5. We enable the underlay communication between MTD and cellular users, while being interfered by a cellular downlink network connection.

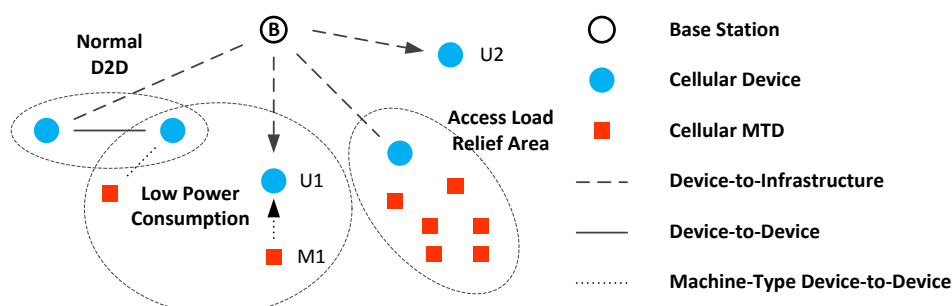


Figure 3-5 Illustration of the different opportunities opened by D2D to enable MMC.

Specifically, we consider the simultaneous occurrence of the links M1-U1 and B-U1, as illustrated in Figure 3-5. The link M1-U1 is characterized by low and fixed rate, while the link B-U1 is characterized by fixed power and adaptive rate. The work here presented has taken both theoretical and system level simulation viewpoints, where the later was based on the METIS TC2 [MET13-D11]. The theoretical part of the work concluded in the definition of zero-outage rate bounds for the link B-U1 assuming infinite block length, while considering both the Joint and Single-User multiuser decoding settings [PP+14]. The zero-outage rate upper bound for the link B-U1, is defined as follows,

$$R_B^{JD} = \log_2 \left(1 + \frac{\gamma_B}{(1 + \Gamma_M)^{N_M}} \right) \quad R_B^{SD} = \log_2 \left(1 + \frac{\gamma_B}{1 + \Gamma_M(1 + \gamma_B)} \right)$$

Where R_B^{JD} and R_B^{SD} denote respectively the maximal downlink transmission that is always decodable for the Joint and Single-User multiuser decoding settings. γ_B , Γ_M and N_M denote respectively the instantaneous SNR realization of the B-U1 link, the equivalent SNR rate of the MTD transmission ($R_M = \log_2(1 + \Gamma_M)$) and the number of MTD links active transmitting to U1. Note that in the Single-User setting, a zero-outage rate only exists for a single MTD link active, while for the Joint Decoding setting it exists for multiple MTD links.

In Figure 3-6 is depicted the achieved zero-outage rate in the downlink for the two considered decoding settings. It can be observed that the higher is the rate of the MTD transmission, the higher is the penalty on the rate achievable by the downlink transmission. It can also be observed that when the MTD rate is very low, the downlink rate tends towards the case where the M1-U1 link is not present, which is of particular interest in a MMC setting.

In conclusion, the defined zero-outage rates although providing outage guarantees in the case where no CSIT is available at B, the achievable rate in the B-U1 link is lower than the one achieved if full CSIT of all the links was available at B. Further numerical results and an evaluation in a LTE setting are found in Section 6.2.1. Other results that consider a similar network scenario, where the downlink transmission is not targeted for the cellular U1 are presented in detail in [MET14-D23] and [PP2+14].

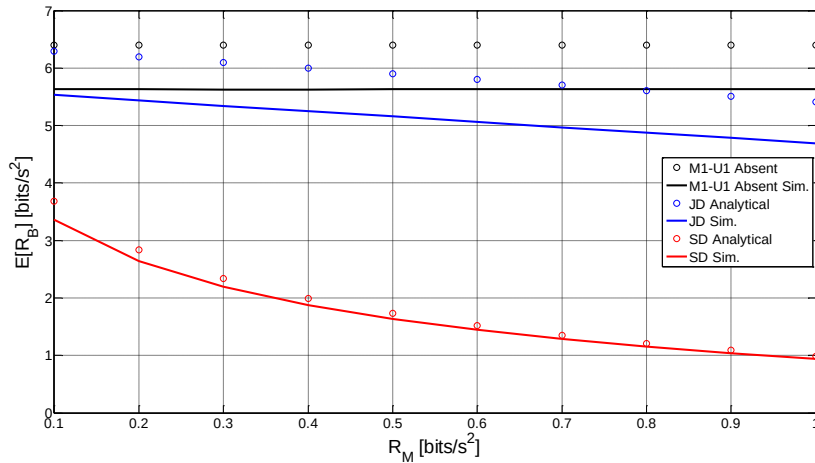


Figure 3-6 GRM1#Throughput - Comparison of the derived analytical upper bounds for $E[R_B]$ with the simulations for the Joint Decoding and Single User Decoding settings, when a single Machine Type Device is associated with the cellular user U1 [PP+14].

Conclusions

The proposed underlay technique and the achieved theoretical results demonstrate that the underlay of several MTDs in a cellular downlink transmission is possible. When the MTDs rate is extremely low a large number of MTDs can be supported (assuming the Joint Decoding setting).

3.2.2 Quasi-Orthogonal Random Access

A quasi-orthogonal random access combined with MMSE-based multiuser detection for MMC was investigated. A narrow-band design was considered using CDMA spreading in the time domain in order to enable low-power (and in turn low cost [NN12]) device implementations, where uplink transmission is contention-based using the Slotted ALOHA protocol. Quasi-orthogonal spreading sequences are preferable as they allow larger code pool and thus reduce collision probabilities. System simulations were carried out to assess delay and collisions statistics as well as achievable capacities in terms of number of devices that can be supported per cell. Further details can be found in [NSW14].

3.2.3 Hybrid Access Scheme for reduced signalling overhead

This TeC aims at efficient schemes for Multiple Access (MA) for machine type communicating devices by exploiting the periodicity of some machine type devices & smaller packet payloads.

The explosive growth of Machine-to-Machine (M2M) devices (e.g., sensors, smart meters, industrial machinery, etc.) requires efficient multiple access schemes with respect to reduced packet drops and collisions. Diversity of some M2M requirements (smaller payloads, periodicity/sporadicity of access etc.) can be exploited in order to design new multiple access protocols for 5G.

The key component is the M2M Access Manager (MAM) that classifies the devices according to their traffic history and characteristics and other properties such as acceptable delay into a contention-based uncoordinated access and a contention free co-ordinated access as shown in Figure 3-7. The device properties allow a degree of autonomy for the MAM in case of applications that are delay/error tolerant. Two orthogonal channels were assumed for both the kinds of machine type devices.

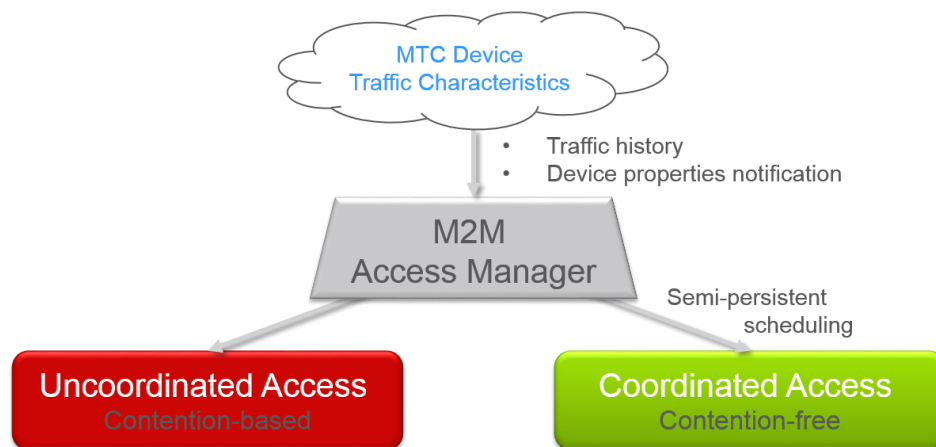


Figure 3-7: MAM Scheme

The Uncoordinated Access channel is efficient for sporadic transmissions with no/limited access grant procedure whereas the Coordinated Access channel is efficient for periodic transmissions with network control and generally result in collision-free transmissions whereas uncoordinated channel may result in periodic collisions.

The proposed scheme was been compared to the slotted ALOHA MA scheme employed presently in the LTE networks. The MAM classifies a device depending upon the Inter-Arrival Times (IAT) of the last request and the counter threshold (set to 10 in our evaluation). If the IAT is constant over the last 10 requests, the device is considered to be periodic and assigned the coordinated access channel and vice versa. The evaluation parameters and methodology were highlighted below.

Table 3-2: Simulation Assumptions

Simulation Duration	12 hours	Modulation	QPSK
Simulation time basis	$T_I = 100ms$	Bandwidth per channel	180 KHz (1 RB)
Number of Devices - periodic - event-driven	200 80	Resource Allocation (RA / Sched.): - Reference Case - Hybrid Scheme	2 / 0 channels 1 / 1 channels
Delay constraint	$D_{max} = 5s$	Random Access Scheme Backoff-time Max. number of retries	Slotted ALOHA Uniformly distr. [0,2] sec 2

Evaluation Results

The proposed MAM scheme outperforms the slotted ALOHA model (with 2 orthogonal channels) which can be seen from Figure 3-8 . The packet drops (collisions) are reduced by 20% whereas the total throughput is increased by 40%. The proposed scheme performs similarly with the reference case (in terms of packet drops) in the initial time period, during which time the MAM learns from the traffic characteristics of the device and classifies it accordingly. Once the classification stabilizes over time, performance gains are achieved due to the semi-persistent scheduling.

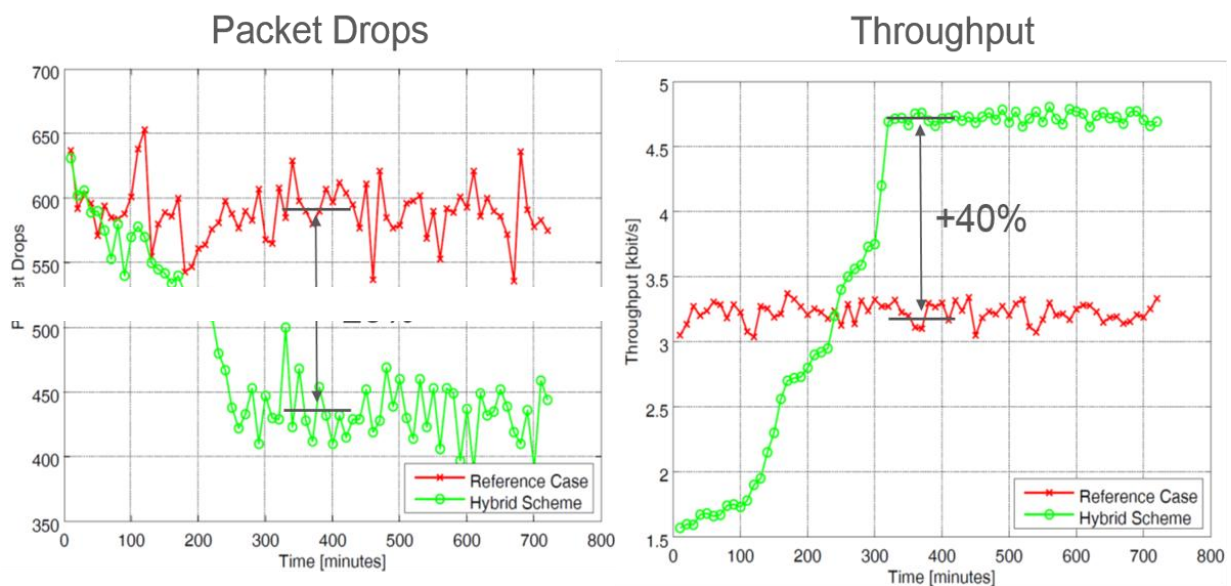


Figure 3-8: Key Performance Indicators

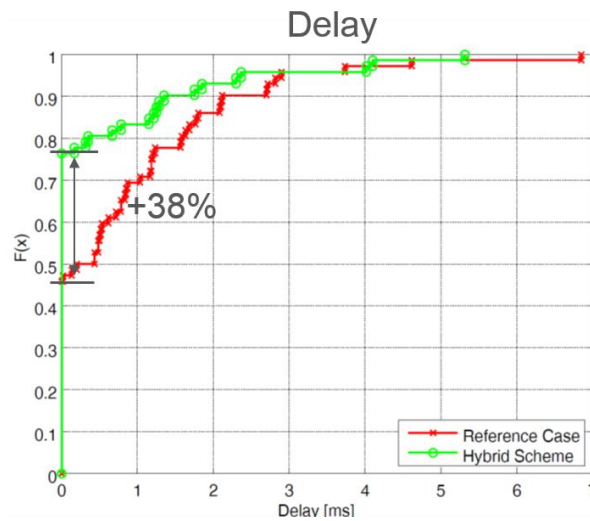


Figure 3-9: MAM delay

However, the MAM scheme also induces a processing delay and increase the overall end-to-end latency at the beginning. This “learning phase” delay of MAM is due to the classification algorithm that uses a minimum counter value in order to classify the device as either periodic or sporadic. Figure 3-9 shows the increase in delay (+38%) due to the initial learning phase. However this delay also constitutes the random access delay induced when a periodic device access collides with another. In that case, two successful retries (total 3) are made with a random delay in between. These collisions as well as the increased delay only hold for the learning phase. Once the system is in a stable state, no increase in delay is observed.

Conclusions

The proposed multiple access scheme (MAM) outperforms the slotted ALOHA model with respect to throughput and collisions with a trade-off of increased delay. However, this can be leveraged by the multitude of device characteristics that can be used as input to make the access scheme more efficient (e.g., target reliability, priority etc.)

3.3 RF Architecture for Dynamic Spectrum Access

The aim of this TeC is the system design of a frequency agile front end enabling dynamic spectrum access.

For the dynamic spectrum access, or in other words multiband access, state-of-the-art transceivers employ homodyne or direct conversion architecture [HW11]. This calls for “distinct” RF chains with mixing and filtering with different bulky RF filters corresponding to the different bands, leading to an increase in effective area and bill-of-materials (BOM). This TeC replaces the traditional homodyne chains with a single high-IF architecture utilizing only one IF filter. Due to this architectural change, the transceivers can attain the RF performance such as linearity, noise and frequency selectivity without an expense of area and cost as expected from a traditional multi-standard homodyne architecture as described above. The high-IF operation and the comparison with a typical low IF is shown in the following Figure 3-10.

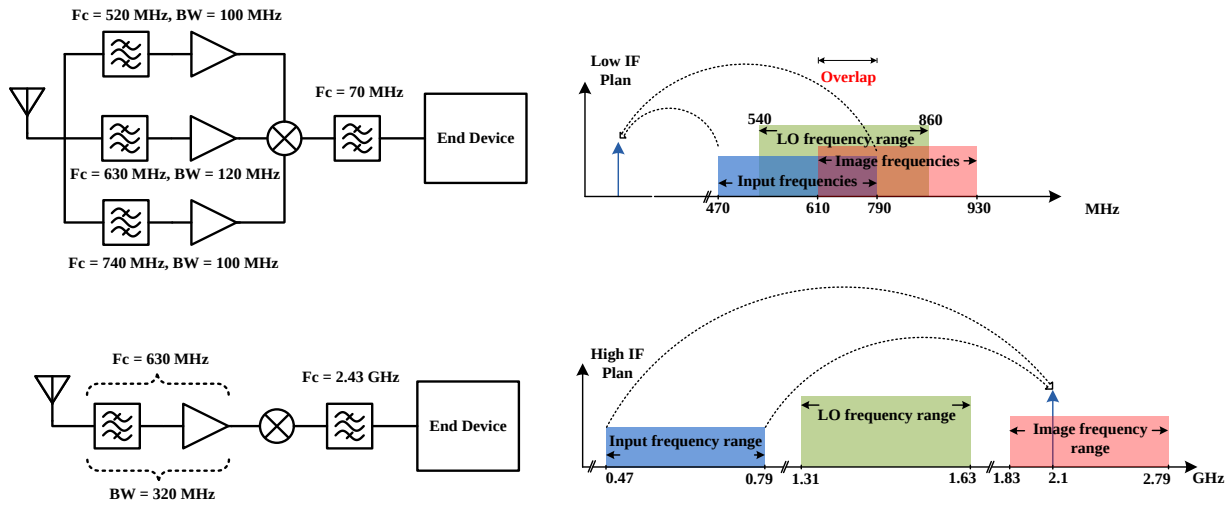


Figure 3-10 Frequency plan a traditional low-IF and high-IF architecture

In the above figure, high-IF of 2.1GHz has been selected for reference. All the frequencies from 470MHz to 790 MHz are upconverted to 2.1GHz for a high-IF receiver. The intermediate frequency or the frequency after mixing referred as IF in the top figure case is 70MHz. It can be learned from the above figure that while for low IF image and input frequencies are overlapping, for high-IF they are separated. This can improve the performance under wide band conditions. Following table shows the system performance with baseline homodyne architecture as already mentioned in METIS D2.3 [MET14-D23].

Table 3-3 performance of Homodyne with high-IF architecture

Parameters	Homodyne	High-IF
Sensitivity	-103.6 dBm	-107.3 dBm
NF	8.8 dB	4.5 dB
IIP3(in band)	-8.3 dBm	-7.7 dBm
IIP3 (out-of-band)	-8.3 dBm	0.8 dBm
Dynamic range	95.3 dB	108.1 dB

Under the wideband conditions the high-IF provides significant improvement to the noise figure (NF) and linearity (IIP3). Detailed performance results for a developed prototype can be found in [ASV+14a].

Conclusions

Under wideband conditions high-IF architecture provides better RF performance than the traditional homodyne architecture requiring less area and cost. Such a front end can be combined with any off-the-shelf commercially available transceiver, converting the whole into a highIF system leading to great deal of flexibility and faster implementation. One potential application is the cognitive radio front ends, where the concerned radio front end should possess high linearity and frequency selectivity under wideband conditions [ASV+14b].

3.4 Multiple air interface management

3.4.1 Multiple Interface Management in MT-HetNets

This TeC addresses Multi-Techno HetNet cross layer mechanisms thanks to dedicated Multi Techno Link Adaptation (MT-LA) metrics integrated in Multiple Interface management modules and RF scalable multi-band processing compliant with single and multiple Air Interface.

The TeC#4.1 fulfils 5G technical requirements and METIS technical challenges in investigating Multi-RAT HetNets system design at PHY and MAC layer levels to implement Inter-RAT management increase the multi-user throughput and perform multi-RAT network densification. Multi-Band (MB) processing using a common air interface able to switch between different RF bands is a simpler alternative to Multi-RAT PHY/MAC layer processing.

Multi-RAT PHY/MAC CQI metric and integration

For that purpose, 3 aspects have been examined in the TeC#4.1.

1) Definition of Multi-RAT link adaptation CQI metrics and performance

- A novel multi-RAT CQI metric, denoted Green Link Budget (GLB) has been designed, using link budget parameters to perform Air Interface (AI) and Transmission Mode (TM) selection in multi-RAT equipment, following green power criteria [SUM14] [SUM13] [UMS12]. The GLB metric is composed of 2 sub-metrics α and β . The α -metric measures the *relative degradations* at the PHY/MAC layer with respect to AWGN and the additional path-loss involved by obstacles with respect with Free Space Path-Loss respectively. The β -metric evaluates the *excess power level* between the *required power level* (Multipath Power Sensitivity denoted S_M) to guarantee QoS and the *Available Radiated Power* ($ARP(d,fc)$) at the receiver side, depending on transmit power level, antenna gains, distance $d(Tx-Rx)$ and propagation conditions. The AI and TM selection resorts from a α -metric minimization and a β -metric maximization. The β -metric performs dynamic power control of the AI/TM selected by the α -metric (see annex A, section 6.4.1).
- The α and β -sub metrics are proper quantized metrics supplying gained performance of MIM and MB mechanisms. The α -metric provides Multipath Power Sensitivity gains when switching AI/TM to another one. This gain may be explicitly translated in transmit power reduction and radio coverage extension by considering the β -metric values.
- The impact of propagation conditions on the GLB metric variations has been examined. Refined TGac path-loss models [TGac09] have been defined for Wi-Fi Hot Spot into a LOS/NLOS separation and different distance ranges. New mm-wave path loss models have been developed for Wi-Fi Hot-Spot extensions, combining indoor attenuation with Oxygen absorption and rain fall rate elements.
- The GLB metric has been evaluated upon METIS Test Cases (TC) {TC n°1, 2 & 3} [MET13-D11]. Different MIMO schemes (SDM, SS and STBC with variable efficient bandwidth sizes) of the IEEE802.11 ac/n standards have been evaluated and collated with the UWB ECMA-368 standard [ECM08] transposed at 60 GHz. Results highlight gains up to 10 dB with AI/TM switching in presence of propagation variations (see annex A, section 6.4.1). Evolved TM has been designed for the ECMA-368 and IEEE802.11ad standards in order to extend multiple data rates for variable QoS services [SUM09] [BSUM14a]. Results show that NLOS/LOS transitions, with 2Gbs data rate, involve AI/TM switching processing, supplying 2-6 dB gain on performance and transmit power reduction.

- A new MT-LA metric, denoted γ , has been combined with the GLB metric for the MT-LA processing, in order to encounter random access based multi-user resource access requirements. This metric has been assessed for mm-wave and Wi-Fi hot spot TC [BSUM14b].
- 2) The assessment method of multi-RAT CQI metrics using available elements in PLCP headers of considered interfaces and measured parameters (RSSI, Channel estimation) at the User Equipment (UE) has been defined and applied to IEEE802.11ac/ad standards. The proposed method to practically measure the $\{\alpha, \beta\}$ metric, exploits normalized multipath propagation parameters and accurate BER look up tables differentiating LOS/NLOS criterion. Normalized parameters are derived from equalization processing performed at the receiver side and PHY system parameters.
 - 3) The implementation of Multi-RAT LA techniques needs to specify multi-RAT architectures able to integrate multi-RAT CQI metrics in existing mechanisms. Several architectures are proposed in the annex 6.4, related to 3 multi-RAT scenarios ensuring backwards compatibilities with standardized mechanisms. The Fast Session Transfer (FST) protocol is considered for IEEE802.11n,ac/ad switching and an additional L2.5 layer is envisioned for independent AI.

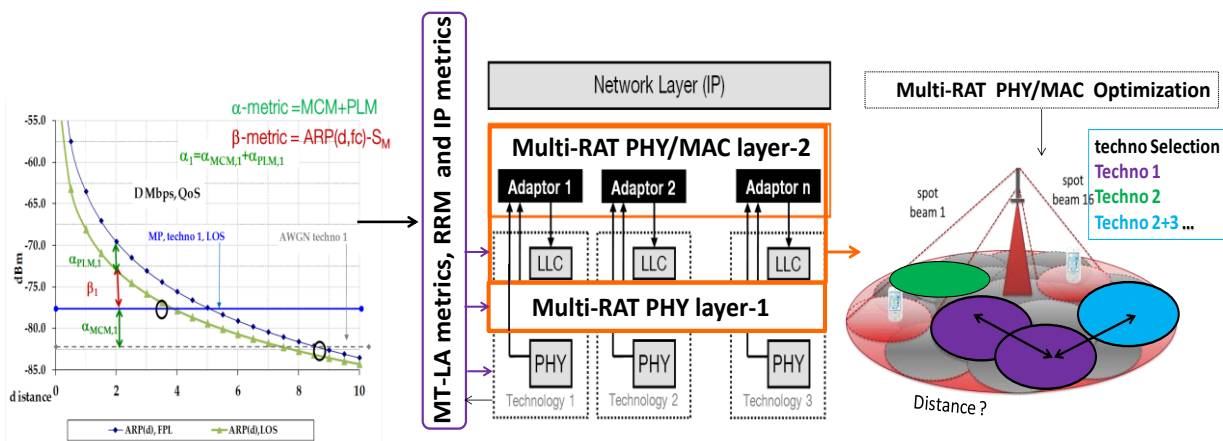


Figure 3-11 Multi-RAT PHY/MAC layer system design

Figure 3-11 illustrates, from left to right, the different steps of definition, implementation and optimization of LA metrics in MIM architectures and the network densification in radio engineering processing. Detailed results are given in Annex A, section 6.4.1.

Multi-Band processing for RF Multi-Band Air Interface

Multi-Band processing addresses air interface possibilities to switch between different RF bands (i.e. licensed and unlicensed bands), regarding multipath propagation signature facing to PHY/MAC layer parameters (OFDM CP length, sub-carrier spacing scalability, harmonized with RF stages, etc...). Possible extensions to additional deployment scenarios (indoor extended to outdoor for network densification) are also investigated.

As a starting point, an analysis has been carried out on previous ICT-FP6-FP7 projects investigating dual-band systems [SUM09] [BROAD02] able to locally and temporally increase throughput following traffic and QoS on demand. An analysis has been performed regarding WiFi Standards (IEEE802.11ac/ad) and UWB standards transposed in mm-wave bands to extend the radio coverage. RF front end architectures are designed to switch between UWB, IEEE802.11ac/ad bands derived from MB processing developed in the ECMA standard and previous studies [GSG08] led in the ICT-FP7 OMEGA project [OMEGA].

Conclusions

The TeC#4.1 provides proper link budget based GRMs to evaluate multi-RAT LA techniques in MT HetNets. GRMs resorts from the $\{\alpha, \beta\}$ metric values giving transmit power reduction levels and radio coverage extension when performing AI/TM switching. Another metric for random access take into account of time resource occupancy. These metrics have been evaluated for Virtual Reality Office TC and Wi-Fi Hot spot extensions in Ultra Dense Network TC. Practical assessment methods are proposed to implement these metrics as well as multi-RAT architectures. The next step is to evaluate MT-LA algorithms in a multi-user context and system level simulator. Network densification integrated in radio engineering platform is pursued in the ICT-FP7 MiWEBA project [Mi-WEBA].

3.4.2 Software Configurable Air Interface

Software Configurable Air Interface (SoftAI) is a framework concept that allows the adaptation of different air interface components to provide the flexibility in adapting the diverse characteristics of future applications.

SoftAI can be realized via adaptation configuration of various air interface building blocks. The procedures to realize air interface adaptation are shown in Figure 3-12. Different candidate schemes for each building block are predefined. Based on a set of input parameters such as the transmission content (e.g. application types) and transmit/receive conditions (e.g. channel variation, mobility information, capabilities of transmitter and receiver), decisions are made to select the best configuration. A SoftAI configuration is formed by a set of candidate technologies from one or more building blocks. Multiple configurations can be defined to support different scenarios. Multiple configurations can also co-exist in the system.

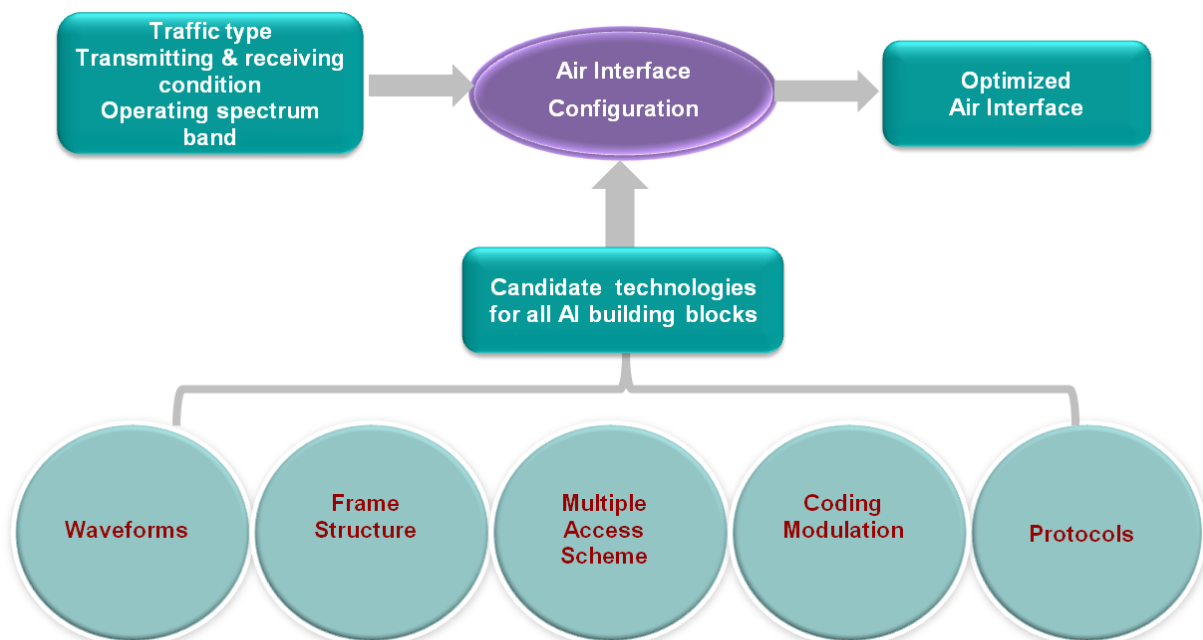


Figure 3-12 Software Configurable Air Interface enabled by adaptation of different air interface building blocks.

SoftAI can form a customized air interface for an application or deployment scenario. Based on the scenario, a predefined SoftAI configuration is applied. One example of a customized SoftAI configuration is that of a UDN configuration in which all the air interface parameters and technologies are pre-defined. Another example of SoftAI adaptation is for machine type communications in which e.g. contention based multiple access with SCMA may be more

suitable than a request/grant mechanism. Further discussions of this particular configuration can be found in TeC#11.1.2 in this document and in T4.2 TeC#16 of [MET14-D43].

Conclusions

The key features of SoftAI are the configurability and adaptability of air interface components according to various conditions. It can be a customized configuration or more dynamic adaptation of components depending on scenarios.

SoftAI can provide the best user experience. It is a framework concept to provide an integrated approach to enable various TeCs to realize a flexible air interface.

3.5 Advanced signalling concepts

Aspects of advanced signalling concepts for non-orthogonal multiple access (NOMA) are discussed in section 3.11.1. Since the former was the only TeC in this TeC cluster, it has been closed during the project.

3.6 Air interface for moving networks

3.6.1 Framework for URC

The TeC proposes a universal system concept for Ultra-Reliable Communications (URC).

The system concept proposed in this TeC is motivated by the strict reliability requirements of some applications such as vehicular and industrial applications, and by the fact that wireless communication systems cannot be designed to provide reliability at all times.

As illustrated in Figure 3-13, the system concept is based on a “Reliable Transmission Link” (RTL) that is optimized to transmit packets successfully and within a predefined deadline, and an “Availability Estimation and Indication” (AEI) mechanism that is able to reliably predict the availability of the RTL under given conditions. In addition, the system concept incorporates a novel link control indicator called “Availability Indicator” (AI) that signals the outcome of the AEI to the application. In this context, an application requests an RTL by sending an Availability Request (AR) to the AEI. Depending on the implementation details, the AR contains information such as the packet size, the maximum acceptable delay until successful reception or the maximum tolerable error probability. The AEI is designed to indicate to the application the availability of the RTL for the forthcoming transmissions given the AR requirements. For the availability estimation the AEI needs to monitor the channel conditions, e.g. by evaluating the Signal-to-Noise and Interference Ratio (SINR) and/or the ACK/NACK statistics of the retransmission protocols used at link level. Typically, the AI is a binary value, i.e. either RTL available (AI=1) or unavailable (AI=0). After indicating the RTL availability, the application will be able to use it by transmitting data packets over the RTL (not shown in Figure 3-13). Depending on the implementation, other options might be reasonable as well (e.g. the AI is a soft value that indicates the likelihood that a “reliable transmission link” is available).

During the METIS project, the concept has been mathematically formulated and evaluated by means of system level simulations. For more information regarding the mathematical formulation and a possible practical implementation of the system concept in automotive scenarios, the reader can refer to [SSG+14]. The simulations compute the reliability on the direct communication links, and also the percentage of instances in which the communication link is available (i.e. used). In this sense, reliability has been defined as the successful transmission of a data packet below a certain deadline according to [MET13-D21]. For the computation of the availability it is assumed that a packet is only transmitted in those

instances for which the reliability can be satisfied based on reliable prediction methods. In particular, the results are computed for the case with and without retransmission and for three different prediction methods. The reliability requirement to be satisfied has been set to 99.999% according to the requirements in [MET13-D11].

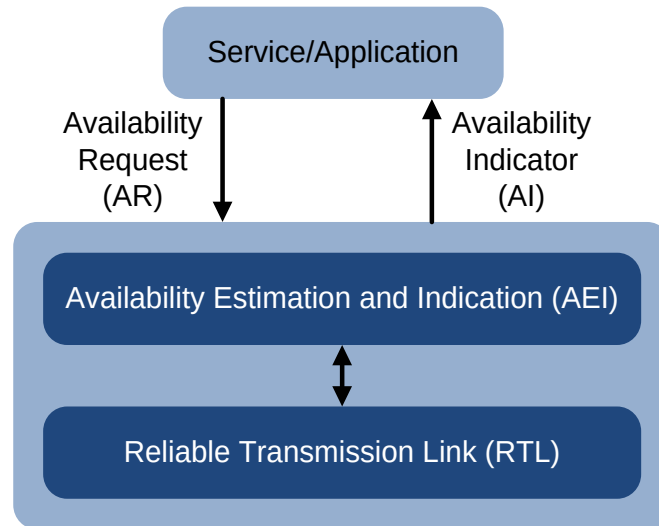


Figure 3-13 Framework for URC

The simulations consider a V2V link underlying a cellular network with 19 hexagonal layout cells and an inter-site distance of 1000 m. A use case is assumed in which 1600 bytes must be transmitted within 6 ms, which corresponds to six TTIs of 1 ms each. In the case without retransmissions the 1600 bytes are allocated over six TTIs, which corresponds to 2134 bits per TTI using the MCS scheme 18 (QPSK, coding rate 3/4). For this configuration, the minimum SINR required to ensure a successful transmission is 14.3 dB. All six transmissions need to be successfully received in order to guarantee a successful delivery of the whole packet. In the case with retransmissions, the 1600 bytes are distributed over three TTIs whereas the remaining three TTIs are used for retransmissions. This corresponds to 4267 bits per TTI using the MCS scheme 23 (64 QAM, coding rate 5/6). For this configuration, the minimum SINR required to ensure a successful transmission is 19.1 dB.

Regarding the prediction methods three different approaches have been considered:

- A scheme that knows in advance the experienced SINRs of the six transmission TTIs (prediction method B), which represents an upper bound of the achievable availability as it considers perfect channel estimation.
- A simple case (prediction method S), which predicts the SINR by taking the minimum SINR of the six TTIs in the previous transmission. It is important to highlight, that this simple prediction approach does not consider the channel variations between time slots due to fast fading, and therefore, is expected to decrease the performance in fast varying channels.
- A scheme based on the simple case but that adds a certain margin to the decision threshold in order to compensate for the prediction errors of the simple prediction approach. The margin is found *a posteriori* so that a reliability of 99.999% is always satisfied.

Table 3-4 Availability and reliability for the three considered prediction methods and different Doppler frequencies

Doppler and Probability		Configuration without retransmissions			Configuration with Retransmissions		
Doppler Frequency	Prob.	B (%)	S (%)	M (%)	B (%)	S (%)	M (%)
10 Hz	Availability	53.3	55.2	28	44.3	26	20.7
	Reliability	100	95.6	99.999	100	99.96	99.999
	Margin	0	0	4.5	0	0	1
33.3 Hz	Availability	48.3	52	1.5	39.7	23.3	5.7
	Reliability	100	84.5	100	100	94.5	99.999
	Margin	0	0	14	0	0	6
100 Hz	Availability	38.2	46.9	0.8	37.8	19.3	3.14
	Reliability	100	71.1	99.999	100	93.3	99.999
	Margin	0	0	15	0	0	6
200 Hz	Availability	35	43	0.7	34.1	18.7	3.1
	Reliability	100	67	100	100	91	99.999
	Margin	0	0	15	0	0	6

Table 3-4 contains the relevant probabilities for both transmission schemes (with and without retransmissions) and the three prediction methods.

The results show that, independent of the Doppler, it is possible to satisfy a reliability requirement of 99.999% as long as the propagation channel (i.e. the SINR) can be predicted perfectly (prediction method B). On the contrary, the reliability requirement cannot be satisfied with the prediction method S due to the very simple channel prediction that is performed in this case. The degradation increases significantly for higher values of Doppler as a result of the faster variations of the propagation channel. In particular, the reliability can deteriorate to values below 70% for Doppler frequencies above 200 Hz if retransmissions are not used. It must be noted, that for the prediction method B, a reliability of 100% was targeted in order to illustrate the best case scenario for reliability.

The results also illustrate that the imperfect channel prediction of method S can be compensated by adding a certain margin to the predicted SINR (prediction method M) so that the reliability requirement is always satisfied. Nevertheless, this is achieved at the cost of lowering the availability. For example, the availability in the case of Doppler frequencies above 200 Hz is below 1% if retransmissions are not used. This behaviour illustrates the trade-off between reliability and availability of the proposed URC concept. In this manner, it would be possible to increase the availability values illustrated in Table 3-4 by decreasing the reliability requirements of the application. For example, by extending the maximum acceptable deadline beyond 6 ms, or by decreasing the packet size below 1600 bytes, more robust modulation and coding schemes and a higher number of retransmissions can be employed.

It is also important to highlight, that a smaller margin for the prediction approach S is required for the case with retransmissions, since the fast channel variations in high Doppler scenarios can be partially compensated by means of improved time diversity. Regarding the availability, it can be seen that for the same amount of resources (six TTIs) the use of retransmissions

achieves better performance for high Doppler frequencies, whereas the configuration without retransmissions is preferable for lower values of Doppler. This illustrates the benefits of configuring the transmission link according to variability of the propagation channel, which in turn is conditioned by the relative velocity between communicating nodes.

Conclusions

The framework for URC that has been investigated in METIS lays the foundations for the support of URC in future 5G mobile communication systems. It is considered as key for the introduction of new services such as road safety and industrial applications. The simulations results show how the proposed concept can guarantee very high values of reliability as long the channel prediction is accurate enough.

3.6.2 Modelling & Predicting the Reliability of a Link

This TeC aims at modelling and predicting the reliability of the communication link according to application requirements and based on the statistics of the propagation channel.

Prediction of Link reliability enables opportunistic scheduling of ultra-reliable services/applications when the link conditions are *fair* enough [SSG+14]. The present channel prediction techniques largely depend upon SINR and CQI metrics and do not explicitly consider delay or other application specific constraints. However, complicated prediction techniques induce computational and signalling overhead into the already capacitated mobile networks. Since 5G networks are ubiquitous and distributed with enabling technologies like D2D communication, offloading the link prediction to the mobile node contributes to higher network efficiency.

In this regard, the inbuilt computation power inside vehicles and other industrial equipment can be exploited since they are the most promising candidates which require high reliable connectivity.

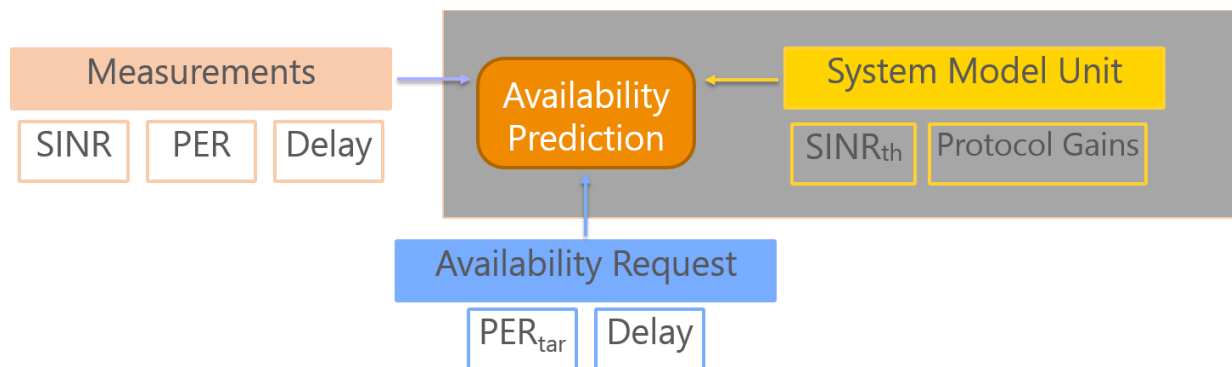


Figure 3-14: URC Implementation Example

Figure 3-14 shows an example logical implementation of the proposed prediction method. The Availability Request (AR) [TeC6.1] is assumed to be provided by the application/service requesting a Reliable Transmission Link (RTL) consisting of information such as maximum tolerable delay and the reliability requirement. The Availability Prediction (AP) is in charge of predicting the availability of RTL in future time slots based on the AR, instantaneous measurements (SINR, Delay etc.) and the System Model Unit (SMU). The SMU stores pre-computed data such as radio-maps, route and trajectory information in case of mobile UE's, queued applications requesting RTL etc. It can also be configured to store short-time local RSSI heat maps generated by instantaneous measurements.

Prediction methods

The wireless communication can be modelled as a sufficiently complex stochastic process (random function of time). Reliability Engineering [Bir10] provides accurate models for predicting the Availability of stochastic processes depending upon the *failure-free* $\lambda(t)$ and *repair* time $\mu(t)$ distributions (See Annex 6.6.2 for more details)

Performance Metrics

Two Performance metrics are used for evaluating the reliability of an RTL. They are

- 1) **Reliability Function ($R_{S_i}(t)$):** gives the probability that the RTL is available in $(0, t]$ given available at $t=0$. It is a mathematical expression analytically relating the probability of successful transmission to time
- 2) **Point Availability ($PA_{S_i}(t)$):** gives the probability that the RTL is Available (RTL=1) or in the set of Available states at time t .

Evaluation using Continuous Markov Models

In order to evaluate the applicability of the Markov model, three services with varying reliability requirements were considered – High, Medium and Low over reliability states denoted by numbers from 1-100 (i.e., State 1, State 2,...State 100) for the purpose of evaluation. State 100 is also referred to as the absorption state since reaching this state terminates the semi-Markov process.

The PA is calculated as the probability of RTL=1 over time t . Since, we depend upon accurate distributions for our evaluation, the time scale and interval can only be defined once we have the mobility model in place. Hence, for a start, we assume time to be dimensionless. The rationale behind taking three different classes of applications is to show how the PA varies over time for each of the service. The services are classified into high medium or low based upon their UP states. An application with high reliability constraints will have very few Up states and conversely a low reliability application can still be able to work in more states. Moreover, such partitioning of RTL reliability into multiple decrement states also enables Reliable Service Composition (METIS) i.e. graceful service degradation. This is also validated by the PA in Figure 3-15 where the RTL availability decreases sharply over time for the high reliable service as shown in the figure (blue). The low reliable service (green) has higher values of availability over time and this validates that the Markov model may be able to predict the RTL availability approximately.

The sojourn time distribution is used to calculate the $R(t)$ of the system. It gives the percentage of times spent in each state. From Figure 3-15, it can be seen that the transmission stays longer in reliability states (1-20 since they are sorted in descending order). A high level conclusion can be made in this regard that the present channel configuration supports Ultra reliable transmissions. However, more analytical analysis is required for further conclusions.

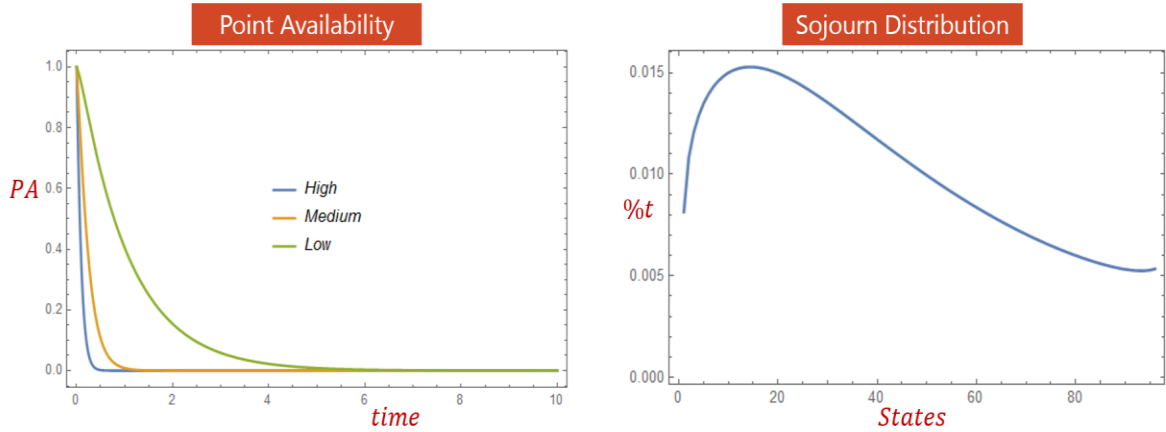


Figure 3-15 Performance Metrics for Availability Prediction

Conclusions

This TeC proposes a novel link prediction method based upon reliability engineering for realizing URC in 5G systems. The availability of the RTL is calculated based on diverse Application requirements and the instantaneous link conditions. The proposed prediction method also enables graceful service degradation instead of “Always best performance” presently offered by LTE networks.

3.6.3 Channel Estimation for V2V Links

The main idea is to exploit the structure found in the delay-Doppler domain in V2V links for channel estimation.

Geometry-based stochastic channel models have been shown to be accurate models of V2V channels. In such a model, the transmitter, receiver, and a (large) number of scatterers are placed in geometry, e.g., a highway. Each scatterer gives rise to a multipath component (MPC), which is characterized by its complex gain, delay, and Doppler frequency. Measurements have shown that there is significant structure in the MPC distribution. In fact, the channel representation consists of a relative small number of strong MPCs and a large number of weaker MPCs, called discrete MPCs and diffuse MPCs, respectively. The diffuse components are concentrated to certain delay-Doppler regions and the channel displays a mixed element-wise and group-wise sparsity. That is, the discrete MPCs are sparsely distributed in the delay-Doppler domain, while the diffuse MPCs are clustered into groups in the delay-Doppler domain. We have therefore suggested to estimate the MPC parameters, and thereby the channel, by applying sparse estimation techniques. A number of algorithms have been developed and compared with the well-known regularized least squares (LS) estimator.

The system model is quite general: we transmit a preamble signal formed by pulse amplitude modulating N_r complex samples with some pulse shape. In the simulations, we used a root-raised cosine pulse shape, but we can in principle use any pulse shape. Hence, the transmitted signal can, e.g. be an OFDM signal. The receiver consists of a receive filter followed by a sampler. Channel estimation is based on N_r received samples, collected into the vector $\mathbf{y}$. It can be shown that

$$\mathbf{y} = \mathbf{A}\mathbf{x} + \mathbf{z}$$

where $\mathbf{A}$ is a known matrix (which can be computed from the training symbols and transmitter and receiver pulse shapes), the elements of the noise vector $\mathbf{z}$ are independent, identically distributed (i.i.d.) complex Gaussian random variables, and the channel vector $\mathbf{x}$ is a sparse vector [BSM14, BMS14]. Each nonzero element of $\mathbf{x}$ corresponds to an MPC. The element

value is the MCP complex gain, while the position of the nonzero element determines the MPC delay and Doppler. Channel estimation is therefore equivalent to estimating $\mathbf{x}$ from $\mathbf{y}$. The straight-forward estimator is a regularized least-squares (LS) estimator, defined as

$$\hat{\mathbf{x}}_{LS} = (\rho^2 \mathbf{I} + \mathbf{A}^H \mathbf{A})^{-1} \mathbf{A}^H \mathbf{y},$$

where ρ is a regularization constant. The proposed estimators are of the form

$$\hat{\mathbf{x}} = \arg \min_{\mathbf{x}} \frac{1}{2} \|\mathbf{y} - \mathbf{A}\mathbf{x}\|^2 + \phi_g(\mathbf{x}; \lambda_g) + \phi_e(\mathbf{x}; \lambda_e),$$

where $\phi_g(\mathbf{x}; \lambda_g)$ and $\phi_e(\mathbf{x}; \lambda_e)$ are group and element sparsity promoting terms. The parameters λ_g and λ_e are used to balance group and element sparsity—the larger the lambda value is, the more sparsity is promoted. We have used four variants of the sparse estimator, called HSD, Structured-Soft, Structured-MCP, and Structured-SCAD, see [BMS14] for details.

To evaluate the performance, we plot the normalized mean squared error (NMSE) as a function of SNR for the proposed estimators and compare with a regularized least-squares estimator. NMSE and SNR are defined as

$$\text{NMSE} = \frac{E[\|\hat{\mathbf{x}} - \mathbf{x}\|^2]}{E[\|\mathbf{x}\|^2]}, \quad \text{SNR} = \frac{E[\|\mathbf{y} - \mathbf{z}\|^2]}{E[\|\mathbf{z}\|^2]}.$$

The simulation parameters are found in Table 3-5 and the results are depicted in Figure 3-16.

Table 3-5: Simulation parameters

Parameter	Description
Geometry	50 m wide, 25 m wide strips on each side containing 1000 diffuse scatterers in a length of 1000 m
Number of MD scatterers	10
Number of SD scatterers	10
Carrier frequency	5.8 GHz
Pulse shaping	Root raised cosine with roll-off 0.25
Number of training samples	4096 samples (200 ns)
Bandwidth	20 MHz

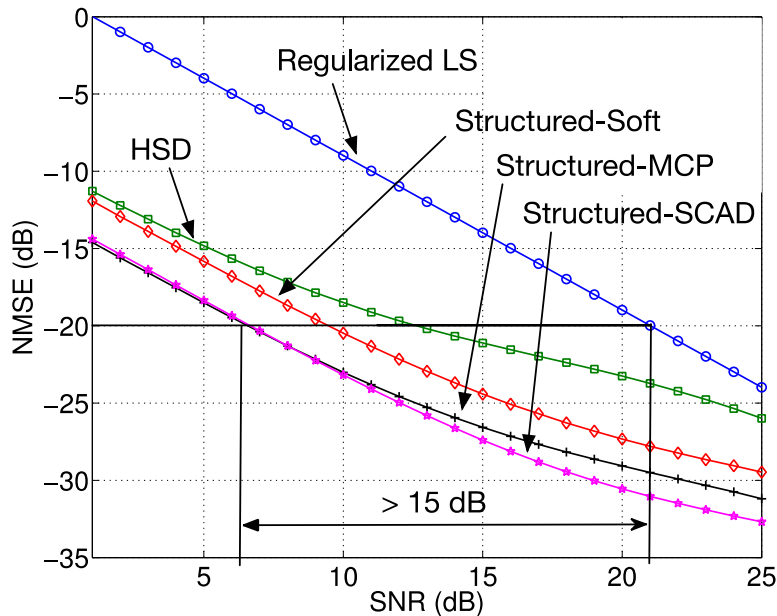


Figure 3-16 Normalized MSE as a function of SNR for proposed estimators compared with the normalized LS estimator. As seen, the performance gain in SNR is approximately 10-15 dB at an NMSE of -20 dB.

As seen from Figure 3-16, significant performance gains are achieved by the proposed algorithms.

Conclusions

Exploiting the delay-Doppler structure found in the V2V channel can yield significant performance gains compared to unstructured approaches, such as least squares. The cost for improved performance is mainly in computational complexity, although no attempt has yet been made to minimize complexity. The approach is currently only suitable for preamble pilot signals. Even though the channel estimation framework requires more work before it can be said to be mature, results are quite promising.

3.6.4 Channel Prediction

The main idea is to use several antennas on a vehicle, placed in the direction of travel. The first of these antennas can be used to predict the radio channel that will later affect the rearward antennas, when they reach that position. This enables various downlink transmit schemes that require accurate channel state information at the transmitter (CSIT) to work also for vehicular users at carrier frequencies above 1 GHz.

Channel state information at transmitters is fundamental in many advanced transmission schemes. However, feedback delays in FDD, framing delays in TDD and transmission control delays of multiple milliseconds result in severe outdatedness of this information for terminals at vehicular velocities. Backhaul delays when using coordinated multipoint transmission (CoMP) increase the problem. Channel prediction can be based on extrapolation of the short-term fading. This has however proven inadequate at vehicular velocities and at high carrier frequencies, above 1 GHz.

We have therefore evaluated a new scheme that was proposed in [SGA+12], which may radically extend the prediction horizon when used on vehicles: Use an additional antenna, a “predictor antenna”, placed in front of the reception antennas in the direction of travel. This

antenna senses the channel that will later be experienced by the other antennas. This approach could enable channel prediction for very large prediction horizons, since the need to extrapolate based on the channel statistics is eliminated. The predictor antenna may be a separate antenna or the first antenna in an antenna array.

The principle can be used to improve downlink transmission that requires CSIT in FDD as well as TDD systems. In [MET14-D33] and in [PSS13] and [PSS15], we demonstrate the benefit of the predictor antenna concept to alleviate beamforming mispointing in massive MIMO backhauling of high speed vehicles, even when local scattering and multipath propagation around the vehicle generate very fast fading. Additional application areas that would benefit from CSIT are CoMP with soft handover and robust backhauling for delay and/or mission critical services to fast moving vehicles.

It is of course crucial for all these applications that the proposed predictor antenna scheme provides channel predictions of high quality. This has been evaluated based on measured data that were collected within the previous EU FP7 ARTIST4G project. The measurement setting and equipment used for data collection is described in Section 6.6.3. We have investigated how different types of antenna designs on vehicles affect the attainable cross-correlation between the channel measured by the forward predictor antenna and the channel that is later experienced by the main antenna, when it has moved to the position previously occupied by the predictor antenna. The attainable precision in the prediction of complex channel gains is directly related to this cross-correlation, see [SGA+12]. As reported in [MET14-D23], the use of monopole antennas placed on flat and uncluttered vehicles roofs has so far provided the highest correlations, resulting in average cross-correlations of 0.97-0.985 for antenna separations of 0.5-3 wavelengths.

In Figure 3-17 we summarize our investigation of the attained normalized mean square prediction error (NMSE) for this case (two monopole antennas on a flat vehicle roof), when predicting the channels for 10 kHz wide OFDM subcarriers at 2.68 GHz, at 45-50 km/h vehicle velocities. The statistics summarizes the variability of the prediction accuracy over 1958 subcarriers, for different settings of the spacing between the prediction antenna and the main antenna. The statistics is over all predicted subcarriers and separate measurements. In Section 6.6.3 we show that the predictor antenna scheme works well also with noisy channel estimates.

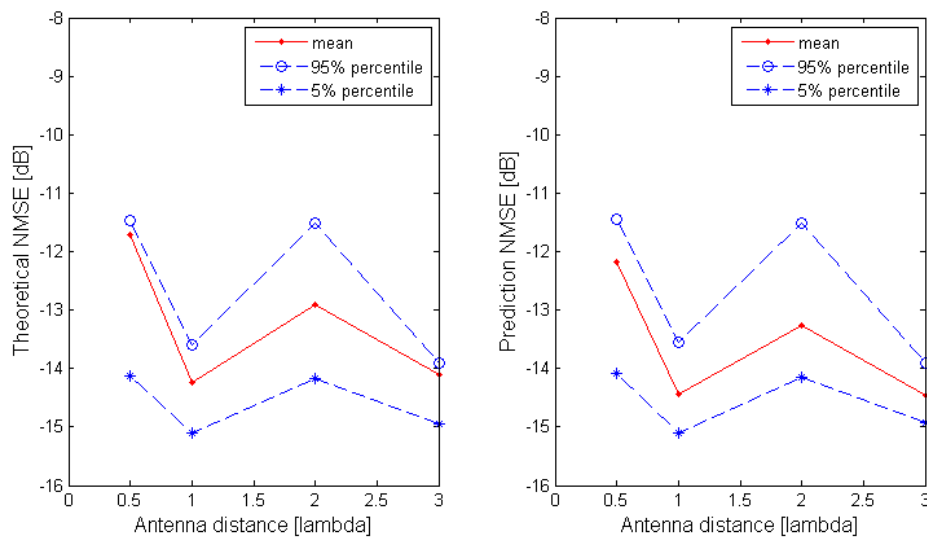


Figure 3-17 Normalized MSE of the predictions of complex channel coefficients as a function of antenna separation, using the predictor antenna scheme – Left: theoretical limits for the NMSE, calculated from the measured correlations between the two antenna signals. Right: corresponding measured prediction performance.

Conclusions

The predictor antenna concept has been shown to be feasible, and to be able to provide accurate channel state information for large prediction horizons in time, that correspond to multiple wavelength distances in space. This concept can be seen as a general enabling technology that would enable the use of all presently known advanced transmit schemes that use channel state information in the transmitter for downlink transmission.

3.6.5 Ad-Hoc MAC for V2V

The TeC allows for V2V communication with strict latency requirements in case of limited or no network connectivity

For safety-critical V2V communication, such as the one described in TC12 [MET13-D11], it is important to allow for communication even when network connectivity is limited or even non-existent. Hence, it is of interest to define an ad-hoc medium access control (MAC) mode that is suitable for V2V communication, meaning that latency and reliability requirements are very strict. We here propose to use coded slotted Aloha (CSA) for this purpose. CSA has been proposed in the METIS for massive machine-type communication (see TeC#12.1.1 – coded random access). However, that version of CSA is not suitable for latency-constrained communication, as explained below, and does not directly consider broadcast-to-broadcast communication. The latter issue is important for V2V, since a vehicle in TC12 wants to share (broadcast) information with all other vehicles in its vicinity.

We therefore modify the conventional CSA setup to address these issues. Firstly, since we are assuming half-duplex transceivers, this implies that a node cannot receive while transmitting. Secondly, we introduce the concept of a frame consisting of N slots into the CSA framework. Each slot is long enough to carry a packet that is assumed to be available at the beginning of the frame and that should be delivered within NT seconds, where T is the slot duration. Hence, given the latency constraint, data rate, and packet size, N is upper-bounded. Performance for CSA increases with N . However, even for modest values of N , CSA significantly outperforms carrier-sense multiple access (CSMA), which is the MAC used by the state-of-the art MAC for V2V communication, namely IEEE 802.11p.

In the N -slot frame, K users transmit multiple identical packets (i.e., with the same payload). At the beginning of the frame, each user randomly selects how many copies of its packet to send and randomly selects the slots in which to transmit. The process is parameterized with the node degree distribution $\Lambda(x) = \sum_{k=1}^N p_k x^k$, where p_k is the probability of transmitting in k slots. The node degree distribution is a critical design parameter. The node degree distribution is precomputed and stored in the nodes before deployment. Hence, the only required node coordination is slot and frame synchronization, which is assumed to be perfect here.

The receiver records and buffers the entire received frame. It is assumed that the receiver can decode a packet in a particular slot without error if, and only if, it is the only packet transmitted in the slot. Each packet contains pointers to the other packets. Hence, if the receiver can decode a packet in a particular slot, it will also know in which other slots the packet is present. It is assumed that the receiver can perfectly cancel the packet from these other slots. After cancellation, new packets might be decodable and the receiver can continue to cancel and decode new packets in an iterative manner. The users whose packets can be decoded are called the resolved users. The overall packet error rate is therefore $P_e = 1 - K_{dec}/K$, where K_{dec} is the number of resolved users.

The simulation results below are for a system with node distribution $\Lambda(x) = 0.86x^3 + 0.14x^8$ and frame duration $NT = 100$ ms. We consider two packet lengths: 200 byte and 400 byte, which implies $N = 317$ and $N = 172$, respectively, using 802.11p protocol overhead and a data rate of 6 Mbit/s (default rate for 802.11p for traffic safety applications). The channel load

is defined as $G = K/N$. As seen in Figure 3-18, CSA can support more than 125% users than CSMA at a packet error rate of 10^{-3} .

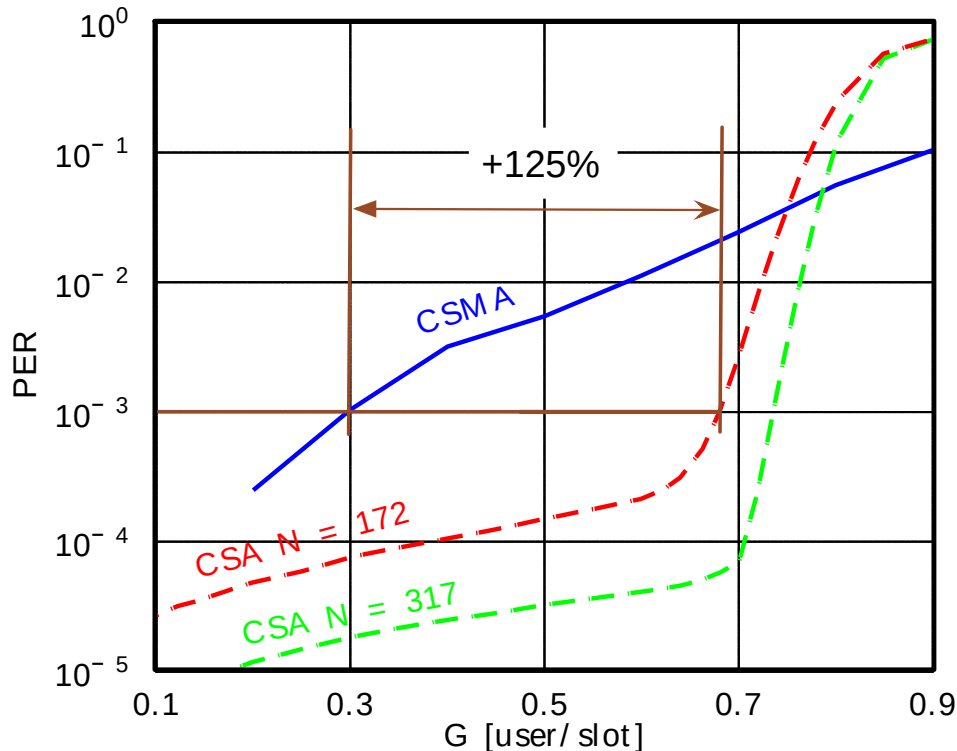


Figure 3-18 Packet error rate versus channel load for CSA with N slots compared to CSMA

In conventional CSA, N can be arbitrarily large, and the throughput (or packet error probability) can be analysed using density evolution. Density evolution is a tool to analyze the behaviour of the interference cancellation process by iterating the probability of a user to be decoded and the probability of a slot to be in collision [Liv11]. It was originally developed for graph-based codes such as LDPC codes [RSU01]. Density evolution relies on the assumption of a cycle free graph, i.e. the asymptotic scenario when the frame length tends to infinity. Density evolution, however, is not accurate for finite frame lengths. We have therefore developed a method for approximating the packet error probability using so-called stopping sets. A stopping set is a set of users that cannot be decoded using the interference cancellation process. The simplest example of a stopping set is two users that transmit their packets in exactly the same slots. In this case, neither user's packet can be decoded. The analysis is quite accurate for interesting channel loads, see [IBG+14]

Conclusions

Coded slotted Aloha with finite frame lengths is an uncoordinated MAC protocol that controls latency and outperforms CSMA with a large margin. Indeed, simulation results show that CSA can support more than twice the number of users as CSMA in some situations. The decoding is based on the ability of the receiver to perform interference cancellation in slots with packet collisions. CSA is still immature, since channel estimation, interference cancellation, and noise and fading impact on decoding performance has not yet been fully characterized. However, since the results so far are quite promising, we consider CSA as a good candidate for future systems.

3.7 Faster than Nyquist (FTN)

FTN signalling offers, in theory, the potential to double the bit rate compared to conventional Nyquist transmission systems.

Faster than Nyquist (FTN) signalling permits in theory to double the useful bit rate without degrading the BER vs. SNR performances and without increasing the bandwidth [RA09]. Naturally, going beyond Nyquist's limit leads to an interference problem. The main features of this problem depend on the modulation type under consideration, i.e., it is a single or a multicarrier transmission. Let us summarize below some key features of the FTN technology.

- The original FTN setting proposed by Mazo in 1975 [Maz75] shows the possibility to transmit $\sin(x)/x$ (sinc) pulses up to 25% faster than Nyquist rate without decreasing the minimum Euclidean distance between symbols in the case of an uncoded system using binary modulation.
- Afterwards still higher throughput gain capability appears when using multicarrier modulations (MCMs).

The MCM case is therefore the most promising but it still requires a better understanding of some theoretical properties.

- Using a waterfilling approach, theoretical limits of the information rate for spectrally shaping OFDM, combined with sinc or Raised Root Cosine (RRC) transmitter filter, can be found [EDK14].
- Using the frame theory, a duality property between oversampled OFDM and FTN-MCM can be exploited to get optimal FTN-MCM systems . that reach the lower bound of interference for any packing factor [SRS+14].

Then, to get practical FTN systems allowing high data rates without any significant degradation in terms of performances, the interferences need to be removed. To this purpose the complexity issue can be handled in different ways.

- For single carrier systems, an FTN multi-rate equalizer can improve the throughput between 30 to 50%, cf. Section 3.10.2.
- In the OFDM case, a maximum likelihood (ML)-based detector can be used when offering in the meantime the possibility of a high degree of parallelization [SW13].
- FTN-FBMC/OQAM can benefit from a fast implementation realization of the corresponding modem. Then, using a turbo-equalizer, together with an EXIT chart analysis, permits an accurate estimation of the FTN limit according to the prototype filter and the constellation size being used [LLS+15]. Up to now, the rate growth that have been obtained are equal to 1.11 for 64-QAM, 1.43 for 16-QAM and to 2 for QPSK.

Conclusions

- From a practical setting, the FTN concept can easily be adapted to conventional MCM systems, as for instance FBMC/OQAM and FMT, making these modulation systems still more flexible. The promise of a throughput multiplied by 2 has been verified by simulation for QPSK [LLS+15]. However there are still some more studies to be carried out :

- Find the boundaries with respect to mobility and delay spread. Without this knowledge FTN can only be envisioned for short range and low mobility scenarios.
 - Compare with conventional MCS strategies.
 - Examine the MIMO extensions of FTN.
 - Look at the PAPR.
- So, for these reasons, FTN cannot just right now be considered as a fully mature technological component.
- However, in future systems FTN can double the bit rate in favourable transmission cases that are associated to two important horizontal topics: D2D and MMC since in this case it can be assumed that the transmission channel is favourable and also that a SISO FTN solution may be preferable to a more energy consuming MIMO solution.

3.8 Filtered and filterbank based multi-carrier

In METIS, two new waveform candidates have been studied that extend classical multi-carrier scheme like OFDM by an integrated filtering component. The filtering enables partitioning a given frequency band into independent sub-bands, which can then be individually configured to optimally fit signal conditions and requirements of individual user links or radio services. Hence, both new waveforms can be considered key enablers for a flexible air interface design, which has been identified as one of the key components for the future 5G systems. While both waveforms pursue the same targets, they use different means to achieve these and thus also differ in their system requirements and implementation aspects: While in UFMC sub-bands being constituted of a minimum number of subcarriers are filtered, which is done to maintain the conventional OFDM signal structure for compatibility issues, FBMC offers enlarged flexibility due to individual filtering of the single subcarriers, which comes with some changes in the signal structure, requiring the redesign of some signal processing procedures. In the following two subsections, the approaches are briefly described and the major advances achieved in METIS are summarized.

Note that the traditional OFDM-based waveforms are proposed for unified air interface design for dense deployments and investigated in section 3.1, aiming at efficient and low-cost implementation.

3.8.1 FBMC based waveform and transceiver design

FBMC/OQAM is a new waveform candidate capable of providing more flexibility to the system design and thus enabling coping with a large variety of different system requirements at the same time.

Filterbank based multicarrier (FBMC) represents a multi-carrier system where the single subcarrier signals are individually filtered with a prototype pulse. Using prototypes with good spectral containment of the signal power allows to partition a given frequency bandwidth into sub-bands that can be individually configured and independently operated without creating mutual interference between the signals in adjacent sub-bands. This feature renders FBMC a key enabler for a flexible air interface, as its application eases spectrum sharing and allows to select the optimum sub-band configuration for a given service according to its requirements and its current channel conditions – without the need to agree on a „best compromise“, as done in today's OFDM based systems. Moreover, by choosing appropriate pulse shapes, FBMC offers a much higher robustness against Doppler and time and frequency synchronization impairments compared to OFDM. The design of the pulse shape thus

represents a novel degree of freedom in the system design, which can be utilized to optimally match the system configuration to desired conditions inherent to a service or the propagation environment. Compared to OFDM, FBMC further offers a much higher spectral efficiency, since it requires less guard bands at the band edges thanks to the good spectral containment of the pulse power, and it does not need any cyclic prefix. Compared to OFDM as it is used in LTE, the spectral efficiency improvement amounts to 13% (see TC6 evaluation in [MET15-D65]). For an illustration of power spectral density and the time domain signal based on a typical prototype filter, see Figure 3-19.

Comparing transceiver complexity of FBMC with OFDM based solutions, latest research results in METIS have shown that the additional complexity required for implementing the subcarrier filtering is only moderate, amounting to a 30% increase at the transmitter and to a factor of two at the receiver [NNB+14].

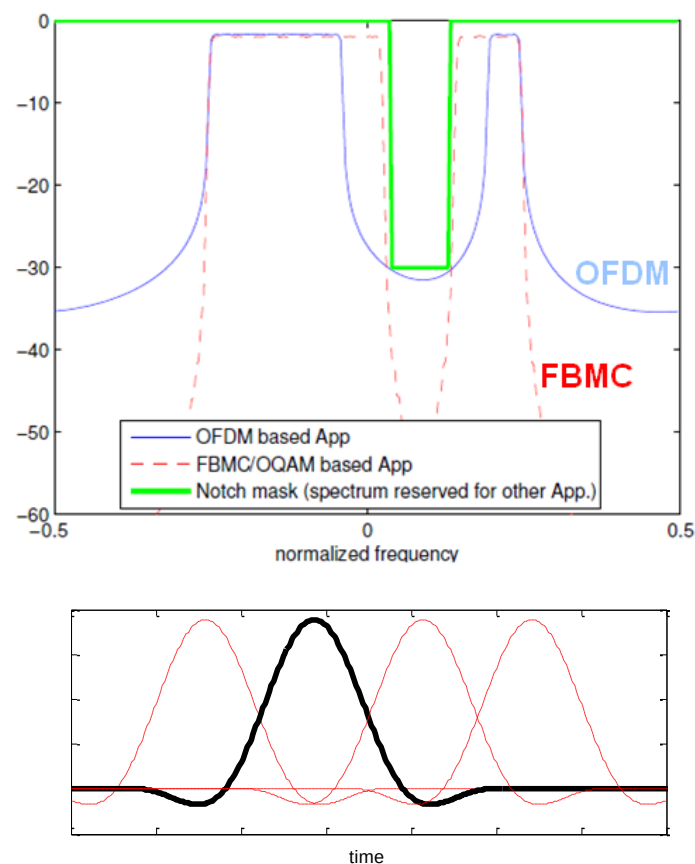


Figure 3-19 Power spectral density of notched FBMC signal in comparison to OFDM (top) and impulse response of the prototype filter in time domain (bottom)

FBMC has been extensively studied in earlier times, but its practical application as an enabling waveform for mobile radio has been less in the focus. In METIS, we therefore investigated the most important aspects of FBMC as an enabler for a flexible air interface design and focused on solutions for practical challenges arising when applying FBMC as the waveform for the future mobile radio system.

Key aspects of FBMC as an enabler for a flexible air interface design

Channel adaptive pulse shaping:

To match the system configuration to given channel conditions, FBMC allows adapting the pulse shape or the subcarrier spacing of a sub-band assigned to a user accordingly. The achievable SIR gains in 2D channels (Delay and Doppler spread) have been evaluated for the

most prominent pulse shape candidates available in the literature as well as for dynamic subcarrier spacing, yielding gains of 3-4 dB by pulse shaping and 6 dB by changing the subcarrier spacing by a factor of two. These gains translate to a higher robustness of the transmit signals against distortions from Doppler and delay spread and correspondingly higher throughput. For proper isolation of the signal to the adjacent sub-bands (which may have a different configuration), a single subcarrier guard has proven to be sufficient. A summary of the research is given in Annex A, section 6.8.1.1.

Synchronization robustness:

1. An analytical derivation of the performance degradation due to synchronization error has been conducted in [LGS14]. This analysis shows the influence of the prototype in different asynchronous cases, revealing that filters with good frequency localization are more suitable for time synchronization error and filters with good time localization are more suitable for CFO distortions.
2. Thanks to the spectral containment of the used pulse shape, FBMC enables asynchronous transmission of different users operating in adjacent frequency bands. Investigations have shown that an interference isolation of more than 60 dB can be achieved in time asynchronous systems, given that a single subcarrier is used as guard band between the adjacent bands. Details are given in Annex A, section 6.8.1.2. A timing advance procedure to align multi-user signals (as known from LTE) is then no longer necessary.

Prototype filter (PF) design:

Most often PFs have to satisfy either a frequency selectivity or a time-frequency localization (TFL) criterion. In the case of short prototype filters, i.e. with length equal to the expansion/decimation factor, analytical expressions have been computed leading to nearly optimal FMT systems [DP11]. The method has been extended in [DP13] to FBMC/OQAM systems considering the TFL criterion. The key finding is the derivation of an analytical expression for short (i.e. with length equal to the number of subcarriers) PFs, providing an optimal time-frequency localization.

Solutions for practical challenges

Filter tails in short package transmission:

The long filter tails have been seen a drawback of FBMC when signal framing for short package transmission is considered, as the tails may then amount to large overhead. This problem can be solved for short prototype filters [DS10]. More generally speaking, in METIS, two solutions based on circular convolution and additional windowing have been developed to overcome this problem: 1) WCP-COQAM provides an advanced version of FBMC. Besides solving the tail issue detailed above, it also solves other key issues that have been identified for FBMC, namely the weak delay spread immunity issue, and MIMO Alamouti issue [LS14ab], [MET14-D23]. 2) Weighted circularly convolved FBMC has been introduced in [MET14-D23] and with further details in [AJM13].

Channel estimation:

1. Channel estimation schemes proposed for FBMC exhibit strong performance degradation in long delay spread channels. To overcome this problem, a novel pilot design and estimation scheme has been developed, enabling improved estimates at the same pilot overhead. It could be shown that with the novel estimation scheme, FBMC can achieve the same coded BER performance as OFDM in typical LTE scenarios [ZVS14].
2. The channel estimation scheme based on auxiliary pilots, which is used in FBMC as the standard channel estimator, exhibits the problem of high precoding symbol transmit power. To overcome this problem, a novel scheme relaxes the constraint that the interference has

to be driven to zero by the auxiliary pilot, but rather to a specific non-zero value out of a predefined set, which is chosen to minimize the transmit power [BWK15].

Equalization:

ERIP Equalization method for FBMC/OQAM: this method takes advantage of the imaginary part of the interference that usually is got rid of directly to improve the equalization quality. The principle is that there exists some correlation between the real and imaginary interference. Thus, as long as the imaginary interference is known, a simple projection can be operated to predict the real interference [NLS12].

MIMO

1. It is known in the literature that FBMC-MIMO achieves the same performance as OFDM-MIMO if linear MMSE equalizer is used. In the METIS project, it has been shown by simulations that all LTE MIMO transmission modes using linear transceivers (precoders and equalizers) can be reused for FBMC without performance loss [ZWS15].
2. In practical use cases, where two transmission blocks in frequency domain may be precoded with different channel parameters, severe inter-block interference will arise due to the real-field orthogonality property of FBMC. The way proposed to avoid this inter-block interference is to establish the complex field orthogonality between the consecutive blocks so that they are immune to the phase drifts of the precoders. In METIS, different methods have been developed to efficiently establish the required orthogonality. One method is to use complex modulated symbols instead of real-valued ones for the boundary subcarrier and cancel the intra-block interference by precoder and receiver design. Other is to use complex valued prototype filter instead of real valued one on the boundary. Details of the solution can be found in [ZG15].
3. The bit error rate (BER) performance of a MIMO-FBMC system is analyzed comprehensively and it is shown that the system can achieve low error performance [SRL14b] comparable to MIMO-OFDM. The system model and further details are given in [MET14-D23]. The punctured Tomlinson Harashima precoding (THP) technique is proposed which shows feasibility of FBMC. The punctured ZF also seems promising. The approach is to remove the error floor, at a slight loss of throughput, by not using one or more antennas for some symbols in some subcarriers. The error floor can be removed, which is quite a significant result. A summary of results is given in Annex A, section 6.8.1.3.

RF imperfections

In any wireless communication device, analogue frontends inevitably suffer from imperfect behaviour due to fabrication inaccuracies, power/area constraints, aging or change in operating conditions, etc. They result in signal distortion and effect signal reconstructability at RX, but these anomalies have unique impact on signal model and, if their characterization is known for a given waveform, they can be effectively and efficiently suppressed. In this study, our focus was the detailed analysis of FBMC/OQAM in the presence of I/Q imbalance of zero-IF quadrature receiver, oscillator phase noise and PA non-linearity.

- In case of I/Q imbalance, we found that FBMC/OQAM suffers from slightly higher ICI/ISI than OFDM/QAM, but when I/Q imbalance parameters were estimated and corrected, there was no potential loss in terms of BER performance [Ish13]. We further extended the compensation scheme to a more generalized I/Q imbalance model i.e., frequency-selective I/Q imbalance in [Ish14], where we found that a combination of the pairwise equalizer and iterative interference cancellation can effectively alleviate the loss of orthogonality caused by the channel frequency-selectivity and I/Q imbalance (Annex A, section 6.8.1.5).

- In [Ish15], we present the performance analysis in the presence of phase noise, where it is shown that FBMC/OQAM and OFDM/QAM are expected not to differ much in terms of SIR and SER evaluation (Annex A, section 6.8.1.6).
- As far as nonlinear PA is concerned, we conducted both real hardware measurements and numerical simulation. The hardware measurements were made on a commercial available Radio-Head-Unit, where PSD steep roll-up advantages of FBMC in comparison to OFDM is observed. The numerical simulation was based on a published PA design. Our extensive simulation results indicate that while FBMC/OQAM once again equals PAPR and EVM of an OFDM/QAM system, its ACLR performance is better than OFDM/QAM and allows up to 1 dBm more average output power while maintaining the same leakage level. Furthermore, beneficial for opportunistic access applications, FBMC/OQAM has the higher potential to coexist with other networks due to sharper band-edge roll-off as indicated by its PSD measurements (see Annex A, section 6.8.1.4).

In conclusion, the characterization of the frontend imperfections has been found to deviate from the well-known OFDM/QAM structure in that FBMC/OQAM creates a complex pattern of ISI and ICI. Nevertheless, our detailed analysis indicated that the sensitivity to the considered imperfections was close to OFDM and their suppression with the presented compensation methods results in a little to no performance gap w.r.t the state-of-the-art.

Conclusions

FBMC can be considered a key enabler for a flexible air interface design, as it facilitates spectrum sharing of a multitude of different radio services with high efficiency and enables the system to be configured according to the individual needs of each service without the need of a compromise, as prevalent in today's systems. Several practical challenges have been identified and suitable solutions have been devised, rendering the technology mature for the application in practice.

3.8.2 Universal Filtered Multi-Carrier (UFMC)

This TeC proposes an alternative waveform approach (evolving OFDM) being better suited to meet the wide range of requirements being anticipated for 5G.

UFMC (aka Universal Filtered (UF) OFDM) is a modification of the well-known 4G waveform CP-OFDM. Instead of applying a cyclic prefix (CP) it applies a per sub-band filtering (though a CP could still be applied, it is not used for the investigations within METIS). By doing so, separation of single sub-bands in frequency domain is improved. This can be used to tune each sub-band independently according to the link characteristics (both related to channel, service type and device class being served). Power spectral density for an exemplary setting of UFMC (12 subcarriers per sub-band, filter length $L=80$, sidelobe attenuation 60 dB) and the time domain UFMC symbols are illustrated in Figure 3-20.

With UF-OFDM high-end devices (e.g. smart phones) can be served with highest spectral efficiency, while low-end devices (e.g. sensors/actors) can be served with highest energy efficiency. For example with applying UF-OFDM protocols used for network entry can be simplified (i.e. no closed-loop synchronization) to reduce on-time of the devices (reducing energy consumption), allow for cheaper devices (oscillator requirements may be relaxed) and reduce control overhead (less messages to be exchanged between device and base station required). In [SW14] it has been shown, that interference can be significantly reduced with devices being only loosely synchronized. For the settings given in the paper (following LTE numerologies) interference could be reduced by a factor of 10. With optimized settings further improvements are possible.

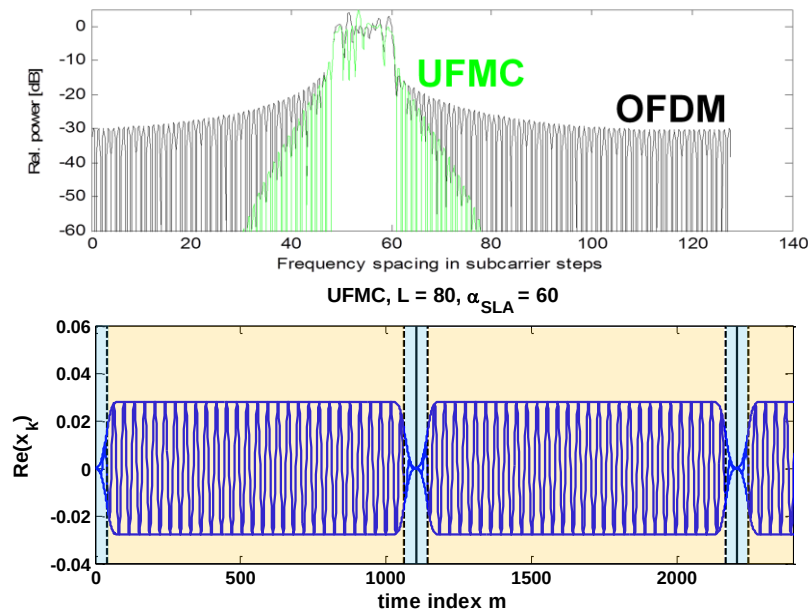


Figure 3-20 Power spectral density of UPMC signal in comparison to OFDM (top) and UPMC symbols in time domain (bottom)

Another means to make use of the sub-band separation is the use of different sub-carrier spacing within a single TTI. A system supporting this can make use of the following advantages:

- Wider spacing translate into shorter symbols. With a given pilot-pattern the absolute time between pilots gets smaller. This way channel estimation for users with high velocities can be improved significantly (see Annex A, section 6.8.2.1) as the pilot pattern better matches to the coherence time of the channel.
- Shorter symbols are a first step towards smaller air interface latencies, as for a given number of multi-carrier symbols per TTI the absolute length of the TTI in seconds shrinks accordingly.
- With allowing for wider sub-carrier spacing, reduced PAPR modes are available. A low-end device transmitting data within a given sub-band of width 180 kHz faces a better PAPR if it is allowed to use 6 sub-carriers with 30 kHz each than if it has to use 12 sub-carriers with 15 kHz each. Naturally, the channel delay spread has to be taken into account, here, as with increasing spacing the single symbols are becoming shorter.

With UF-OFDM the time domain overhead scales with the number of time-domain symbols per burst. In [SWC14] this feature has been investigated. Gains compared to CP-OFDM and a basic variant of FBMC (SMT) has been given. In relation to CP-OFDM the gains are in the order of 10%. Compared to FBMC (PHYDYAS variant, but with burst truncation) gains depend on the size of the burst. With very short bursts (e.g. up to 8 multi-carrier symbols) gains are in the range of 10-25% (depending on the truncation factor). For extremely short structures (e.g. UL sounding symbol) the gain is between 28-80% (the latter for no truncation).

The applicability to and potential gains of UF-OFDM in an UL CoMP setting has been shown in [VWS13]. In a rather simple setting UPMC proved a superior behaviour in terms of symbol error rates (up to 5 dB gain) compared to CP-OFDM. Though, future work (after METIS) will push this into a less abstract setting.

An overall design proposal (frame structure, synchronization mechanisms, waveform design options, multiple access scheme) has been proposed in [WSC14].

A common first impression is to assume UF-OFDM to lose orthogonality between subcarriers in back-2-back mode. Though, this holds only for specific receiver types (following the matched filter principle). If applying the FFT-based receiver as presented in [MET14-D23] orthogonality between subcarriers is maintained (for prove, see Annex A, section 6.8.2.2).

As UF-OFDM in its version being discussed in METIS does not apply a CP, orthogonality is lost in case of channels having non-zero delay spread. However, with reasonable settings (e.g. 15 kHz spacing) and for relevant channel profiles (e.g. eVEHA) the arising inter-symbol and inter-carrier interference is low enough to be negligible e.g. against noise for reasonable SNR working points [WWS15]. With channels having a more extreme delay spread characteristic, the use of zero-tail DFT spreading similar to [BTS+13] is advocated. Alternatively, parameterization of the waveform might be adjusted accordingly.

Conclusions

Overall UF-OFDM promises to be a signal format being capable to meet the broad requirement space being anticipated for 5G. E.g. a system applying UF-OFDM is able to be adapted on a per sub-band base to any means like reduced energy consumption, higher supported velocities, lower air interface latencies. This way a system can scale according to its current needs (i.e. it can serve any mix of low-end/high-end, low speed/high speed devices without waste of resources as it would be the case with CP-OFDM, as here adaptation is only possible on symbol level instead of on sub-band level). Instead of compromising between different needs (as e.g. done in LTE) the system is able to adapt to different needs improving the quality of user connections.

A system applying UF-OFDM is able to implement any advanced scheme being available for CP-OFDM with lowest effort (e.g. any kind of multi-antenna procedures). As an example recent results have shown, that for a realistic channel (VEHA) the resulting distortions with not using a cyclic prefix (with LTE settings) is below -42 dB and thus not relevant compared to other distortions in cellular networks (to be published).

So far investigations have concentrated on link-level. Future work will assess performance gains of system level. This work will be conducted after METIS.

3.9 Modulation & coding and new channel coding concepts

3.9.1 Constrained envelope coded modulation

In [MET13-D22] we identified that constrained envelope coded modulation would be beneficial to support energy efficient, robust and low cost transceivers, which are important aspects in selected METIS test cases. To this end, in [MET14-D23] we introduced a precoded Single Carrier Frequency Division Multiple Access (SC-FDMA) scheme. This scheme is compatible with Orthogonal Frequency Division Multiplexing (OFDM) transceivers, and still controls the envelope variations, which is beneficial for efficient and low cost non-linear amplifiers. Due to dispersed subcarrier mapping it is also robust to frequency selective fading and interference. The precoding step is based on Continuous Phase Modulation (CPM), and the scheme is denoted CPM-SC-FDMA. The properties and performance of the scheme were described in [MET14-D23] section 2.9.1.

Conclusions

As summarized in [MET14-D23], CPM-SC-FDMA can outperform convolutionally encoded SC-FDMA with a gain of up to 4 dB in end-to-end power efficiency, under the same requirements on bandwidth, information rate and complexity. The scheme has been

elaborated on within the MMC horizontal topic and also within the UDN horizontal topic for high bandwidth and coverage limited mmW scenarios, and is seen as a promising solution for the corresponding test cases.

3.9.2 Advanced coding and decoding

3.9.2.1 Adaptive complexity flexible base band

For this TeC, the previous deliverable D2.3 [MET14-D23] provides detailed technical description and corresponding results. Therefore, in this document we report only a short summary related to this research activity where original complexity adaptive receivers combining multiple iterative processes have been explored and proposed. The main focus was on the digital baseband implementation, and more specifically on channel coding and MIMO techniques. In fact, the multiplication of requirements has been leading the parameters of wireless communication systems to increase drastically. Moreover, each of these parameters has a wide range of mandatory and/or optional values: coding schemes (with various frame sizes and many code rates), modulations (with several types of constellation), multi-antenna configurations, multiple interleaving rules, etc. In this context, many recent efforts have been conducted towards the design of flexible digital baseband processing [MBK+12]. Most of the efforts at this level target the proposal of efficient computational and storage resource sharing between the different supported modes. Few others have proposed to switch between different algorithms depending on the system parameters, user requirements, and/or the transmission environment conditions.

Furthermore, several recent efforts have considered the iterative processing which is widely adopted nowadays in modern wireless receivers for the decoding of advanced channel codes. The extension of this principle with an additional iterative feedback loop to the MIMO equalizer and to the demapper blocks has proven to provide excellent error rate performances in severe channel conditions (erasure, multipath, real fading channel models). However, the adoption of iterative processing with turbo decoding is constrained by the additional implied implementation complexity, heavily impacting throughput, latency and power consumption. In this context, most state of the art works have considered mainly the error rate performance aspects of the iterative processing [DJB+95, AD06, KW07, GMB10]. Other works have studied the optimization of individual components [ASL08][LBL05] without deep investigation of potential optimization techniques from a system-level point of view. Further works have presented different optimization techniques for limited set of parameters [KW07]. Finally, by considering various communication techniques and system parameters, novel system-level optimizations have been proposed in [HBJ12a, HBJ12b, HBJ12c] based on new iteration scheduling of inner and outer feedback loops. These previous results illustrate how for certain configurations additional feedback to the SISO demapper and equalizer can reduce the overall receiver complexity in terms of computations and write memory accesses along with lower error rate performances. The extension of this study to include multiple antennas schemes has been performed in the context of METIS. Several low-complexity detection algorithms have been investigated in terms of measured complexity for target performance with and without iterative processing at the receiver side. Obtained results confirm that for several channel conditions and system parameters, iterative processing can achieve better complexity-performance trade-offs. This constitutes a very interesting result as it demonstrates the opposite of what is commonly assumed.

Conclusions

All of the above in mind, the conducted research work within METIS illustrated how iterative MIMO detection offers significant improvement in error-rate performance for a reduced SNR at the cost of additional baseband complexity. Furthermore, the conducted analyses with various system parameters and channel conditions have shown that good trade-offs between

performance and complexity can be achieved. It goes farther than the choice of the reduced complexity algorithm to the large number of receiver parameters. In conclusion, the conducted analyses and obtained results on possible complexity-performance trade-offs can be exploited to design multi-mode adaptive-complexity MIMO detector supporting both MMSE and LSD with variable list size. In such a flexible detector, selection between the executed algorithm and choice of the main parameters should be based on channel conditions and targeted performance.

3.9.2.2 Practical lattice codes

The research in this TeC was addressing the following question: Are lattice codes a possible alternative to the state of the art in coding-and-modulation?

Lattice codes have been proved to achieve the capacity of the AWGN channel [EZ04], and have been shown to solve various communication theory issues [ELZ05]. However, most of the results are theoretical. When it comes to implementation, some issues have to be addressed before promoting lattice codes as an alternative to other modulation and coding schemes.

One of the implementation issues to be solved is the shaping of lattice code encoding. A lattice code Λ_B can be defined as the intersection between an n -dimensional lattice Λ and a compact bounding region B of $\mathbb{R}^n$ called the shaping region. The codewords are all the lattice points that belong to the shaping region B . Depending on the shape of B , the error rate performance may vary. The gain related exclusively to the shape of B is called the shaping gain and is upper bounded by 1.53 dB [For92], which is reached when B is a hypersphere. Unfortunately, hypersphere shaping is too complex to implement. If B is a hypercube, the shaping gain is 0 dB. Note that the shaping gain can even be negative, if the shape of B is worse than a hypercube. A survey of popular shaping techniques was proposed in [SFS09] and applied to low-density lattice codes to improve the shaping gain. Among them, the best shaping gain is achieved by the so-called nested shaping.

The advantage of lattice encoding is the linear link between the information bits and the resulting codewords in the signal domain which can be exploited for example at the receiver side.

Compared to uncoded QAM, sphere decoding of shaped lattice codes has been improved in order to reach the shaping gain, even for low spectral efficiencies. Moreover, the proposed algorithm [AG15] performs also well on the Rayleigh channel, even if the shaping process is not of utmost importance, unlike for Gaussian channels.

Varying spectral efficiencies is straightforward since only one parameter variation is required instead of the couple (code rate, modulation size). Moreover, the spectral efficiency can vary with tiny increments.

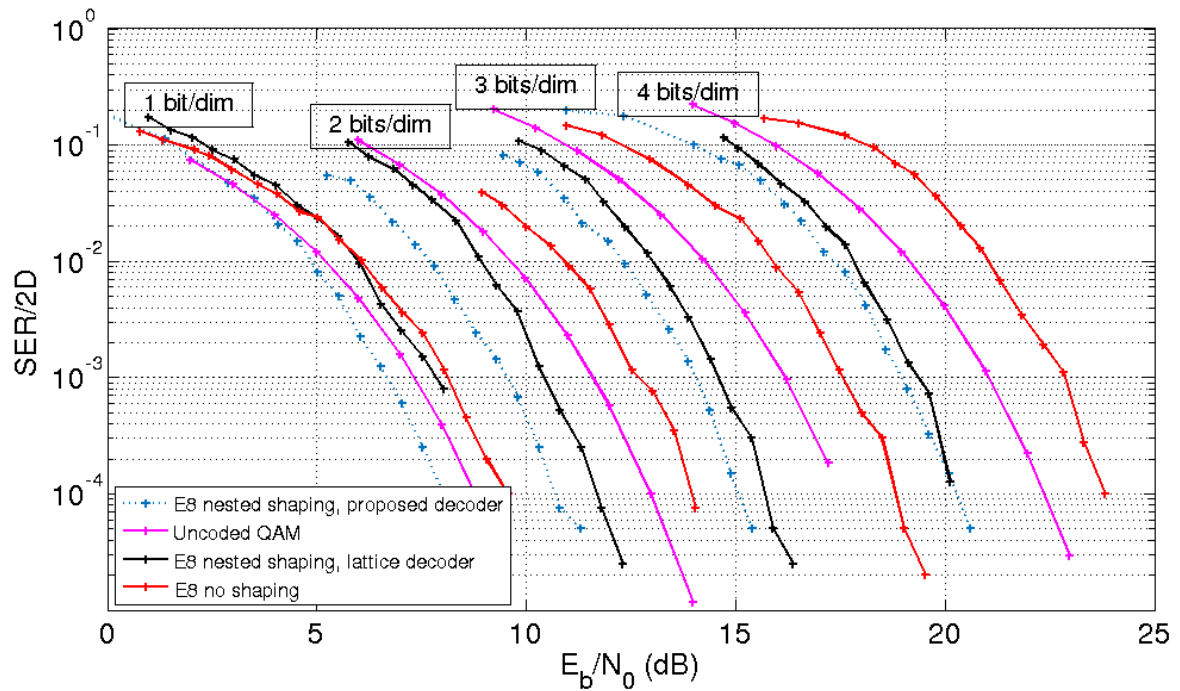


Figure 3-21 error rate comparison between uncoded QAM modulation symbols and lattice E8 based modulation symbols and the influence of shaping in the AWGN channel.

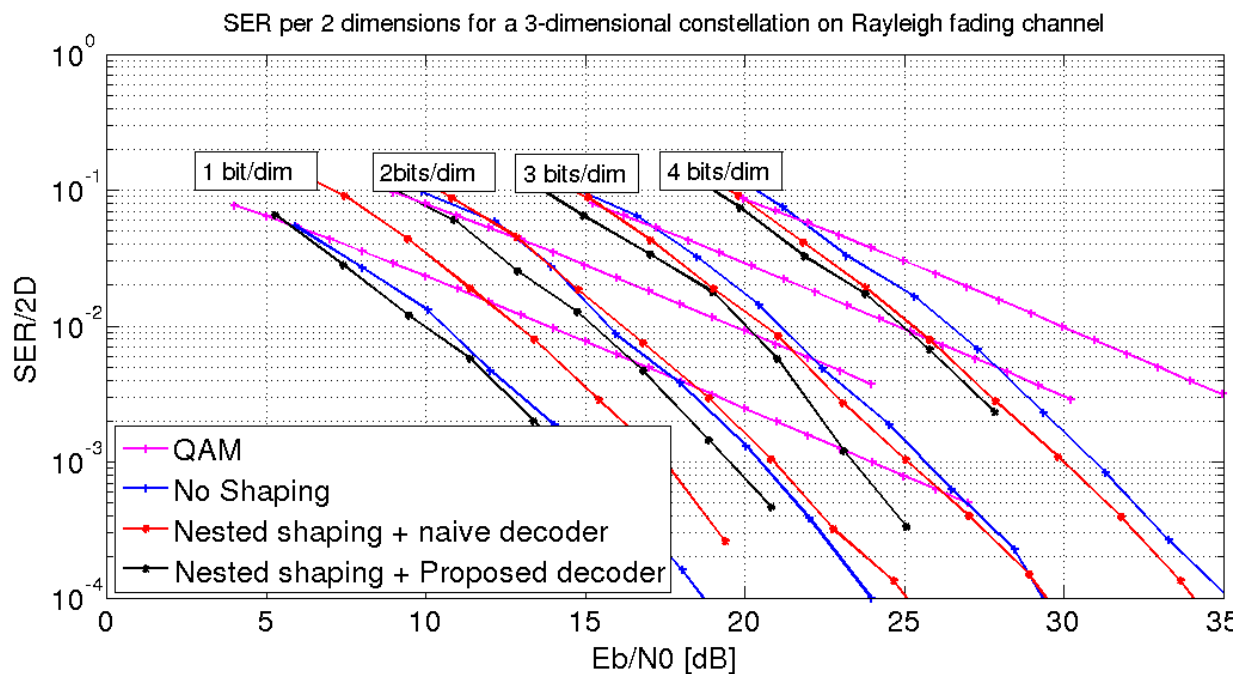


Figure 3-22 error rate comparison between uncoded QAM modulation symbols and 3-dimensional lattice codes designed for uncorrelated Rayleigh fast fading channel.

Conclusions

Implementing lattice codes with an ad-hoc shaping on the Gaussian channel offer their promised coding and shaping gain for any spectral efficiencies provided that the proposed sphere decoder based algorithm is implemented.

On the Rayleigh fading channel, no shaping is required, but in case shaping is still employed, then the proposed algorithm achieves the same performance and diversity as unshaped transmissions.

However, practical implementations with lattice codes as an alternative to classical coding and modulation schemes still require for some more investigations, especially for longer codes.

3.10 Advanced transceiver design

3.10.1 Full Duplex communications

Full duplex D2D underlay communication system.

We investigate the feasibility of full duplex (FD) communications, in underlay D2D networks [HRL14]. We study the sum ergodic rate of an underlay D2D network when D2D users operate in FD mode and compare the performance to a half duplex (HD) underlay D2D network with equivalent total energy and radio frequency (RF) hardware complexity. We consider a D2D link (2 users) in an underlay configuration within a single cell. Figure 3-23 shows the throughput of the system when only one cellular user is sharing resources with D2D. Throughput of the system with full duplex (FD) D2D will increase as the amount of self interference (SI) cancellation increases, it can be seen that for less than 78 dB of SI cancellation half duplex (HD) D2D has better performance due to large residual of SI. However as amount of SI cancellation increases, residual of SI will be small and SINR of D2D receivers will increase and FD will outperform HD. At 110 dB cancellation, FD D2D link has almost double throughput of HD D2D.

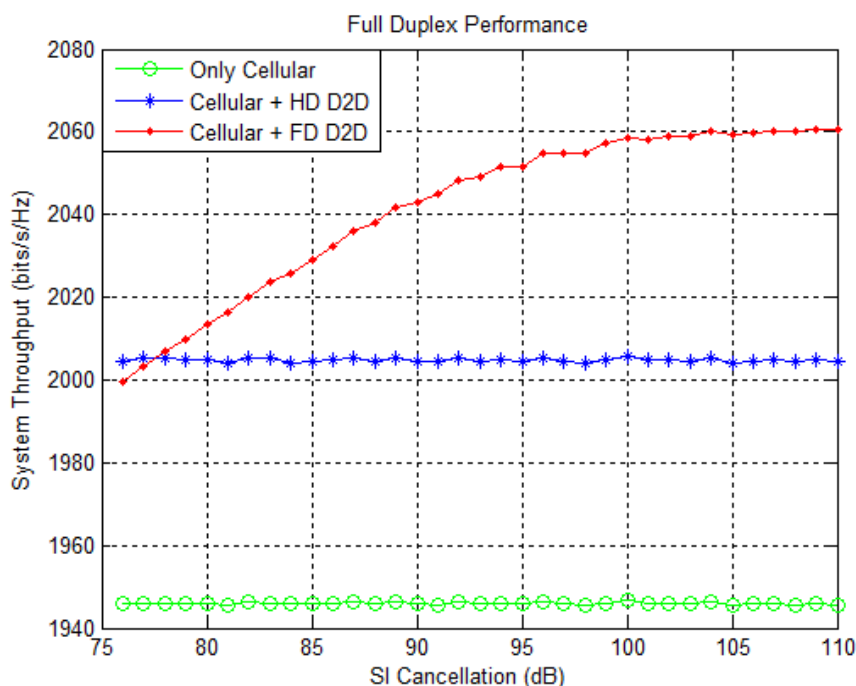


Figure 3-23 Throughput of System for one user resource sharing for minimum SNR at BS of 10 dB

Conclusions

Transmission and reception at the same time on the same frequency band, potential to double the spectrum efficiency. In the case of same bandwidth, double rate will be achieved. This however requires that sufficient self interference cancellation be employed. For the same rate, half spectrum is required. Mature enough for small transmit power systems such as WiFi, D2D, small cells. Naturally more investigations on RF implementation and complexity issues are warranted.

3.10.2 Multi-rate equalizer for single-carrier communications

The multi-rate equalizer (MRE) provides a multi-rate signal processing solution to inter-symbol interference (ISI) observed in single carrier communications and in faster-than-Nyquist (FTN) signalling.

The main goal of this TeC cluster is to develop an equalizer which is based on successive interference cancellation (SIC) and employs multi-rate signal processing tools. The idea can be seen as the ISI analogous to the Bell Labs Layered Space Time (BLAST) system. BLAST deals with the interference caused due to the spatial interference whereas MRE deals with ISI. MRE can achieve the independent identically distributed (i.i.d.) input capacity (C_{iid}) of the channel. Furthermore, MRE can be implemented in frequency domain. Thus, it enjoys a complexity in the order of a linear equalizer.

Although the formation of layers is rather trivial in a system dealing with spatial interference as in BLAST, formation of layers is not obvious for an equalizer dealing with ISI. We pick the layers as the size- K polyphase components of the transmitted signal, where K is the number of layers. However, we sort the layers in bit-reversed ordering rather than the usual ordering.

The bit reversed ordering leads to the efficient multi-rate signal processing architecture shown in Figure 3-24 for two layers. The transfer functions of the filters shown in this figure are specified in [BJ13]. This signal processing structure essentially decomposes the original ISI channel into two virtual channels between pairs $(x_0[n], y_0[n])$ and $(x_1[n], y_1[n])$. This architecture can be recursively extended to other number of layers which must be powers of 2. As number of layers increases these virtual channels tend to become memoryless. If the number of layers is sufficiently high the equalizer blocks shown in Figure 3-24 becomes unnecessary. Notice that interference cancellation takes place after decoding in Figure 3-24. Therefore, MRE does not suffer from error propagation.

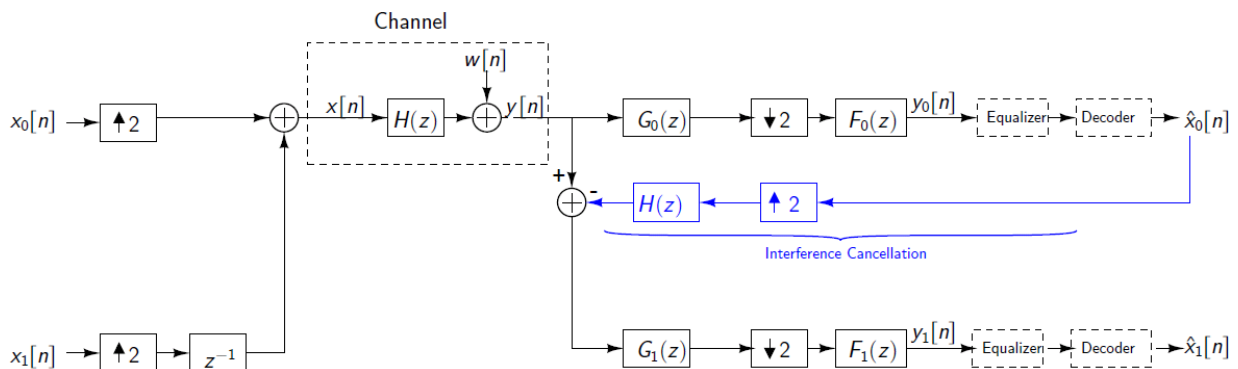


Figure 3-24 Block diagram of MRE transmitter and receiver

Thanks to the error-propagation-free SIC, higher layers enjoy higher signal-to-interference-plus-noise ratio (SINR) than lower layers. Consequently, higher layers can be used to transmit higher throughput. Figure 3-25 presents average throughput of MRE with different number of layers in Pedestrian Channel B settings. It can be observed from figure that the throughput is equal to that of the minimum mean square error (MMSE) equalizer for a single layer. As

number of layers increases throughput increases until it reaches C_{iid} . This observation can also be proven analytically. The lowermost black curve in Figure 3-25 indicates the throughput achievable by the MMSE equalizer in DFT precoded OFDM transmission specified in LTE. MRE provides 20% throughput increase when compared to this baseline.

Obviously every filter shown in Figure 3-24 can be implemented in frequency domain. Furthermore, implementation of this block diagram is possible even if a cyclic-prefix is not transmitted by moving the whitening filters, $F_i(z)$'s, to the front of downsamplers by using the multirate identities. This operation ensures that the resulting filter $G_i(z)F_i(z^2)$ has a quite short impulse response, thus, it allows implementation via overlap-save or overlap-add techniques. Getting rid of cyclic-prefix provides a further throughput improvement of 25%.

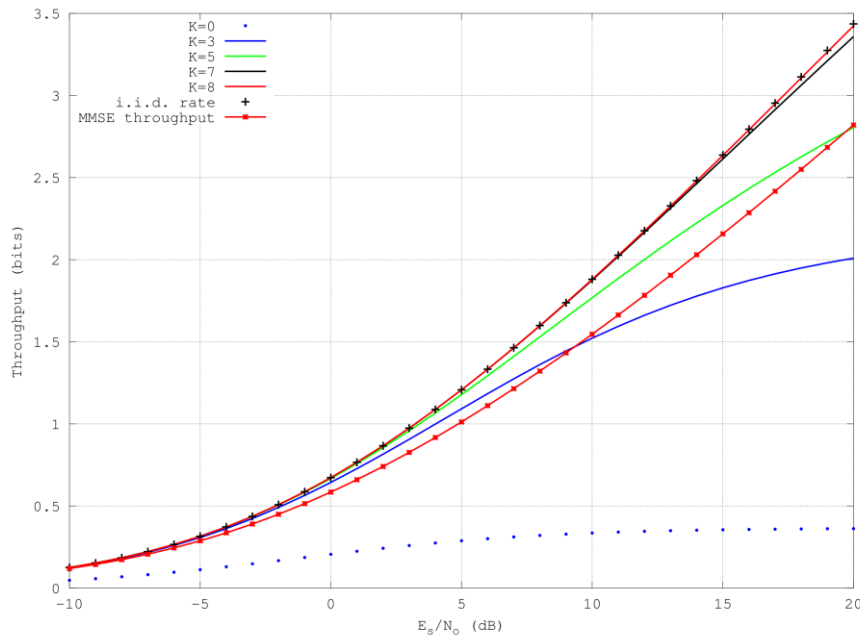


Figure 3-25 Average throughput achieved by MRE compared to the C iid in Pedestrian Channel B settings where M denotes the logarithm of number of layers in base 2.

MRE Inspired Faster Than Nyquist Signalling

Another scenario which experiences ISI is FTN signalling explained in section 3.7. FTN promises an increase in throughput when maximum likelihood sequence detection (MLSD) or marginal a posteriori probability (MAP) algorithms are employed at the receiver. However, the computational complexity of these algorithms prevents them to be employed in practice. The practically feasible equalization algorithms such as minimum mean square error (MMSE) equalization leads to a decrease in overall throughput when employed to equalize ISI occurring due to FTN. MRE equalizer could prove useful for equalizing ISI observed in FTN scenarios.

An FTN transceiver based on MRE is shown in Figure 3-26 and Figure 3-27. At the transmitter side the data stream is separated into two parallel streams. Each of these streams are encoded independently with two encoders of rates R_0 and R_1 . Then both of these streams are modulated and pulse shaped with pulse shaping filter $H(f)$ which is of Nyquist rate T . Hence, both of these streams are actually Nyquist rate streams. FTN transmission is achieved by superimposing these two streams after delaying the second stream by $T/2$. Hence, the rate of the overall transmission is two times the Nyquist rate. However, the transmitted signal still occupies the same bandwidth.

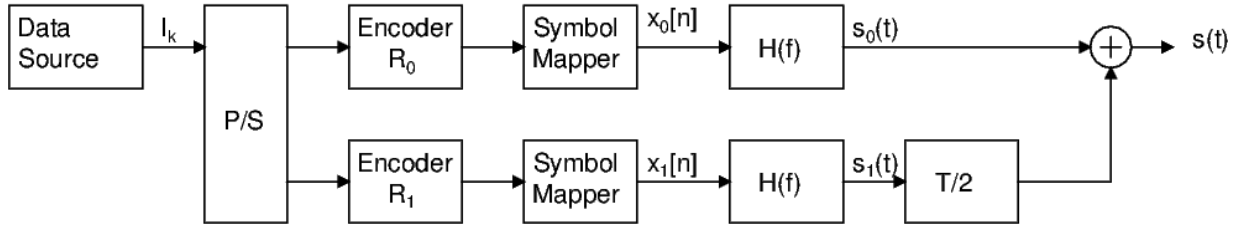


Figure 3-26 FTN transmitter inspired by MRE

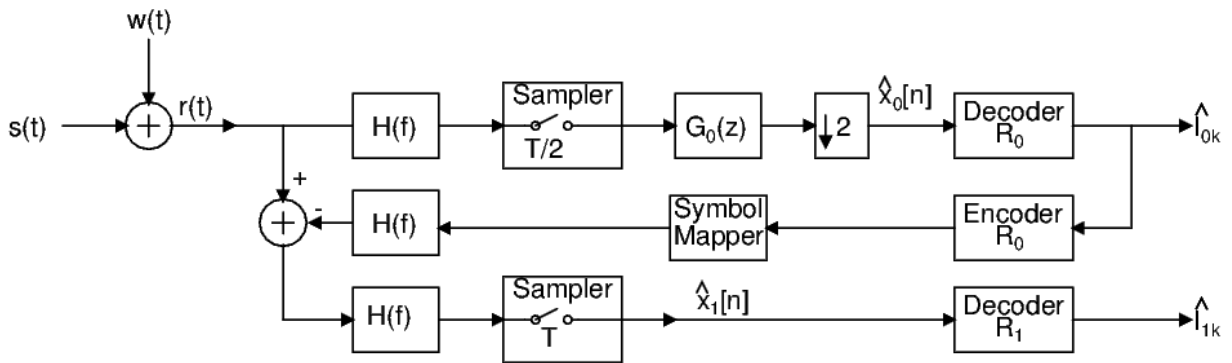


Figure 3-27 FTN receiver associated with the transmitter shown above based on MRE

Since both of $x_0[n]$ and $x_1[n]$ are Nyquist rate signals, there will be no intra-stream interference. However, the ISI will be observed in the form of inter-stream interference. The receiver aims at handling this inter-stream interference via successive interference cancellation. The filter $G_0(z)$ is optimized so that it suppresses $x_1[n]$ as much as possible while preserving $x_0[n]$. Then the interference due to $x_0[n]$ is cancelled after decoding. Provided that $x_0[n]$ is correctly decoded, the channel observed by $x_1[n]$ is a perfect Nyquist rate channel and is not contaminated by any sort of ISI at all. Hence, R_1 can be as high as in the Nyquist rate. The rate R_0 is the extra rate provided by employing FTN and it should be chosen so that correct decoding is ensured. Since $G_0(z)$ cannot suppress the interference completely due to $x_1[n]$, R_0 has to be less than R_1 . The empirical results showing the performance if this proposed system is shown in Annex A, section 6.10.2. These results indicate that MRE inspired FTN can provide up to 30% increase in throughput in the SNR region of interest.

Conclusions

MRE improves the achievable throughput of the SC systems by 20% due to better equalization performance and a further 25% due to getting rid of cyclic-prefix. These throughput improvements can potentially make single carrier systems a serious contender for multi-carrier systems. MRE achieves this throughput improvement with a low complexity algorithm.

3.11 Multiple Access

3.11.1 Non- and quasi-orthogonal multiple access allowing spectrum overload

3.11.1.1 NOMA

The main idea of NOMA is to exploit the power-domain in order to multiplex multiple users and rely on advanced receiver such as successive interference canceller (SIC) to separate multiplexed users at the receiver side. In our work, we also enable the combination of Non-Orthogonal Multiple Access (NOMA) with single user MIMO (SU-MIMO) transmission, where the transmit signals intended for multiple users are multiplexed in the power-domain and for each single user multiple beams (SU-MIMO transmission) are multiplexed in the spatial domain [BL14]. As a result, for 2-UE multiplexing case, using only 2 transmit antennas we are able to perform 4-layer transmission (4-beam multiplexing). The benefits of NOMA (i.e. OFDMA plus power-domain user multiplexing) compared to OFDMA are summarized as follows:

- Improved spectrum efficiency/system capacity/cell-edge user performance in both macro-cells and small cells (up to 50 % gain);
- The number of simultaneously served users can be almost doubled (up to 2 users multiplexed in power-domain);
- Spectrum efficiency/system capacity/cell-edge user performance can be improved even in high mobility scenarios by exploiting the open-loop nature of NOMA.

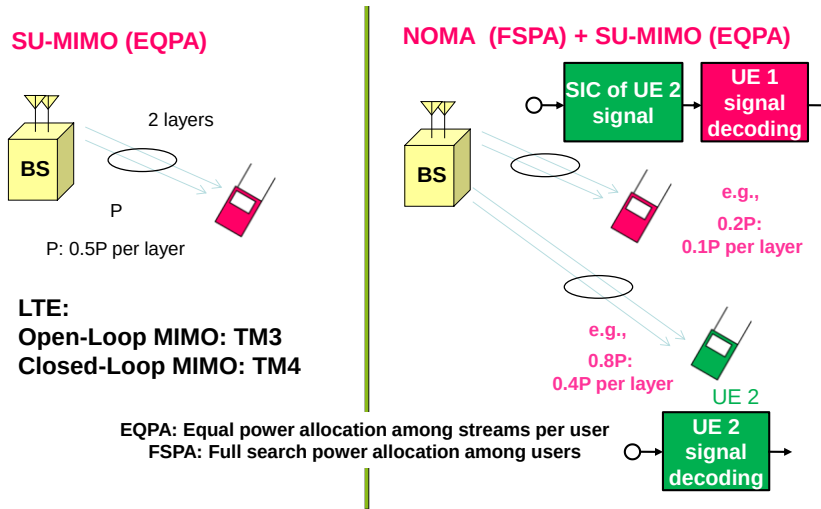


Figure 3-28 NOMA combined with SU-MIMO (2x2 MIMO & 2-UE) (cf. Eqs. (1) and (2))

Signal model

For orthogonal multiple access (OMA) case, the received signal Y_n at UE_n for 2x2 SU-MIMO is described by

$$Y_n = H_n \cdot W_n \cdot \sqrt{P} \cdot X_n + N_n \quad (1)$$

where H_n denotes the channel matrix, W_n denotes precoding matrix, X_n denotes data symbol transmitted for UE_n , N_n denotes inter-cell interference plus additive white Gaussian noise at UE_n , respectively. For each UE_n , the modulation symbols corresponding to one or two transport blocks are first mapped to N_L layers. The number of layers is referred to as Rank. The Rank is decided by the UE semi-statically and is identical over all subbands. After layer mapping, the symbols of N_L layers are mapped to transmit antennas by precoding matrix W_n . The following

two MIMO transmission modes: open-loop (TM3) and closed-loop (TM4) are considered [3GPP-25.814].

As shown in Figure 3-28, for NOMA combined with SU-MIMO, the BS simultaneously transmits a superposed signal to UE₁ and UE₂ and up to 2-layer transmission is supported for each user. From (1), the received signal $\mathbf{Y}_n$ at UE _{n} ($n = 1, 2$) can be expressed as

$$\mathbf{Y}_n = \mathbf{H}_n \cdot (\mathbf{W}_1 \cdot \sqrt{\beta_1 P} \cdot X_1 + \mathbf{W}_2 \cdot \sqrt{\beta_2 P} \cdot X_2) + \mathbf{N}_n \quad (2)$$

where β_i denotes the power ratio for UE i and $\beta_1 + \beta_2 = 1$, P denotes the transmit power of BS.

NOMA gains compared to OMA for TM4 and TM3 for different granularities of MCS selection and scheduling are summarized as follows in Table 3-6.

**Table 3-6 Performance evaluations for NOMA combined with SU-MIMO (2x2).
(3GPP case 1, 2x2, 10UEs/sector, 3km/h, FSPA (10 power sets), proportional fairness)**

Genie-Aided CQI estimation

	2x2 MIMO, TM3			2x2 MIMO, TM4		
	OMA	NOMA	Gain	OMA	NOMA	Gain
Case 1: Subband scheduling and subband MCS selection						
Cell (Mbps)	21.375	27.053	26.56%	21.97	27.866	24.73%
Cell-edge (Mbps)	0.472	0.633	34.11%	0.544	0.777	40.77%
Case 2: Subband scheduling and wideband MCS selection						
Cell (Mbps)	21.59	26.29	21.77%	22.291	27.499	23.36%
Cell-edge (Mbps)	0.476	0.62	30.25%	0.552	0.769	39.31%
Case 3: Wideband scheduling and MCS selection						
Cell (Mbps)	19.068	24.894	30.55%	19.577	25.515	30.33%
Cell-edge (Mbps)	0.401	0.538	34.16%	0.451	0.649	43.90%

Assuming 3GPP LTE simulation parameters, for case 2 (subband scheduling and wideband MCS selection (3GPP compliant MCS selection)), around 20% cell throughput gains and 30% to 40% for cell edge user throughput gains are achieved. This provides upper bound performance as genie-aided CQI (UE received SINR perfectly known at BS) is assumed. These NOMA gains are the result of improving spectrum efficiency/system capacity/user fairness by exploiting power-domain user multiplexing at the BS side and SIC receiver at UE side.

BS scheduling and power allocation, received design and signalling aspects are studied in WP2 T2.3 (TeC#11.1.1). CSI feedback design and enhancement are studied in WP3 T3.2 (TeC#8).

Conclusions

In this NOMA TeC, we designed the multi-user scheduler/power allocation and NOMA combination with SU-MIMO and evaluated the system-level performance of power-domain user multiplexing at the transmitter and inter-user successive interference cancellation (SIC) at the receiver. NOMA enables system-level improvements in terms of spectrum efficiency/system capacity/user fairness. The technology is promising to enhance system-level performance of macro-cells, small cells and moving networks.

3.11.1.2 Sparse Code Multiple Access (SCMA)

Sparse code multiple access is a non-orthogonal code/power domain multiplexing of data streams and users in either downlink (DL) or uplink (UL)

SCMA is a new modulation and a non-orthogonal multiple access scheme in which coded bits are mapped to multi-dimensional sparse codewords. Codewords of multiple layers or users are overlaid in code and power domains and carried over shared OFDMA time-frequency resources [NB13]. The system is overloaded if the number of overlaid layers is more than the length of multiplexed codewords. Overloading is achievable with moderate complexity of detection thanks to the sparseness of SCMA codewords. Inspired from LDPC, message passing algorithm (MPA) is run over a sparse factor graph of SCMA to detect transmitted codewords with a near-optimal quality of detection [BYN+14].

User multiplexing improves the overall throughput of a downlink wireless network. Spatial domain user multiplexing is realized through multi-user MIMO (MU-MIMO) transmission. Despite the promising throughput gain, MU-MIMO as a closed-loop scheme is only applicable to low mobility scenarios due to its sensitivity to channel aging. Even for low mobility users, the sensitivity to quantization error of channel state information (CSI) and the cost of feedback overhead limit the practical gain of MU-MIMO. DL multi-user SCMA (MU-SCMA) is an open-loop multiplexing scheme where different code domain layers are assigned to different users without the need of full CSI knowledge of the co-paired users [NYB+14]. Compared to MU-MIMO, this system is more robust against dynamic channel variations in high speed scenarios. The simulation results presented in [MET15-D65] for test case 2, 6 and 8 (for a short summary, see Annex A, section 6.11.1.2) show 25-70% cell average throughput and cell edge throughput gains over LTE open-loop MIMO OFDMA. Moreover, performance gain is maintained at high speed. In addition, MU-SCMA can support more than 20% network total traffic load with stringent delay and reliability constraint [MET15-D65] (for a summary, see the Annex). Therefore, this scheme is suitable for UDN and MN scenarios with high data rate and/or mobility-robustness requirements. MU-SCMA provides up to 1.5 times higher spectral efficiency compared to OFDMA, addressing the METIS goal of “higher data rate”.

One application scenario of SCMA is grant-free uplink contention-based multiple access where each SCMA layer represents a user [AZN+14]. A layer is spread across the entire time-frequency resources of a contention region. Each layer (or user) is characterized by a specific SCMA codebook. Transmission is carried over predefined contention region with predefined modulation and coding schemes without dynamic request/grant signalling. The system can be overloaded where the number of multiplexed layers is more than spreading factor. The grant-free contention-based multiple access scheme with SCMA offers the following advantages: 1) massive connectivity and coverage, 2) low signalling overhead, 3) energy efficiency, 4) low transmission latency, 5) low detection complexity due to sparseness of codeword and design of codebook, and 6) supporting low cost devices with low PAPR requirement. The KPI evaluated by system level simulation is the number of devices supported. The simulated results demonstrate that a contention-based SCMA system can support up to 3.95 times more devices than LTE R11 baseline under 3GPP case 1 scenario at an average packet drop rate of 1% [MET14-D43].

Conclusions

SCMA enables open-loop multi-user transmission in DL with low feedback overhead, less sensitivity to channel variations to support high speed users, with improved capacity and coverage gain. Furthermore, SCMA supports CoMP for seamless handover for high speed users, especially in UDN with large TP density, with reduced end-to-end delay and load balancing through inter-TP layer/power sharing. In addition, UL SCMA with the grant-free contention-based multiple access is able to provide scalability for massive connectivity and coverage, and low latency and significantly reduced signalling overhead for energy efficiency, as well as the high reliability and practically feasible detection complexity.

3.11.2 FBMC based multiple access and Cognitive Radio

3.11.2.1 Multiple Access for MIMO FBMC Systems

FBMC system with fixed subcarrier allocation for users in multiple access.

The uplink multiple access is considered here with different processing techniques to alleviate interference [SRL14a]. A multicarrier MIMO-FBMC system with $2M$ subcarriers is studied (the details of the FBMC system were given in annex 5.8, in [MET14-D23]). A single receiver and N_u transmitters in the uplink, each equipped with N_A antennas are considered. A similar scenario was considered in D2.3 where a single transmitter and $2M$ receivers in the downlink, each equipped with N_A antennas for the downlink [SRL14a]. The subcarrier filter used is $g[k]$ and the filter length is L . A time-invariant Rayleigh fading channel is assumed, where the channel delay spread spans Λ sampling intervals and where all antenna paths undergo independent fading. The channel energy is $1.2J$. A convolutional code of rate $\frac{1}{2}$ and a memory 6 is used with equalization and interference cancellation (EIC) [SRL14a]. EIC is used to mainly cancel self-interference of FBMC, whereas the convolution code is used to remove the error floor due to multiuser interference. The performance results are shown in Fig. 3-27 where 16 subcarriers per user are considered in a 256 subcarrier system with 16 users. The main observation is that with EIC and error correction code the performance of the uplink is promising and that there is no error floor.

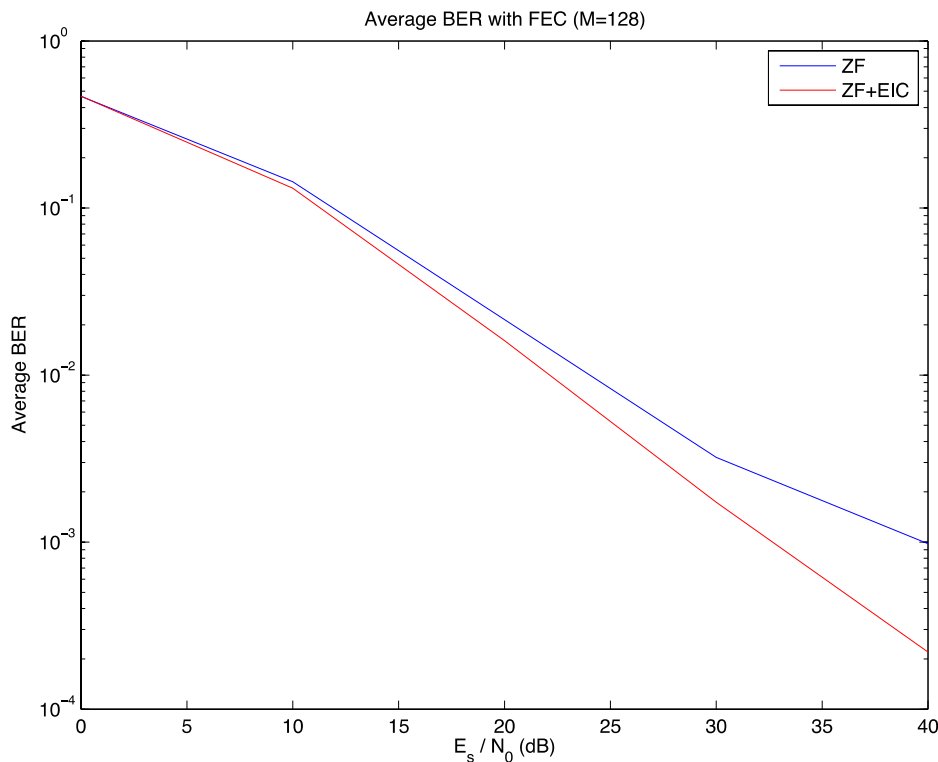


Figure 3-29 Uplink BER vs. SNR for different receivers, No. of taps in channel (Λ)=2, No. of subcarriers =256, No. of users, N_u =16, L =1023, No. of antennas=2

Conclusions

FBMC system with fixed subcarrier allocation for users in multiple access was investigated. The main KPIs are BER, SNR. Both uplink and downlink were considered. Main objective was to show feasibility as this TeC is tied to FBMC addressed in TeC#8.1. From all BER curves it is shown that with suitable receiver processing FBMC multiple access can be utilized as an alternative to OFDM.

TeC#11.2.2 - MA using Cognitive Radio

This TeC aims opportunistic spectrum access using Cognitive Radio enabling Multiple Access.

Using dynamic spectrum access with a wideband front end, opportunistic multiple access can be provided using a cognitive radio. The extra spectrum demand can be met by utilizing the TV white spaces. The demand for a wideband front end for cognitive radio (CR) puts tough challenges on the RF performance of the front end hence impacting the system SNR. The front end researched in TeC#3 is promising in this aspect. Besides this, high-IF converter can modify any commercial transceiver into a CR capable transceiver. A WLAN based cognitive radio prototype was developed and the performance was tested as in [ASV+14a] and [ASV+14b]. The comparison of WLAN-CR and WLAN alone is shown below in Table 2.1.1.

Table 3-7 Comparison of CR enabled WLAN with 802.11b/g

Parameters	802.11b/g	CR enabled WLAN
Sensitivity	-82 dBm	-90 dBm
IIP3	-20 dBm	0 dBm
Noise Figure	11.7 dB	5 dB
Max Tx Power	20 dBm (Europe)	21 dBm
Tx ACPR I	30 dBr	35 dBr
ACPR II	50 dBr	53 dBr
EVM	5.62%	2.8%

As can be seen from the above results, addition of high-IF stage satisfies the standard 802.11b/g leading to the result that addition of the high-IF converter doesn't impair the SNR of the system. The same is tested with application layer measurements which showed that the system provided steady 30Mbps goodput under varying conditions of power and frequency[SAV+14].

Conclusions

The CR radio implementation using the high-IF converter was tested against the WLAN specification and with the packet level measurements. While the testbed was developed for WLAN based applications, the duplex method is TDD, but with the replacement of TX/RX switch with duplexer of separating TX and RX chains as in [VSS+14][SVA+14], any FDD based end solution can be enabled cognitively.

3.12 Medium Access Control

3.12.1 Contention based massive access

The aim of TeCC#12.1 is twofold: (1) to propose an efficient direct random access, which combines coded random access (TeC#12.1.1) and CS-MUD (TeC#12.1.3) to enable MMC in 5G; and (2) to propose a cellular access reservation protocol capable of withstanding the MMC traffic requirements through the expansion of the contention space to the code domain.

3.12.1.1 Coded Random Access

Coded Random Access (CRA) is the embodiment of the analogies between successive interference cancellation (SIC) in slotted ALOHA framework and iterative belief-propagation erasure-decoding, where the theory and the tools of erasure-correcting are applied to enhance slotted ALOHA access. An example of Coded Random Access is depicted in Figure 3-30.

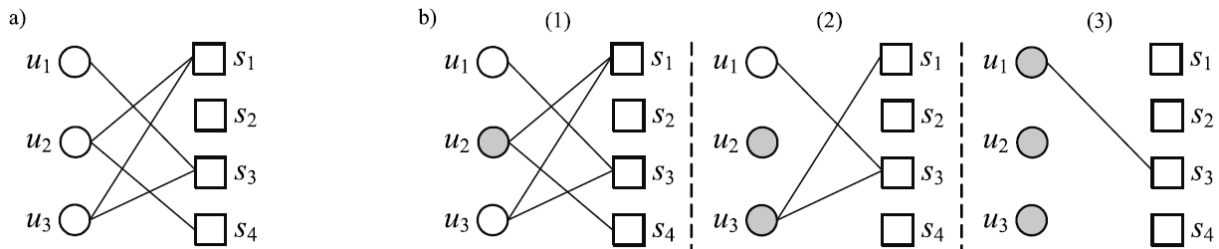


Figure 3-30 Coded Random Access where u denotes users and s slots: (a) Graph representation of coded slotted ALOHA; (b) Execution of SIC on a graph [SP13a]

Frameless ALOHA is a variant of CRA in which: (1) users contend on a slot basis, using predefined slot access probabilities, (2) the length of the contention period (in number of slots), is not fixed a priori, but determined on-the-fly such that the expected throughput is maximized. In [SP13a] it was shown that, in a single-user detection system, using a simple criterion for contention termination that is based on the monitoring of the fraction of resolved users F_R coupled with an optimized slot access probability p_a , achieves expected throughputs T that are the highest in the reported literature for low to moderate number of contending users, i.e., when the number of users N is in the range 50 – 1000. In Table 3-8 is shown a sample of the obtained results.

Table 3-8 Coded random access results, where T corresponds to GRM2#RASE.

N	50	100	500	1000
T	0.82	0.84	0.87	0.88
p_a	$\frac{2.68}{N}$	$\frac{2.83}{N}$	$\frac{2.99}{N}$	$\frac{3.03}{N}$
Threshold on F_R	0.83	0.87	0.88	0.89

Further discussion on the obtained results as well as detailed analysis of the considered scenarios is found in [SP13a] and [SP13b]. We also note that the frameless ALOHA framework, as well as CRA in general, can be coupled both with single- and multiple-user detection PHY technologies. In latter case, throughputs that exceed 1 can be achieved, as elaborated in further text.

3.12.1.2 Coded Access Reservation

Expansion of the contention space of the first stage of the access reservation protocol by creating contention codewords using orthogonal contention resources.

In this TeC a new class of access reservation protocols, denoted as coded reservation, is proposed. The contention phase is here redefined through the introduction of coded reservation tokens, which allows the expansion of the contention space to the code domain. These coded reservation tokens are obtained by combining multiple contention frames in a

virtual contention frame, and then allowing these users to select one orthogonal resource per contention frame [PTS+12, TPS+13b]. Through this procedure, the users create a coded reservation token, where the codeword length corresponds to the number of frames within the contention frame and the set of possible codewords is denoted as codebook. The contention frame construction is depicted in Figure 3-31.

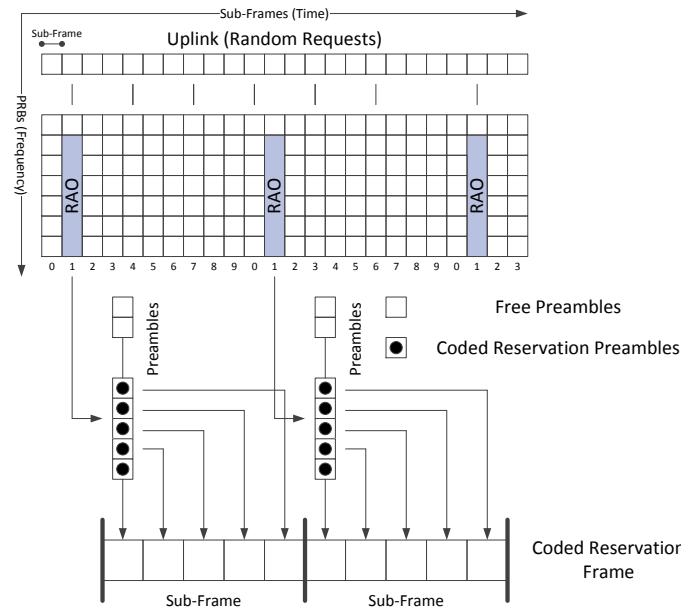


Figure 3-31 Coded access reservation contention frame construction from the LTE Random Access Opportunities (RAO) composed by multiple RACH preambles generated from Zadoff-Chu sequences

The channel that enables the Coded Access Reservation is the Multiple Access OR Channel, where the concept of superimposed codes is applicable. In Figure 3-32 is depicted how in this setting the transmission and detection is performed. A drawback of this approach is the occurrence of False Positives, which can be controlled through proper codeword design.

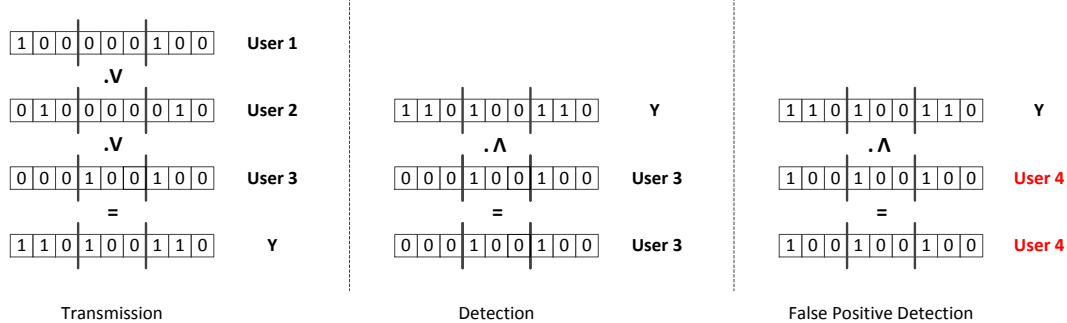


Figure 3-32 Coded access reservation illustrative example of transmission, detection and False Positive Detection

The Coded Access Reservation protocol enables three access modes:

- (B) where the random access occurs using random generated orthogonal codewords, denoted as Baseline Mode, which represents the one present in current cellular systems such as LTE [TPS+13a];
- (R) where the random occurs with random generated codewords, denoted as Random Coded Access Mode [PTS+12, TPS+13b];
- (S) where the access is performed with unique signatures, where each user has been assigned a unique access codewords, denoted as Signature Mode.

The results, as the ones depicted in Figure 3-33, show the existence of switching points between these three regimes, which depend on the user access load. Namely, for low, medium and high loads the suitable modes are respectively the B, R and S modes.

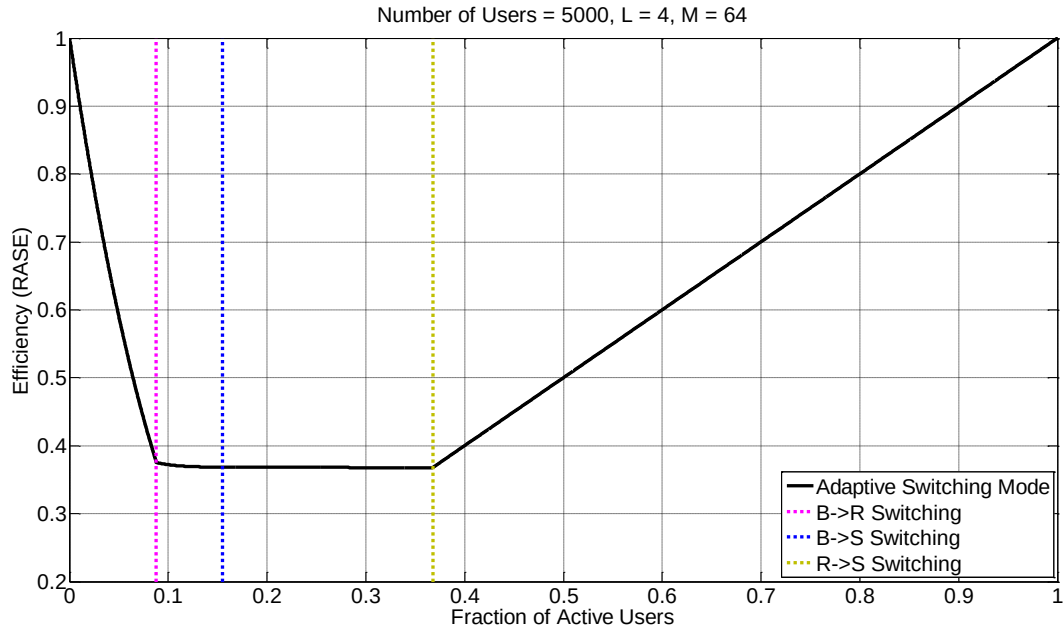


Figure 3-33 Coded Access Reservation supported access modes: (B)aseline (with orthogonal codewords similar to LTE); (R)andom (with Non-Orthogonal randomly selected codewords; and (S)ignature Access Reservation (Non-Orthogonal uniquely assigned codewords).

TeC#12.1.2 Conclusions

The Coded Access Reservation Protocol allows creating a large contention space, which the main benefit is the reduction of the occurrence of collisions. An alternative use of this approach allows the creation of a large amount of contention signatures which then allows the network to assign a unique signature to each user. The design of codewords that minimize the occurrence of phantoms is still an ongoing research topic.

3.12.1.3 Advanced physical layer processing for enhanced MAC: Joint detection of node activity and data

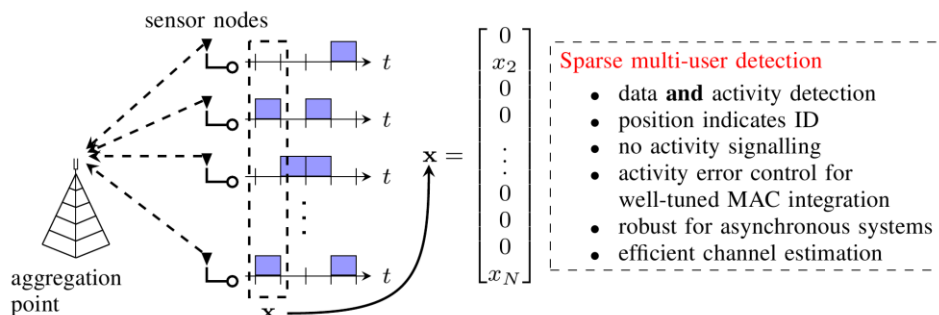


Figure 3-34 CS-MUD considered scenario and main results

Massive Machine Communication requires signalling-efficient MAC and PHY technologies, which are able to cope with the huge number of future MTDs in a resource efficient manner. LTE will be unable to handle the massive amount of MTDs envisioned for the future due to a system design focusing on high data rates and human only traffic needs. LTE's current PRACH procedure is used to grant access to scheduled resources which is very inefficient for

small packet sizes and intermittent/sporadic user activity. Therefore, direct random access procedures seem better suited to facilitate future low data rate MMC. To this end TeC#12.1.3 “Advanced physical layer processing for joint activity and data detection” aims to provide efficient PHY concepts for direct random access by exploiting the properties of sporadic uplink communication through sparse signal processing. Assuming MTC with sporadic activity as indicated by the time axis in Figure 3-34, many MTDs will only be active occasionally which leads to a sparse representation of the multi-user vector in typical MUD problems indicated by the vector $\mathbf{x}$ in the figure. Compressive sensing based multi-user detection (CS-MUD) exploits this sparsity to achieve a reliable joint activity and data detection to facilitate efficient direct random access. In METIS deliverable D2.1 the following research directions have been defined: (1) measures of system impact, (2) channel estimation, (3) channel coding aspects, and (4) effects of imperfections. These points will be addressed in the following by a short summary of achieved results and appropriate references for further information. Some results have already been presented in METIS deliverable D2.3 [MET14-D23], specifically in Annex 5.12.1, and will not be repeated here.

Regarding the first point, activity errors have been clearly defined as important measures of system impact as higher layer processing (MAC protocols, HARQ, etc.) may strongly depend on these. Based on these measures, multiple approaches have been developed to control the impact of CS-MUD on higher layers by tuning activity errors [MBD13a, MBD13b, MBD14]. Most promising is the application of Neyman-Pearson estimation which allows to tightly control activity errors [MBD15].

Regarding the second and fourth point, channel estimation and asynchronous communication have been investigated and solved by appropriate CS-MUD algorithms [SBD13b, SBD13c] such that nearly optimal (perfect CSI) performance could be achieved with reasonable complexity and pilot overhead. Reasonable assumptions required for these approaches are statistical model knowledge like typical channel models, delay distributions and maximum delays.

Regarding the third point, CS-MUD has been successfully integrated with channel coding to exploit the structure of the channel code for enhanced user activity detection [SBD13]. Through iterative processing a nearly error free activity detection is achievable and therefore performance is similar to scheduled access without control signalling or access reservation schemes [SBD13a].

Figure 3-35 summarizes the achieved results in terms of the scalability of TeC#12.1.3 assuming non-orthogonal medium access, 128 users, an SNR of 0dB and a trivial MAC (single slot, no ARQ, etc). On the x-axis the overloading factor for non-orthogonal access (users/orthogonal dimension) and on the y-axis the supported average number of active users is depicted. The former is a medium access design variable, while the latter depends on the traffic statistics as well as users in a cell. The depicted GRM5#PER 10^{-2} contour [METIS D2.1] provides a clear indication for the medium access design given a certain user load at a required QoS level. Furthermore, the scheduled access (known user activity) with least squares (“scheduled access LS”) and CS-MUD with channel estimation results are shown for comparison. Compared to the scheduled access CS-MUD exhibits a near constant performance loss over all shown overloading factors. This gap can be closed by iterative strategies like the ones discussed in [SBD13a] to achieve direct random access with the same performance as scheduled access. Additionally, channel estimation introduces only a small performance loss in terms of GRM5#PER if the maximum channel delay $\tau_{\max}$ is known or overestimated as indicated by “CS-MUD with CE” in the figure. However, if the maximum channel delay is underestimated, performance is severely degraded demonstrated by “CS-MUD with CE, $\tau_{\max}$ underestimated”. Note, that Figure 3-35 indicates the average # of active users that can be supported by CS-MUD, the actual # of active users for each access slot may vary according to the traffic statistics which are here modelled by a Bernoulli assumption.

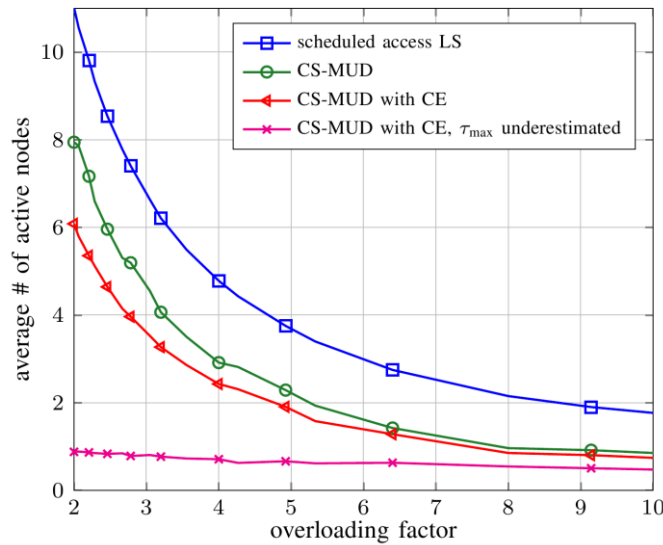


Figure 3-35 GRM5#PER contour lines @ 10^{-2} for an SNR of 0dB assuming a trivial MAC for different degrees of overloading (users/dimension) for non-orthogonal access and the average # of active users out of 128. The contours show the scaling of supported average users with the overloading of resources by non-orthogonal access schemes.

In summary, the concept of CS-MUD has been fully investigated in terms of the posed research directions with excellent PHY layer performance. Furthermore, CS-MUD is highly adaptable to various communication system aspects like channel coding, high layer requirements and - as will be shown in the following - a combination with an appropriate MAC protocol. Therefore, CS-MUD provides a mature PHY TeC to tackle the challenges of massive machine communication in future 5G systems.

3.12.1.4 Joint TeC#12.1.1 and #12.1.3 evaluation

Furthermore, TeC#12.1.1 “Coded Random Access” and TeC#12.1.3 “Advanced physical layer processing for joint activity and data detection” have been combined towards an integrated PHY/MAC-Solution for MMC [JSC+14]. Figure 3-36 depicts results in terms of GRM2#RASE for an SNR of 5dB and 10dB. GRM2#RASE denotes the MAC efficiency of this combined PHY and MAC processing approach in terms of the number of successfully served users per slot, see [METIS D2.1]. The x-axis denotes the number of slots as a design parameter for the MAC scheme. The optimal number of slots per active users with highest GRM2#RASE is marked by a cross in the plot and is clearly dependent on SNR and other assumptions thus requiring dynamic control [SP13a, SP13b]. The most important assumptions are: (i) perfect channel knowledge, (ii) the average number of active users per slot, i.e., the slot access probability, is controlled to achieve maximum throughput [SP13a, SP13b] and (ii) the PHY layer employs CDMA spreading with two different codebooks. The “full” codebook uses random PN sequences with spreading factor $N_s = 32$, the “limited” codebook contains a subset of 128 random PN sequences from which users randomly choose. Furthermore, the considered message size is 50 information symbols for all active users. For further details refer to [JSC+14].

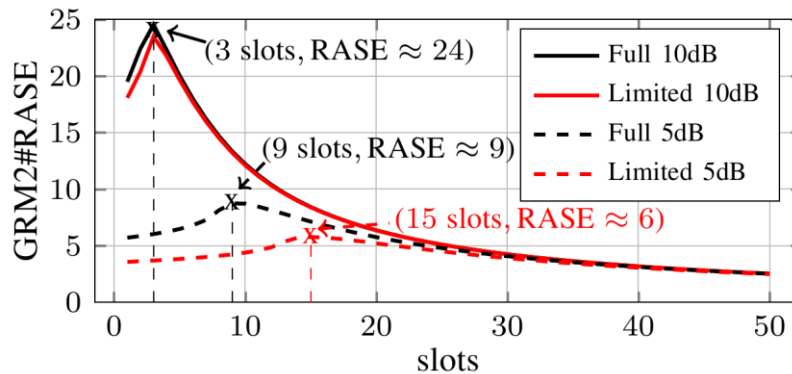


Figure 3-36 Integrated PHY/MAC approach using TeC#12.1.1 with #12.1.3. Results in terms of GRM2#RASE vs. slots using an exemplary setup with 128 users at SNR=5dB and SNR=10dB.

At an SNR of 10dB TeC#12.1 achieves 24 users/slot using overall 3 slots and at 5dB this reduces to 9 users/slot and 9 slots for the full codebook. When a limited codebook of PN sequences is introduced, every node accessing the system has to choose one codebook entry randomly introducing collisions. As a result performance of the physical layer processing is slightly lowered and thus the overall system performance decreases. The impact at 10dB, however, is negligible whereas at 5dB performance is severely degraded from 9 to 6 users/slot using 15 instead of 9 slots. In summary at low SNR even small differences in the PHY performance highly impact the overall MAC performance due to the smaller probabilities of successfully decoding a user compared to high SNR requiring careful system design. At high SNR successful decoding is more likely and just slightly impacted by PN sequence collisions that may be resolved by the MAC interference cancellation proposed by TeC#12.1.1 “Coded Random Access”. For further details please refer to the annex.

This joint approach shows promising results in comparison to the LTE Rel. 11 baseline. Up to 10x the number of MTD can be supported for short messages. Note, that an exact comparison is almost impossible due to differing assumptions and the fact that the LTE resource requirements for sporadic MTC one time data transmissions are hard to calculate exactly due to the PRACH procedure and fixed PRB size. For further details please refer to the annex.

Further, we note that the same combined approach can be applied in the context of other non-cellular technologies such as the ones based on the 802.15.4 (Zigbee, WirelessHART, etc...), where gains in similar or higher magnitude as compared to the cellular systems would be expected. A more detailed analysis of the 802.15.4 PHY and MAC would be required to provide specific results, which has not been the focus on METIS, since the baseline was the LTE-A Rel.11.

Joint TeC#12.1.1 and Tex#12.1.3 Conclusions

TeC#12.1.3 provides an efficient PHY solution to solve sporadic MMC in combination with MAC protocols like TeC#12.1.1 “Coded Random Access”. It provides a well-integrated mature approach well matched to asynchronous, sporadic MMC uplink communication, where the active users, delays and channels are generally unknown. First results of the combined TeC#12.1.1 and #12.1.3 integrated PHY/MAC shows a gain of up to 10x more users supported users given by GRM2#RASE [METIS D2.1] compared with LTE Rel. 11 showing the potential for a future application in 5G systems.

3.12.2 Distributed network synchronization

TeC#12.2 studies efficient distributed network synchronization schemes for the networks without a central controller or with limited contact with a central controller.

Time agreement, i.e., a common notion of time, is crucial for many applications. In the context of METIS, for instance, time related channel access mechanism in MN (e.g., T2.1-TeC#6.5 Ad-Hoc MAC for V2V) requires time agreement. Besides, in D2D communications, device discovery and synchronous transmission need time agreement as well. The procedure to achieve time agreement across a network is defined as network synchronization.

For networks without a central controller or with only limited contact with a central controller, the nodes need to conduct network synchronization in a distributed fashion. This is a more challenging problem and is needed in METIS to manage V2V or D2D links when the vehicles or devices have limited or no contact with the fixed infrastructure. Note that here we consider a vehicle is connected to a central controller if it has an accurate enough GPS no matter whether it is indeed within the coverage of a central controller or not. In this case, the possible synchronization scenarios are partial-coverage synchronization and out-of-coverage synchronization, as illustrated in Figure 3-37.

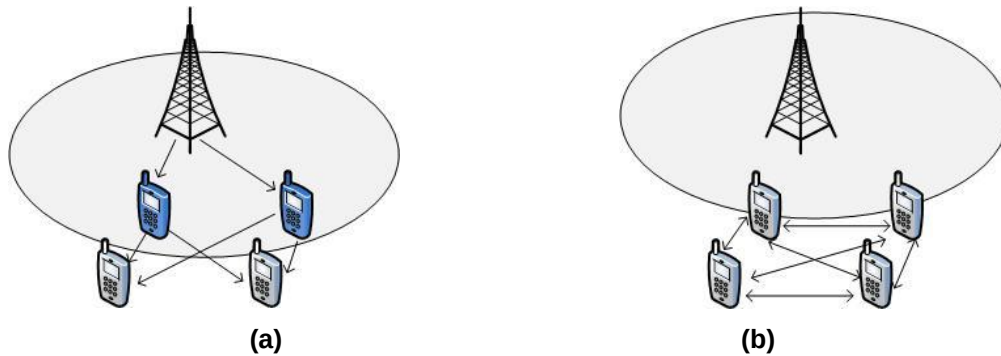


Figure 3-37 Two synchronization scenarios in MN and D2D, where the arrow lines represent the possible transmissions of synchronization messages. a) partial-coverage synchronization scenario; b) out-of-coverage synchronization scenario.

For notational consistency, from now on, we denote vehicle or device as node; vehicle or device which is connected to a central controller as leader; and vehicle or device which is not connected to a central controller as follower. Besides, assume each node is equipped with an affine clock model, i.e., $T_i(t) = f_i t + \theta_i$ for node i , where t is the perfect time, f_i indicates the clock frequency, and θ_i denotes the physical clock offset. Moreover, we assume that all the leaders have the same frequencies and offsets. Therefore, the objective of network synchronization is to synchronize all the nodes to the leader(s) for the partial-coverage scenario or synchronize all the nodes to an (arbitrary) common value for the out-of-coverage scenario.

In this study, we focus on MAC layer distributed clock synchronization. In other words, nodes broadcast timing messages which contain the timestamps recorded by the clock of the transmitter; these messages are in turn used to adjust the clocks of the receivers. Even though extensive researches have been carried on in this direction, very few of them are specially designed for MN or D2D scenarios which impose the following **5 specific challenges** on synchronization.

1. Initially, followers do not know if it is the partial-coverage or out-of-coverage case.
2. For partial-coverage synchronization scenario, how to spread leaders' clocks in a fast and reliable manner? Especially, when multiple leaders exist, how to further make use of this advantage?

3. For out-of-coverage synchronization scenario, how to conduct distributed operation with only local information to achieve the consensus on both clock frequency and clock offset?
4. When a new node joins an almost synchronized group, it should not induce any big change.
5. Even though MAC-layer time stamping [IEEE802.11] can be utilized to largely reduce the effects of transmission delays, there will still be some delays remaining. As a result, how to deal with the inaccurate timestamps?

To deal with the above 5 challenges, we propose an Adaptive distRibuted nEtwork Synchronization (ARES) scheme which is a complete framework including both transmission and receiving mechanisms. Detailed descriptions of the main principles are presented in the Annex.

For performance evaluation, we compare the proposed ARES with the following three baseline methods:

- 1) Time synchronization function (TSF) specified in [IEEE802.11].
- 2) Modified Automatic Self-time-correcting Procedure (MASP) in [PTM10] which is a variety of the TSF.
- 3) Random Broadcast based Distributed consensus clock Synchronization (RBDS) proposed in [SSBG15] and [MET14-D23].

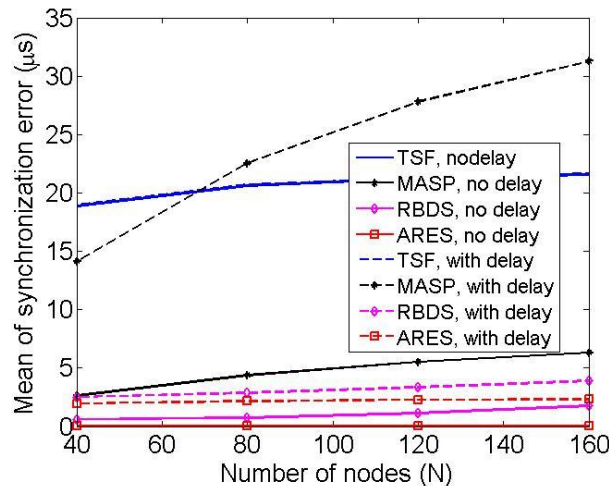


Figure 3-38 Synchronization error versus number of nodes

Figure 3-38 evaluates the mean of synchronization error with respect to different number of nodes. Here we consider the scenarios without or with transmission delays, where the delays are uniformly distributed with mean 0 and variance 4 microseconds for the latter case. As shown in Figure 3-38, the error of TSF is quite high, where the reasons are mainly twofold. Firstly, only clock offsets but not frequencies are adjusted in TSF. Secondly, the fastest node asynchronization problem [SSBG15] may dominate its performance due to the converge-to-max principle. Additionally, even though MASP significantly outperforms TSF in the no delay case, it is very sensitive to delays, which limits its feasibility in practice. Moreover, compared to TSF, MASP, and RBDS, the ARES reveals not only obviously improved accuracy but also robustness against the number of nodes for both scenarios.

Conclusions

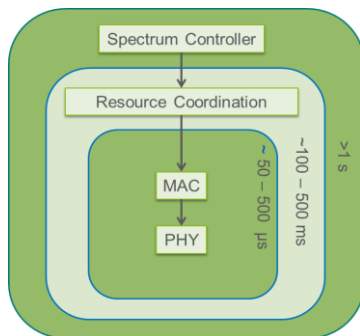
Proposed ARES is a promising synchronization scheme that can tackle the specific synchronization challenges in MN and D2D systems where not all of the vehicles or devices are in the coverage of the central controller. Compared to the baseline TSF specified in [IEEE802.11], the proposed ARES scheme can support much higher number of vehicles or devices for a given accuracy threshold and a target probability of asynchronization. The approach is mature in the sense that it is a complete procedure including both transmission and receiving mechanisms, and that it can be applied into practical scenarios where transmission delays exist.

3.12.3 MAC for UDN and mmW

This technical component provides MAC protocol operating at mmW frequency in dense deployment requiring high-gain beamforming and wireless self-backhauling.

High frequencies, which are needed to get enough spectrum for multi-Gbps data rates, require high-gain beamforming (BF) to make the link budget work. However, with high gain BF the problem of hidden nodes in a plain-vanilla contention based MAC protocol (e.g. CA/CSMA as in Wi-Fi) increases. A scheduled MAC is in this aspect better.

In a mesh network as we have in the wireless self-backhauling (BH), which is an interesting deployment option for very dense deployments to save costs and add flexibility, a scheduled MAC is not obviously the best choice since “to schedule someone” implies a hierarchy of nodes. Such a hierarchy is not obvious in a meshed network as we see in the wireless self-BH. Therefore a distributed MAC protocol is here interesting. The contention-based MAC is an example of a distributed MAC. Our UDN requirements of high-gain BF and wireless self-BH have therefore contradicting requirements on MAC.



With these aspects in consideration, a layered resource management (cf. figure below) is proposed for the UDN scenario from spectrum availability to the physical layer as reported in [MET14-D23]. The spectrum controller functionality (long term spectrum coordination functions) typically operates on a time scale of the order of $>1s$ and determines resources a UDN may use. The resource coordination functions, such as interference aware routing functions, operate on a shorter time scale, e.g., 100-500 ms, determine routes in a wireless self-backhaul and perform partial resource assignment. MAC and PHY layers operate at even shorter time scales, 50-500 μs .

The MAC determines the actually used resources. PHY operates on the indicated resources.

A MAC approach with contention control signalling to allocate data resources provided by the resource coordination functionality is carried out and evaluated with the METIS test case 3.

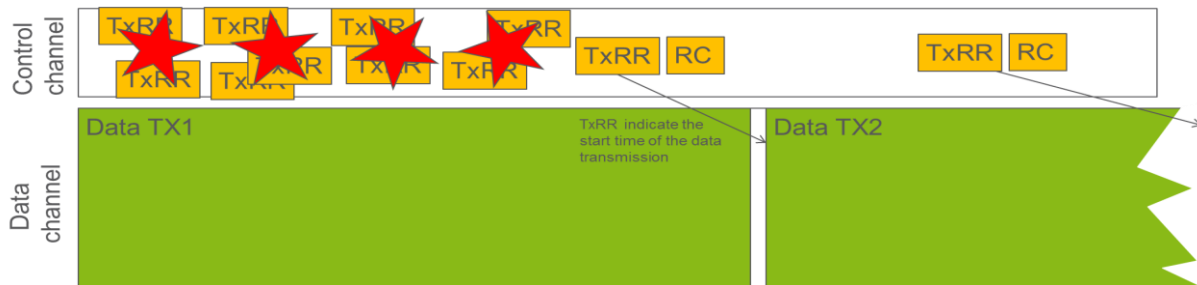
Control signals, *transmission reservation request* (TxRR) and *resource confirmation* (RC), are transmitted in time-frequency resources separated from the data resources, using directive beamforming allowing only the set of other nodes that will be interfered by the transmission to overhear the resource reservation.

The access to the control channel is via the same type of *carrier sense* (CS) mechanism that is used in 802.11 Wi-Fi systems, i.e., the DCF⁵ where a random back-off is selected and if

⁵ DCF: The so called distributed coordination function is used in 802.11 Wi-Fi systems in which each node listens for the channel status for a short time (DCF inter-frame space - DIFS) and decides if the channel is unused. If unused, the node waits a random back-off time before accessing the channel, thereby allowing other nodes to get a fair chance to access the channel.

there has not been any signal detected present in the control channel the transmission is performed.

Once a successful TxRR-RC exchange has taken place the planned (as specified in the TxRR message) data transmission will take place using the time-frequency resources indicated in the TxRR, as depicted in the figure below.



The performance evaluation showed that this MAC approach works well with wireless backhauling and the METIS TC3 throughput and file delay transfer KPIs are met (cf. table below). See Annex A for more details.

Performance	Target value	Simulation result	
		6 AgN	6 AgNs + 4 wAN
Experienced throughput*	DL: 300 Mbps	690 Mbps	565 Mbps
	UL: 60 Mbps	225 Mbps	339 Mbps
Traffic volume	5.6 TB/h (70% DL)	Ok**	Ok**
RAN latency*	5 ms	~ 1 ms	~ 1 ms
Availability	95% of the space	128/128	128/128
Reliability*	1 s file transfer	296 DL (745 UL) ms	430 DL (495 UL) ms
Cost		Simple beamforming	+wAN.
*Worst 5 th percentile		**Set as user/packet arrival setting	

Conclusions

The contention based MAC works well in the studied scenario with wireless backhauling and high gain beamforming (cf. TeCC#1) indicating the possibility to deploy low cost and easy installed access nodes. METIS TC3 throughput and file delay transfer KPIs are met by the MAC (cf. tables below). Performances are expected more efficient in other deployment (NLoS, coverage extension, etc.)

Adding more access nodes (ANs) would result in less users per AN and the contention will be easier (less contention, higher resource availability at AN per user). The results also show that high gain BF sufficiently isolates the transmissions between access nodes.

3.13 Hybrid Automatic Repeat Request (HARQ)

3.13.1 Backtrack Retransmission with Ternary Feedback

Delayed Channel State Information: Backtrack Retransmission with Ternary Feedback.

In this TeC, we consider the problem of achieving high average throughput with an average delay constraint over a block-fading channel with quantized delayed CSIT. In particular, we address the performance of the recently proposed *Backtrack Retransmission (BRQ) scheme* (in TeC#13.1) when only ternary CSI feedback is available after each block. The main observation is that the average throughput is only slightly lower than that of BRQ. The scheme thereby provides substantial throughput improvements over HARQ for a fixed delay constraint.

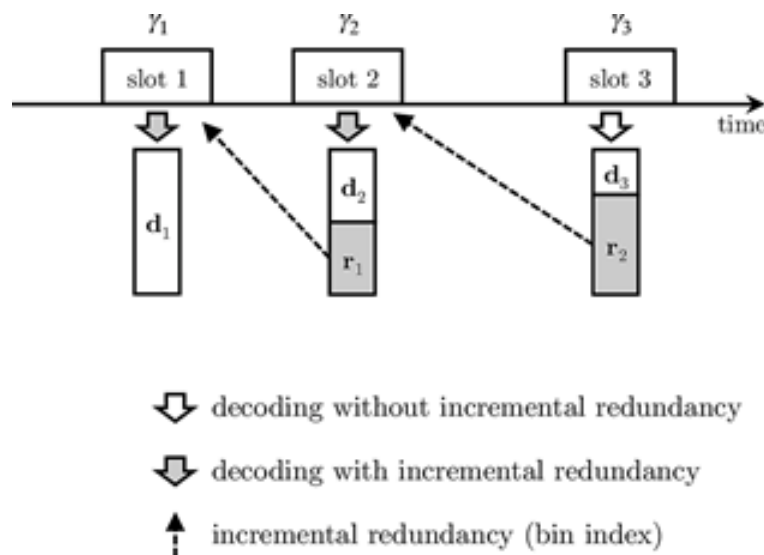


Figure 3-39 Example of backtrack retransmission with delayed CSIT. In the n -th block, d_n are new bits and r_{n-1} are incremental redundancy bits used to decode the transmission in slot $n-1$.

In contrast to communication with prior CSIT, zero outage cannot be guaranteed when transmission is done over a single block and only delayed CSIT is available. BRQ with full delayed CSIT uses fixed-length packets and a fixed-rate R codebook, and operates as Markov block coding [P+14]. In a given block n , the transmitter encodes a combination of new information bits and incremental redundancy (IR) bits. The key idea is to send the exact number of IR bits needed to decode the previous block and using the remaining bits for new information bits. If the packet in block $n-1$ was successfully decoded, the number of IR bits is zero, whereas the number of needed IR bits is otherwise $N(R - \log_2(1+\gamma_n))$, where N is block length of each slot, R is the slot rate and γ_n is the SNR in the n^{th} slot. When a packet is successfully decoded, the IR bits are used to decode the previous undecoded packets successively. See Figure 3-39 for an example. The main novelty of BRQ is thereby the concept of injecting new data before previous transmissions are successfully decoded. An important property of BRQ is that the ergodic capacity of the block-fading channel is approached as the rate R increases.

While the fixed-rate requirement on the codebook enables low encoder and decoder complexities, it also requires the use of full delayed CSIT to approach the ergodic capacity. In most practical systems, it is of interest to minimize the number of feedback bits. In the proposed scheme, we only employ ternary feedback, where each feedback value directs the transmitter to a specific action. The scheme is thereby a generalization of HARQ and uses the concept from BRQ. A disadvantage of the scheme is that the blocks cannot be decoded in a block-by-block fashion, but a joint decoding approach similar to that used for IR-HARQ is needed.

The proposed scheme is evaluated by assuming Rayleigh block fading, independent from block to block. Figure 3-40 depicts the average throughput of various schemes in terms of

average delay for SNR of 10 dB and 30 dB. The plots show the achievable throughput for the following schemes: (1) the ergodic capacity, (2) an outage approach where the transmitter encodes information bits over a fixed number, M , of blocks, (3) BRQ with full CSI, (4) BRQ with ternary feedback (the proposed scheme) (5) IR-HARQ with an unlimited number of allowed retransmissions, and (6) IR-HARQ with a limited number of allowed retransmissions. All schemes are optimized to provide the best possible throughput subject to the corresponding delay constraint and thereby also approach the ergodic capacity as the average delay constraint is increased. The outage approach yields a lower bound on the achievable throughput. The IR-HARQ achieves significantly higher average throughput since it uses feedback to terminate transmissions as soon as the receiver is able to decode. It is observed that BRQ yields significant rate improvements compared to IR-HARQ, especially at high SNR. The average throughput of BRQ with ternary feedback is slightly lower but still provides substantial throughput improvements compared to IR-HARQ.

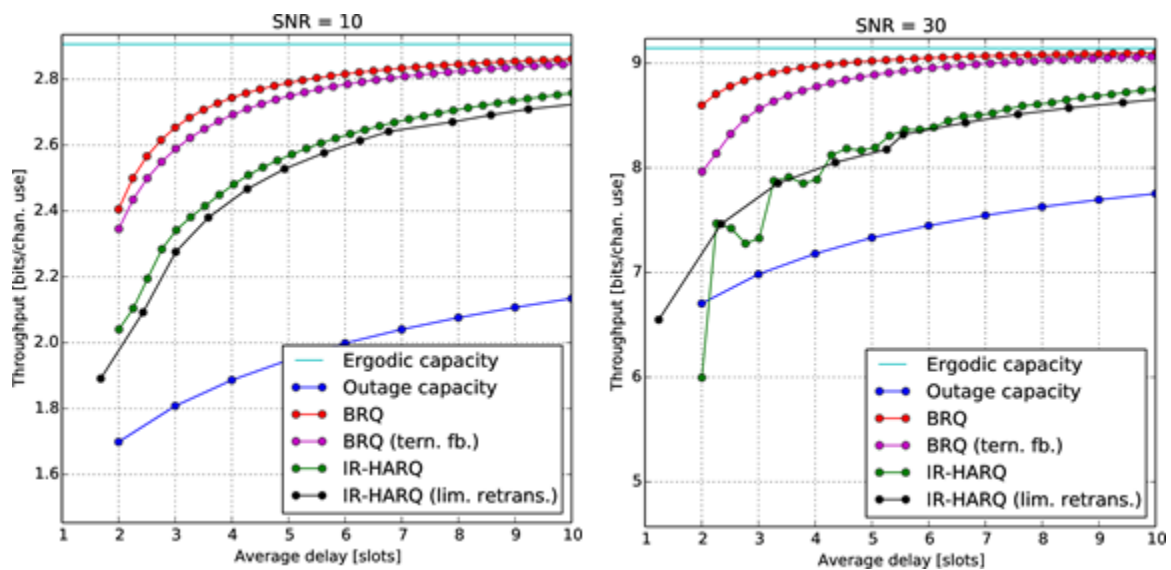


Figure 3-40 - Average throughput in terms of an average delay constraint

Conclusions

BRQ and BRQ with ternary feedback offers substantial improvements compared to IR-HARQ and outage transmission for low latency. BRQ with ternary feedback achieves throughputs that are only slightly worse than for BRQ. BRQ is an optimal protocol within a certain class of protocols. An implementation using practical codes is seen as future work.

3.13.2 Reliability-based HARQ

Reliability-based HARQ [TVP+03] [MET13-D22] uses information or feedback about the quality of an incorrectly received packet, in order to adapt the format of the retransmission, such as size (in terms of physical resources), modulation or coding of the retransmission.

In the first paragraph, the throughput and delay performances of reliability-based HARQ are assessed for user velocities up to 250km/h by using link level simulations with a detailed modelling of the signal processing functions under conditions realistic for the LTE system.

In the second paragraph, a practical implementation of a reliability-based HARQ scheme using a 2bit ACK/NACK for the reliability feedback [WWD+14] is considered, and the performance of asymmetric QPSK modulation for conveying the 2bit ACK/NACK is assessed.

Throughput and delay performance with aggressive link adaptation

Figure 3-41 (a) depicts the normalized throughput versus SNR for data transmission on LTE physical uplink shared channel (PUSCH) over VehA channel with 30, 60, 120 and 250km/h user velocity, obtained by means of a single user link level simulation with Incremental Redundancy. An aggressive link adaptation is used with a 90% target BLER after the first transmission in order to enhance throughput over a more common ~10-20% target BLER [WWD+14] and to have good benefit from variable size retransmissions, where 5MHz BW (25 PRB pairs) are allocated to the user for the first transmission. The LTE baseline scheme (dashed) uses full-size retransmissions, i.e. the size of the retransmission is identical to the size of the first transmission of a packet, while the reliability-based HARQ scheme (solid) adapts the size of the retransmissions (i.e. the number of allocated PRB pairs) according to BER estimates obtained from Turbo decoder soft output (LLRs), where it is assumed that the reliability-information is available at the transmitter free of transmission errors. The throughput in terms of correctly received bits is normalized by the number of required PRB pairs, as in [WWD+14]. At moderate SNRs, reliability based HARQ provides throughput gains of up to 34%, 27%, 29% and 32% at user velocities of 30, 60, 120 and 250 km/h, respectively.

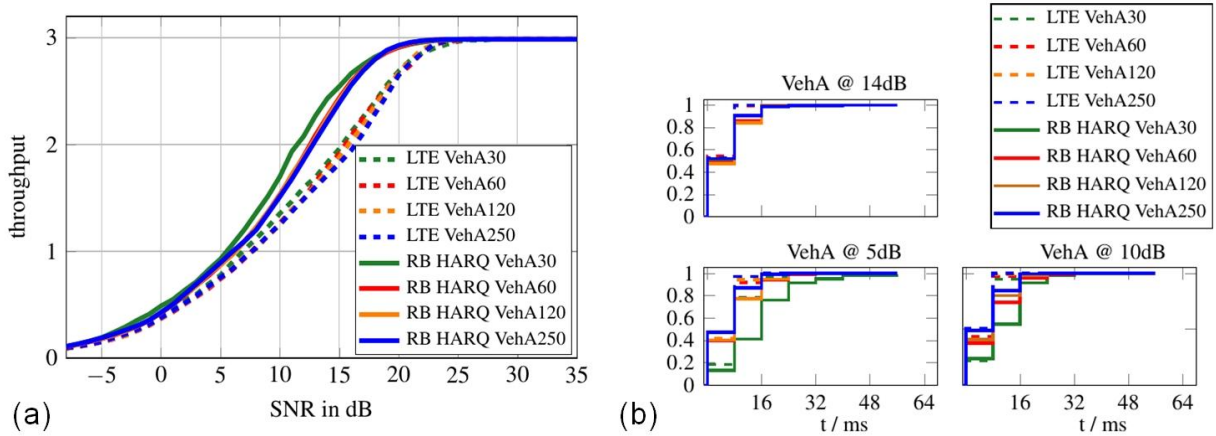


Figure 3-41 Normalized throughput versus SNR (a) and delay CDFs at selected SNRs (b).

Figure 3-41 (b) depicts CDFs of the respective retransmission delays from the first transmission of a packet until successful decoding of the packet. It can be seen that reliability-based HARQ requires an extra retransmission for about 10-20% of the packets (i.e. ~0.8-1.6ms extra delay on average), where the increase in retransmission delay decreases with increasing user velocity.

Asymmetric QPSK modulation for conveying the multi-level ACK/NACK

As shown in [WWD+14], the reliability feedback for reliability-based HARQ can efficiently be implemented by means of a 2bit ACK/NACK message with three NACK levels $NACK_i, i \in \{1, 2, 3\}$, de facto without impacting the data throughput while accepting a $NACK_i \rightarrow NACK_j$ error rate below 3-5%.

Figure 3-42 (a) shows an asymmetric QPSK (A-QPSK) modulation scheme with parameter α , and Figure 3-42 (b) shows the transmit power saving of A-QPSK over QPSK, when conveying the 2bit ACK/NACK over AWGN channel. It can be seen that A-QPSK achieves power savings over QPSK of up to ~1.38dB for conveying the ACK/NACK. This power saving is achieved by relaxing the $NACK_i \rightarrow NACK_j$ error rate to below 3%, while the $NACK \rightarrow ACK$ error rate is below 0.1% [NKM+07]. The result is obtained numerically from approximations of the ACK/NACK error rates by assuming equal a priori probabilities of the $NACK_i$.

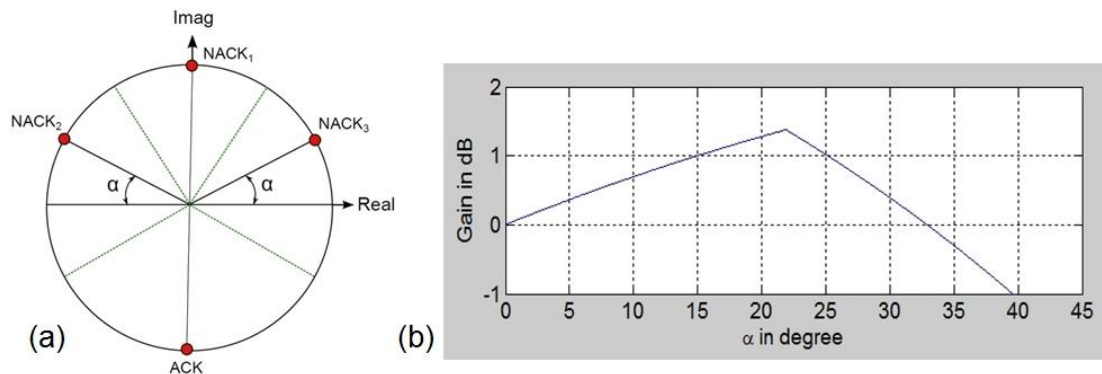


Figure 3-42 A-QPSK modulation constellation (a) and power saving over QPSK (b).

The performance of A-QPSK modulation was also assessed by means of link level simulations for conveying the 2bit ACK/NACK on LTE Physical Uplink Control Channel (PUCCH):

- For AWGN channel, the power saving of A-QPSK over QPSK and optimum value of α are a bit higher as compared to Figure 3-42, with 1.9dB gain for $\alpha = 30^\circ$.
- With multi-path fading (ePedA3, VehA30), the power saving of A-QPSK over QPSK appears similar as in the presence of AWGN, with 1.6dB – 1.9dB gain for $\alpha = 30^\circ$.

Further details on A-QPSK performance for 2bit ACK/NACK can be found in [BDZ+14].

Conclusions

The major conclusions obtained in this work can be summarized as follows:

- It was confirmed that reliability-based HARQ has potential for throughput improvements over conventional HARQ of about 25% [CWP03] to 35% [WWD+14], and those can be achieved also for the higher user velocities up to at least 250km/h.
- By the very detailed modelling of the signal processing functions (e.g. computation of reliability information via soft decoder output), it was given further evidence that reliability-based HARQ is feasible to implement.
- We proposed a practical implementation of the reliability feedback by means of a 2bit ACK/NACK. Mapping the 2bit ACK/NACK to an asymmetric QPSK modulation symbol provides transmit power savings of up to 1.6dB – 1.9dB versus QPSK.

3.14 Radio link enablers for Radio Resource Management (RRM)

In order to support a smart radio resource management (RRM) algorithm for Device-to-Device (D2D) communications, channel state information (CSI) of overall cellular and D2D links are preferable if they are available at base station (BS) side. However, due to the fact that CSI from potential cellular users to D2D receivers can only be collected in a very cumbersome manner, we propose a signalling scheme with partial channel state information available and exploiting user position information.

As can be seen from Figure 3-43, BS will gather channel gain information of cellular uplinks and D2D links, and also the channel gain information from all D2D transmitters to serving BS. The BS performs channel estimation for cellular uplink users and D2D-Tx users using for e.g. pilots. Besides, position information of users can also be collected within the same signalling scheme. For instance, GPS position can be reported by each user to serving BS in Step 1 in Figure 3-43. These user positions will be used in RRM algorithm carried out by BS, in order to mitigate interference from cellular links to D2D links.

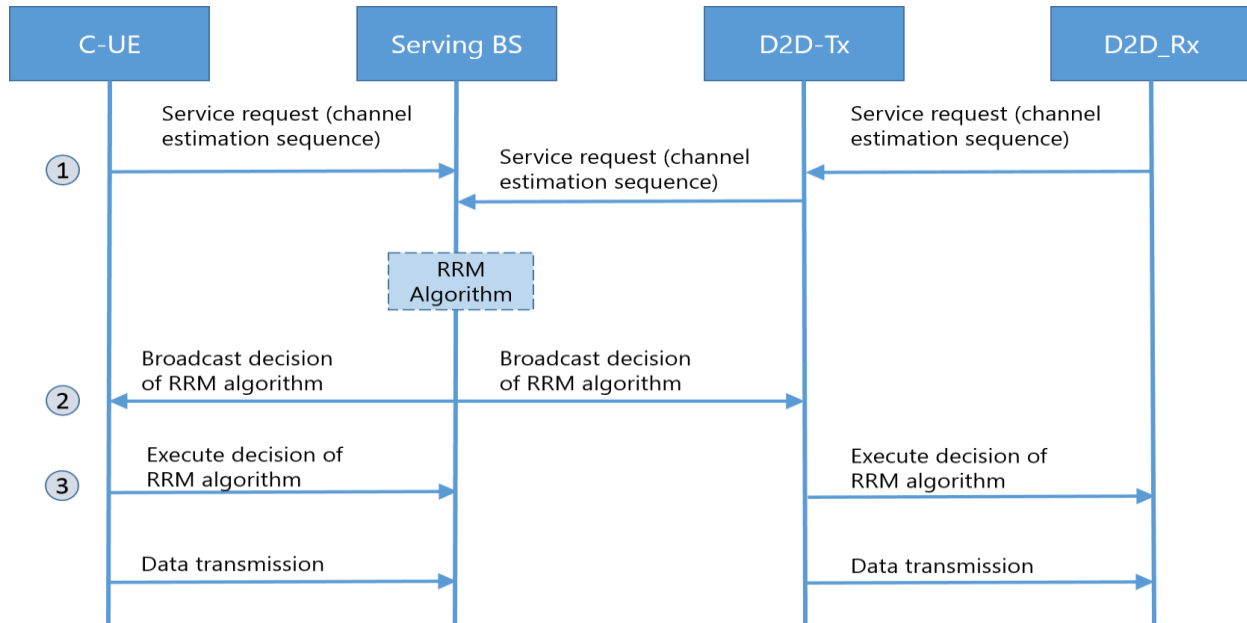


Figure 3-43 Signalling scheme for D2D communications with partial CSI

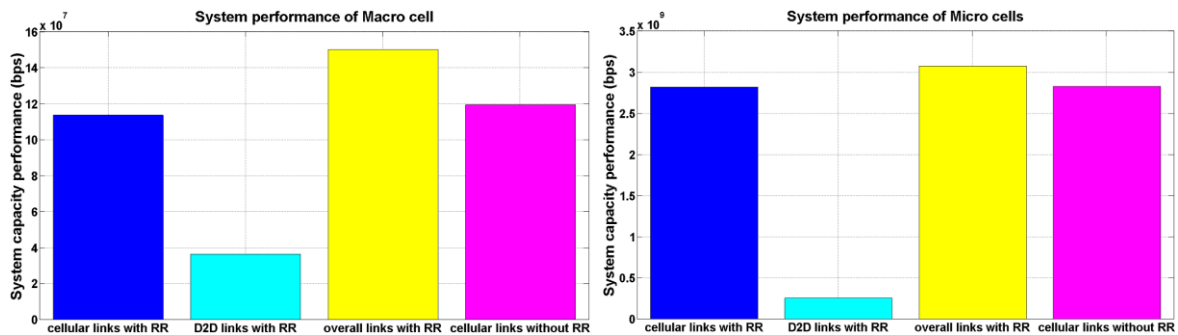


Figure 3-44. System capacity performance of both Macro and Micro cells

System level simulation is performed within TC2 scenario and results are shown in Figure 3-44, where legacy LTE network is used as a baseline for performance comparison. As shown in this figure, resource reuse (RR) of cellular network can enable D2D communication and offload approximately 30% of Macro cell traffic to D2D links by applying our smart signalling scheme. Besides, cellular links performance is very little influenced by underlying D2D communication due to the exploiting smart RRM algorithm. Meanwhile, it can also be seen that D2D links contribute a limited amount of traffic offload in Micro cells, which is approximately 10% in our scenario. This is due to the dense deployment of Micro cells in urban dense scenario. Therefore, D2D has limited performance in UDN deployment.

Conclusions

This TeC enables a centralized D2D RRM scheme with a small cost in additional signalling overhead. Partial CSI and user positions can be exploited to achieve a smart RRM scheme where D2D links can contribute to traffic offloading and have very limited influence on cellular links.

4 Conclusion and future work

This deliverable provides a comprehensive overview on the radio link research conducted in the METIS project. This research covers three main areas, which are considered key aspects for developing a self-contained air interface design for 5G:

1. Flexible air interface
2. Waveforms, coding & modulation and transceiver design
3. Multiple access, MAC and RRM

For each of these areas, a multitude of novel technology components have been proposed. These have been investigated in detail, including an assessment of their individual contribution to the METIS overall goals. Moreover, each of them has been connected to the METIS Horizontal Topics as well as to the test cases defining the METIS scenarios for 5G in Deliverable D1.1 [MET13-D11]. The connections have served as a basis for selecting potential enablers for the METIS overall system concept [MET15-D66]. This has been facilitated by selection of TeCs for the individual HT specific concepts, which continuously triggered additional evaluations throughout the project. In the overall test case evaluations presented in [MET15-D65], those TeCs promising significant improvements towards the METIS overall goals have been considered and evaluated with respect to their performance on system level.

A high level summary of all TeCs considered in each of the above research areas has been presented in chapter 2 of this document, highlighting the key aspects of each TeC and assessing their maturity for practical application and their potential for 5G. Chapter 3 then presented a detailed description of each TeC, describing the concept, emphasizing most important results and drawing an overall conclusion. In the Annex, a profile table is given for each TeC capturing the evaluated KPIs in terms of the GRMs and GRTs as identified in Deliverable D2.1 [MET13-D21] and the achieved gains, together with information on connections to METIS HTs and test cases. Moreover, their individual contribution to the METIS overall goals is specified. The table further contains a judgement on whether the TeC can be considered an evolution of current systems or if it represents a revolution in the sense that it introduces a completely novel system approach. The Annex further contains detailed evaluation results for a number of TeCs, complementing those published in conference papers and Deliverable D2.3 [MET14-D23], as referenced in the main part in chapter 3.

From all the TeC clusters in the radio link research, five of them have been selected as the “most promising” ones, exhibiting highest potential for 5G by providing novel functionalities for an efficient system design that translates to considerable performance gains. Those most promising TeCCs are in the following briefly described and the gains are highlighted. As stated above, a brief summary of all the investigated TeCs can be found in chapter 2.

TeCC#1: Unified air interface for ultra-dense deployments

The cluster provides a concept of an air interface targeting mobile broadband services in ultra dense networks, characterized by high data rate and reduced latency. A harmonized OFDM concept allows using a unified baseband design for a broad range of carrier frequencies going up to millimetre waves. This concept enables a frame structure that is fully scalable over the range of operating frequencies, supports self-backhauling and advanced multi-antenna techniques while facilitating a low cost and energy efficient solution. The concept is designed for the TDD mode, enabling a dynamic partitioning of UL/DL periods to allow for easy adaptation to the highly asymmetric traffic expected in UDN. With the TeCs in this cluster, significant gains towards the METIS overall goals can be achieved.

TeCC#6: Air interface for moving networks

The cluster addresses aspects of an air interface design for moving networks and provides radio link enablers for future V2V services. The objective is to improve the robustness of

mobile communication links and enable services with strict reliability requirements such as road safety applications. A framework for URC enables opportunistic access of ultra-reliable services by introducing an “availability” giving information on the presence or absence of reliability in the communication link. The framework is well complemented by a model toolbox for modelling and predicting the reliability of the radio link. Novel schemes for channel prediction and channel estimation in V2V scenarios have been introduced, providing significant performance improvement compared to state of the art solutions. Finally, an ad-hoc MAC for moving networks has been proposed, allowing for V2V communication in the case of limited or no network connectivity.

TeCC#8: Filtered and filterbank based multi-carrier

Multi-carrier waveforms with filtering provide spectral containment of the transmit signals. By this means, they enable partitioning the spectrum into independent sub-bands that can be individually configured according to the needs of a service, while requiring relaxed synchronization between different sub-band signals. Thus, these waveforms can be considered key enablers for a flexible PHY air interface design. Two candidates are under investigation, namely FBMC and UPMC; the latter is also known under the term UF-OFDM. While FBMC comes with some changes of the overall signal structure, UPMC tries to maintain the signal structure used in OFDM. However, both approaches target the same scenarios and can realize gains from the flexible configuration of the spectrum similarly. For FBMC, several bottleneck problems known from the literature, such as the filter tail length in short burst transmission or the need for guard carriers in MIMO precoding, have been solved in the METIS radio link research, paving the way for an application in 5G systems. OFDM waveforms and its variants are still further pursued for selected 5G scenarios, in particular for UDN mobile broad band (see TeCC#1). A direct comparison between all these candidate waveforms with respect to their performance in mixed service scenarios envisaged for 5G systems has not yet been carried out and is left for future studies.

TeCC#11.1 Non- and quasi-orthogonal multiple access

Novel multiple access schemes allow for overloading the spectrum by multiplexing users in the power and the code domain, resulting in non-orthogonal access, where the number of simultaneously served users is no longer bound to the number of orthogonal resources. This approach enables gains in user and system throughput of up to 50% compared to similar LTE setups using orthogonal user multiplexing, while the number of connected devices can be increased by a factor of 2-3. Candidate schemes are non-orthogonal multiple access (NOMA) and sparse code multiple access (SCMA). Both schemes can be well applied with open and closed loop MIMO schemes, where it could be shown that the MIMO spatial diversity gains could be achieved. If applied in the context of massive machine communication, SCMA can further reduce the signalling through grant-free access procedures.

TeCC#12.1: Contention based massive access

This cluster introduces novel MAC schemes for contention or reservation based access of a large number of devices with low overhead. For reducing signalling overhead, scheduled access is avoided and replaced by alternative access schemes: Coded random access and coded access reservation make use of repeated transmissions following a code pattern to enable resolving collisions. An advanced PHY processing for enhanced MAC uses CDMA codes and compressive sensing techniques at the receiver to resolve potential collisions already on PHY. The two TeCs “coded random access” and “advanced PHY processing” have further been combined, providing a powerful technique for efficient access of a large number of MMC devices. Evaluation shows an increase in the number of supported devices by a factor of up to 10 compared to LTE schemes, while the battery life of the devices may be enhanced significantly.

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Annex A

6 TeC profiles and detailed performance evaluation results

The Annex contains the profile tables for each investigated TeC and provides further evaluation results.

6.1 Unified air interface design for dense deployments

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM1 (MU/C-TP): Full UL/DL flexibility results in increased maximum link direction specific spectral efficiency compared to LTE-A, about 100% improvement compared to LTE UL and about 25% improvement in DL. GRM2 (MAX-#CONN): more than 50 times higher discovery capacity in PHY compared to LTE-A. GRM4 (MAC-LAT): in cmW PHY HARQ RTT of ~1 ms (8x improvement w.r.t LTE-A) -> user plane latency improvement ~5x GRM7 (ECON-B/AU): ~7-40x improved battery life time on PHY level w.r.t. MTC-optimized LTE-A. GRT1 (COST-HW): Reasonable cost is achieved via usage of TDD, simple frame design enabling pipeline processing, low latency, wireless self-backhauling and OFDM based waveform. GRT2 (CFR) & GRT3 (SBW): Harmonized OFDM concept provides full flexibility in terms of used carrier frequency (from < 3GHz up to mmW). Flexibility is provided also with respect to the amount of used bandwidth, dependent on the used carrier frequency and connection type. GRT5 (ENHET): UDN concept feasibility for MMC, D2D, coverage etc.
Baseline for performance comparison	LTE-A Rel.11. TDD.
Update of the results w.r.t D2.3	More detailed studies on harmonized OFDM numerology. Agreement on UDN frame structure, common channel principles explained in more detail. More detailed description and results related to multi-antenna technologies. New conceptual descriptions related to self-backhauling and scalable latency.
Revolution or evolution	Revolution (new physical air interface numerology and design).
Targeted test cases (TC)	TeC has been selected for TC evaluation of the following TCs: TC2. TeC also proposed for TC1, TC3, TC4, TC6, TC7, TC9, TC11.
Impacted HTs	UDN is the main focus. TeCC#1 solutions could also be

	utilized in D2D, MMC (and MN-N) HTs (in UDN context).
System aspects addressed	Main focus in xMBB (some properties may be found useful also for other system aspects)
Cross-connection to other TeC clusters	TeCC#1 – TeC#12.3: PHY and MAC layer are considered as a combined design. TeCC#1 provides physical layer enablers for mmW MAC solutions developed in TeC#12.3.
Contribution to METIS overall goals	
1000x data volume 10-100 user data rate	10—100 higher user data rate via adaptive TDD, wider BW, short latency, beamforming etc.
10-100x number of devices	More than 50 times higher discovery capacity compared to LTE-A.
10x longer battery life	~7-40x (w.r.t. theoretical MTC-optimized LTE-A)
5x E-E reduced latency	~5x user plane latency reduction w.r.t LTE-A. For cmW: PHY RTT ~1ms (8x better compared to LTE-A), TTI ~0.25ms. For mmW even smaller
Energy efficiency and cost	Via TDD & frame design, low latency, wireless self-backhauling, and beamforming.

CP length analysis

The spectral efficiency performance per layer for two channel Power Delay Profiles (PDPs), namely ITU indoor hotspot (InH) NLoS and ITU urban macro (UMa) NLoS are presented correspondingly in Figure 6-1 and Figure 6-2 for several SC spacing values varying from 60 kHz to 600 kHz. Results for ITU urban micro (UMi) NLoS would lie in between InH and UMa. We use 40 dB channel SNR (including error vector magnitude (EVM)) and maximum modulation of 256-QAM with rate 0.9, which gives $SE_{max} = 7.2$ bps/Hz. By comparing the difference of the optimal spectral efficiency values to the spectral efficiency achieved with 1 μ s CP length in Figure 6-1 and Figure 6-2 with 60 kHz SC spacing, we notice that the degradation in SE performance is at maximum about 5-6%. Thus, to overcome the channel delay spread, constant 1 μ s CP length seems to be sufficient for all investigated channel models. Timing alignment (TA) has been used in macro and large cells to compensate the propagation delay. Thus, with TA, we can conclude that 1 μ s CP length should be feasible for channels varying from indoor hotspot to outdoor micro and macro, causing less than 6% overhead with 60 kHz SC spacing.

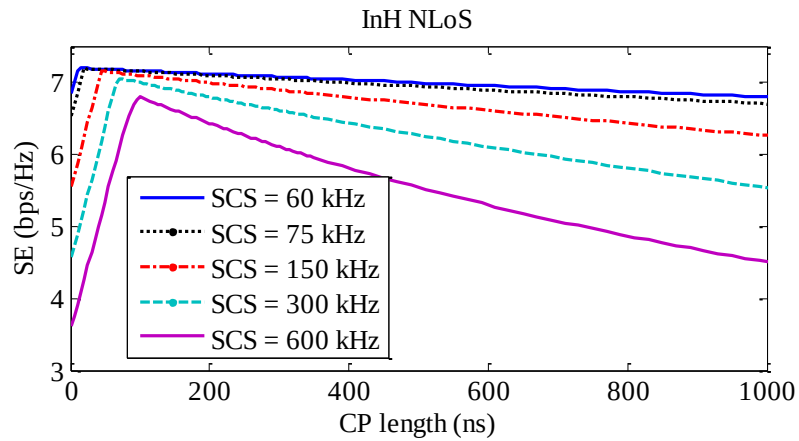


Figure 6-1 Spectral efficiency as a function of CP length for InH

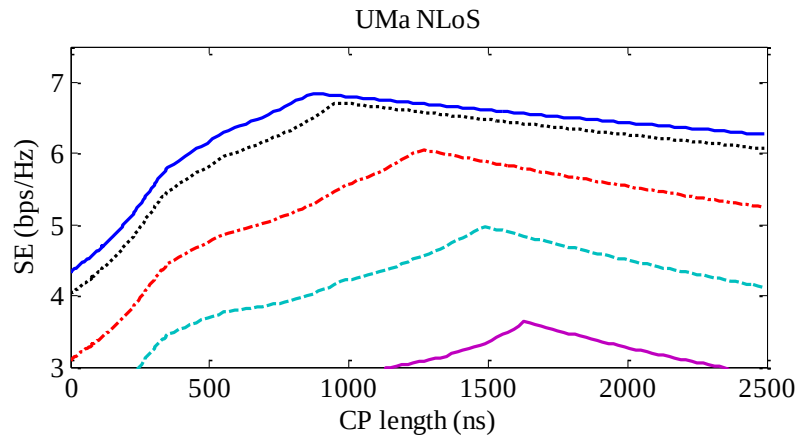


Figure 6-2 Spectral efficiency as a function of CP length for UMa

As a corresponding example for mmW region, we use the 802.11ad conference room channel model [IEEE802.11-10]. This channel has very short delay spread, and as seen in Figure 6-3, the optimal CP length is very small.

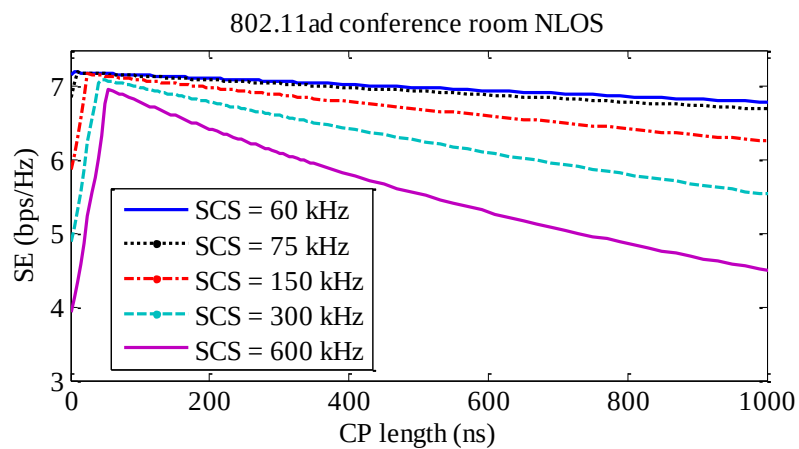


Figure 6-3 Spectral efficiency as a function of CP length for 802.11ad conference room

Phase noise impacts to air interface design in higher frequencies

Phase noise model

Figure 6-4 below is an example of phase noise model used in the study considering phase noise from one side of the radio HW. In reality, in case both the transmitter and the receiver have the same phase noise contribution, the error will be 3 dB higher. Note that the model does not include the low frequency portion of the phase noise contribution, which is called common phase error (CPE) and is assumed to be estimated and compensated together with the channel estimation. As a result, the phase noise model is flat below the phase noise loop bandwidth.

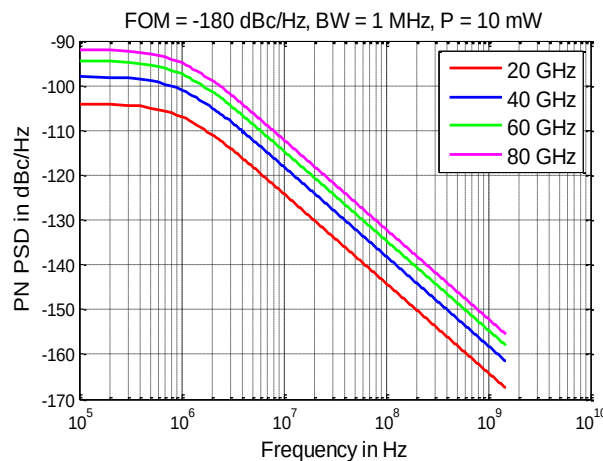


Figure 6-4 Phase noise model for different carrier frequencies and 1 MHz loop bandwidth. The x-axis is the frequency offset relative to the carrier frequency

Phase noise pilots

Phase-noise compensation can be categorized into two main groups: (i) CPE compensation and (ii) full phase-noise compensation. CPE compensation only needs a single phase error estimate for each OFDM symbol, and then de-rotates the entire symbol using that single estimate; often such compensation is partly achieved as a side-effect of regular channel estimation. Full phase-noise compensation, on the other hand, requires tracking of the phase noise during each OFDM symbol. For efficient full phase-noise compensation, we therefore consider having a special phase-noise pilot (PNP) signal that is present over a certain part of the system bandwidth in every OFDM symbol, as illustrated in Figure 6-5. The receiver can use this known signal to estimate the phase noise and then compensate for it. Both phase noise in the transmitter side and the receiver side can be compensated.

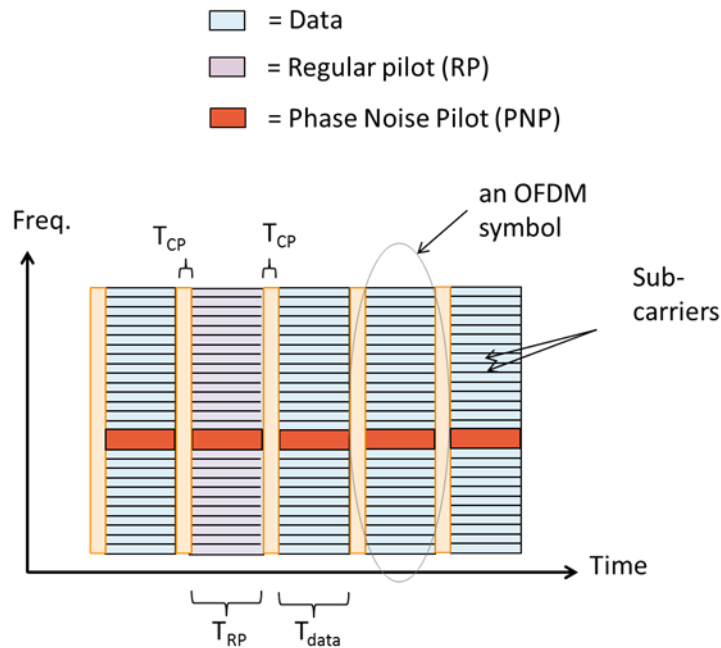


Figure 6-5 Example phase-noise pilots

Channel model

As channels in the simulations, we use measured channels provided kindly by Aalto University and described in [HTWM11], [GHWT14] and [GTHM11]. We would like to simulate 2-stream transmission based on polarization diversity. Unfortunately, the measurements have been performed using only a single polarization direction. We therefore used the following approach to model polarization diversity based on the measurements. The measurements consist of 7x7 antenna elements in both Tx and Rx nodes. We use the following simple trick to create four subarrays at one node (allowing slight overlap between the subarrays) as shown in Figure 6-6, which results in a total of four slightly different channels. We use two of the channels for transmission and the other two with down-scaled power to act as inter-stream interferers. Figure 6-7 shows an example of the four effective channels obtained using this method.

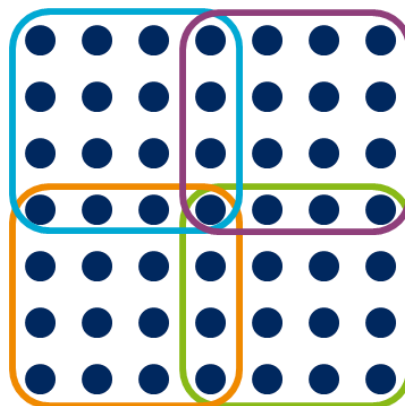


Figure 6-6 Example of how measurements using a single large array can be used to model multiple channels with smaller arrays.

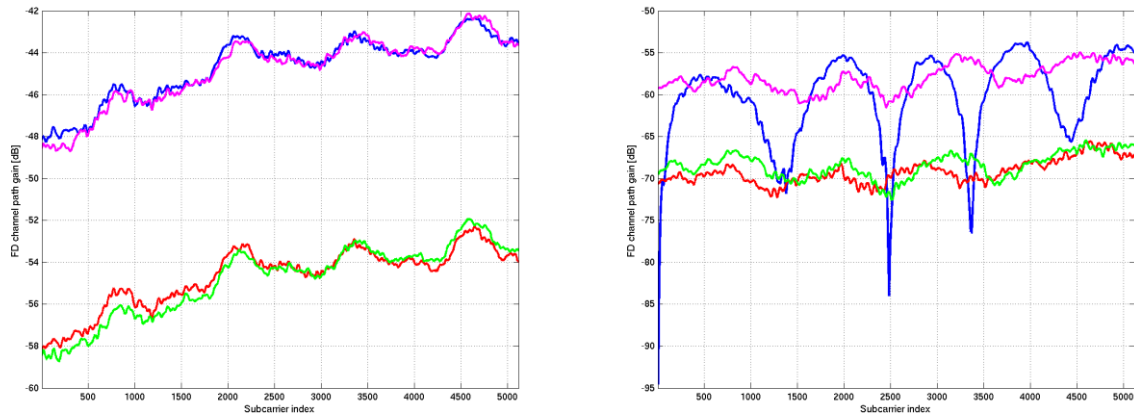


Figure 6-7 Example of the four effective channels for LOS (left) and NLOS (right) channels with interference channels down-scaled with 10 dB, measurement position c5 [GHWT14]

Simulation settings

Table below lists the most important settings used in the performance evaluations.

Table 6-1 Simulation Settings

Number of ANs	1
Number of Active UEs	1
Channel Model	Measured data, LOS, NLOS
Carrier Frequency	~63 GHz (measurements)
Bandwidth	1843.2 MHz
Multiplexing Scheme	OFDM and DFTS-OFDM
FFT Size	8192 (see Section 2.1 for motivation)
Number of Subcarriers	5120
Subcarrier Spacing	360 kHz
Cyclic prefix	347.222 ns
Number of OFDM Symbols per Frame	2 (small to keep simulation times short)
Maximum Tx Power per AN	20 mW
Antenna Configuration	Planar square antenna array with elements 0.4λ wavelength apart
Number of Antennas	Access Node: $7 \times 7 = 49$; UE: $4 \times 4 = 16$
Type of Transmitter Beamforming	Effective beamformed channels obtained from pre-processing, see session “ Channel model ”.
Type of Receiver Beamforming (Combining)	Effective beam-formed channels obtained from pre-processing, see session “ Channel model ”. In case of 2-stream transmission, interference rejection combining (IRC) between the two streams (i.e. inversion of one 2×2 matrix per subcarrier) is on top of that performed
Channel Estimation	Ideal
Phase Noise Compensation (when Enabled)	Based on estimated phase noise, with 5% pilots
Thermal Noise Level	-174 dBm/Hz
Noise Figure	6 dB

Simulation results

The figures below showed that with good oscillators of FOM -180 dBc/Hz and BW 1 MHz, the performance loss due to phase noise is negligible for 16QAM and 2-stream MIMO, for both

modulation schemes evaluated (OFDM and DFT-spread OFDM), cf. Figure 6-8 to Figure 6-10. For 64 QAM, the loss is larger but the performance can be improved with the simple phase noise compensation, cf. Figure 6-11 and Figure 6-12. With lower cost oscillators e.g. of FOM - 180 dBc/Hz and 0.2 MHz BW, the performance loss due to phase noise can also be significantly decreased with phase noise compensation.

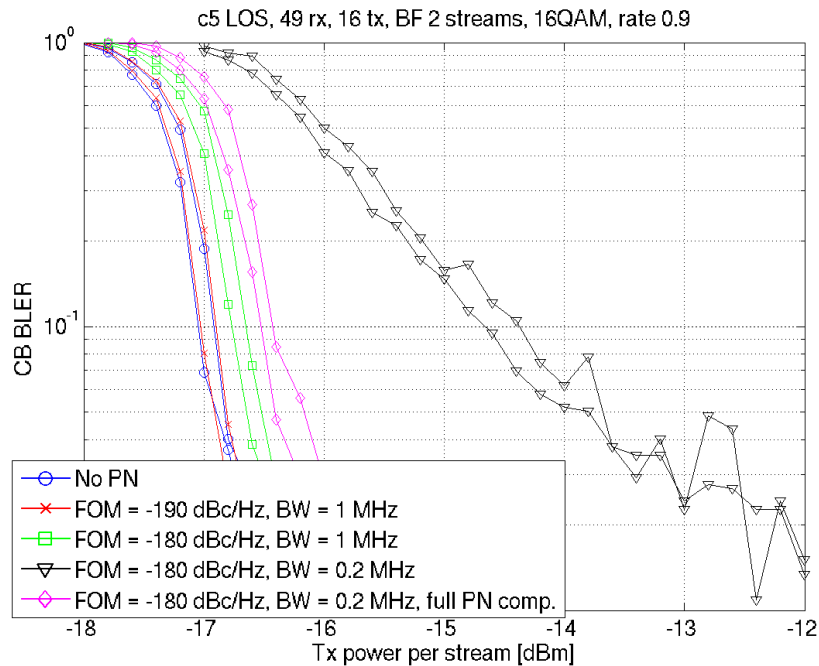


Figure 6-8 LOS, variant FOM, variant BW, 16QAM, OFDM, BLER

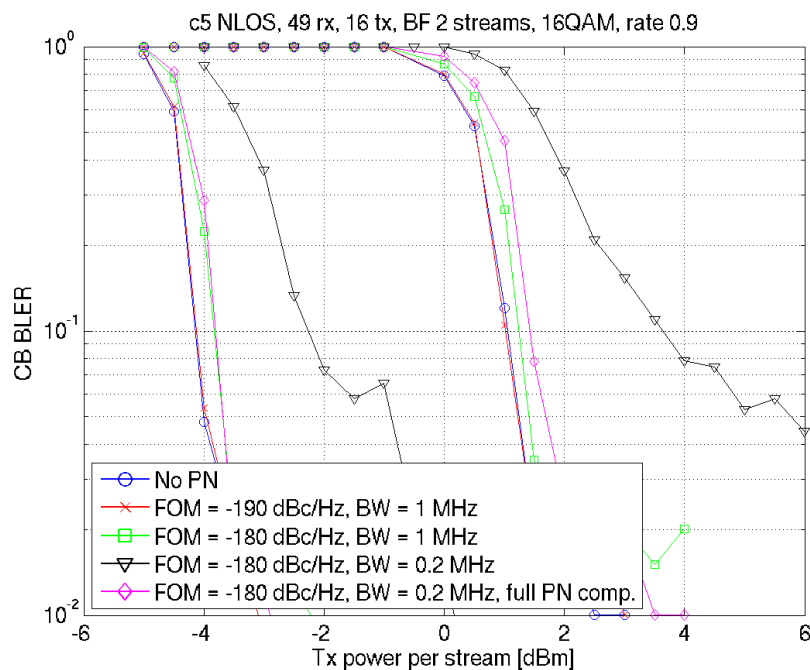


Figure 6-9 NLOS, variant FOM, variant BW, 16QAM, OFDM, BLER

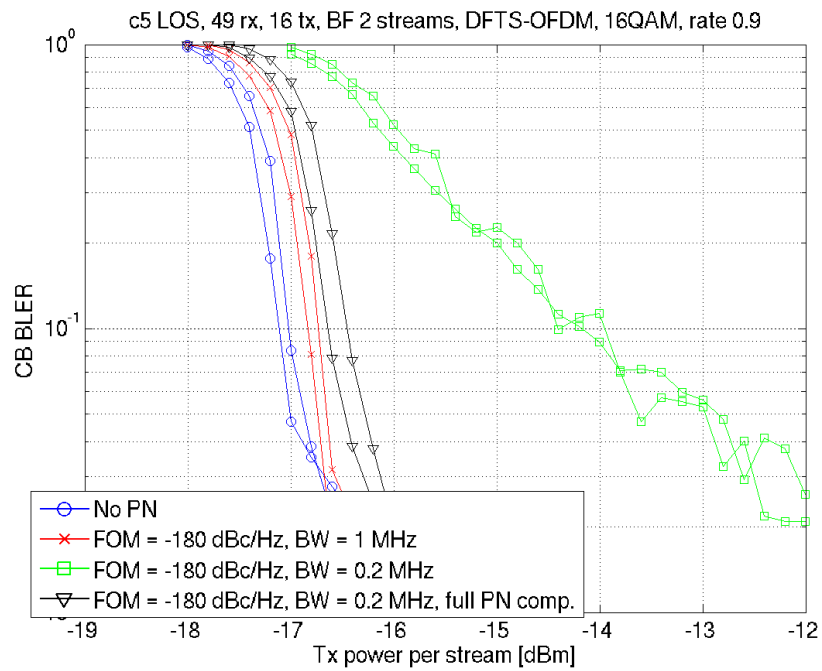


Figure 6-10 LOS, variant BW, 16QAM, DFTS-OFDM, BLER

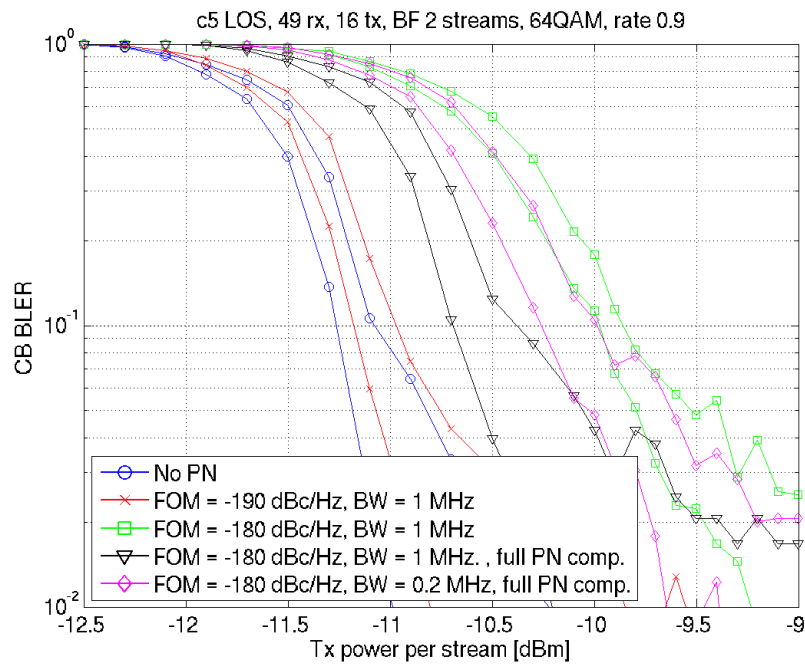


Figure 6-11 LOS, variant FOM, variant BW, 64QAM, OFDM, BLER

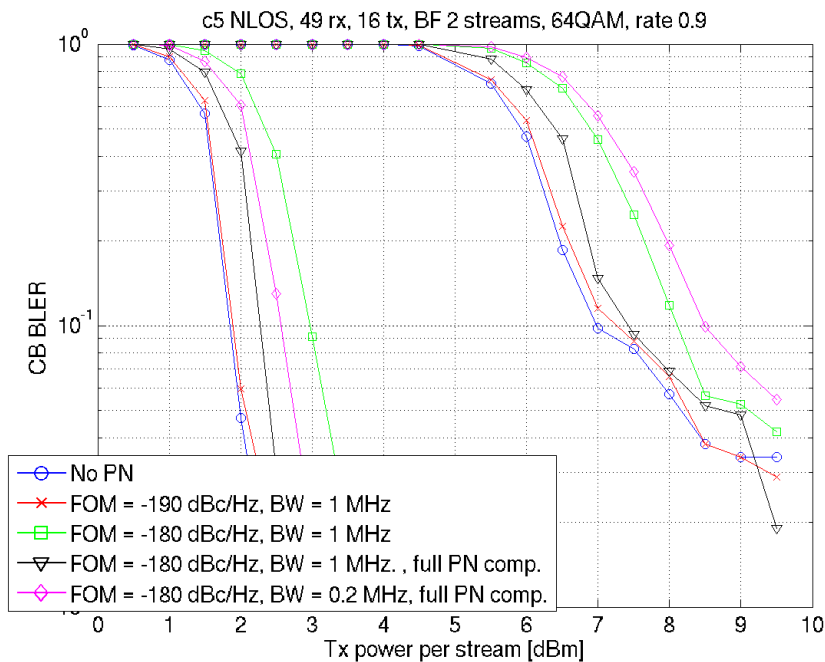


Figure 6-12 NLOS, variant FOM, variant BW, 64QAM, OFDM, BLER

User plane latency

User plane latency [ITU-R M.2134] is defined as the one-way transmit time between an SDU packet being available at the IP layer in the user terminal/base station and the availability of this packet (protocol data unit, PDU) at IP layer in the base station/user terminal. User plane packet delay includes delay introduced by associated protocols and control signalling assuming the user terminal is in the active state.

The UL user plane latency characterization for METIS UDN concept and comparison to LTE-A [3GPP-36.912] is presented in Table 6-2 for cmW. For DL similar analysis was presented in Section 3.1

Table 6-2 User plane latency analysis for TDD & FDD in UL with 10% BLER

Delay component	METIS UDN (cmW)	LTE-A TDD	LTE-A FDD
eNB Processing	0.25 ms	1 ms	1.5 ms
Frame Alignment	0.125 ms	1.1-5 ms	
TTI Duration	0.25 ms	1 ms	1 ms
UE Processing	0.375 ms	1.5 ms	1.5 ms
HARQ Re-transmission (10% x HARQ RTT of 1ms)	0.1 ms	1.0-1.16 ms	0.8 ms
Total Delay	~1 ms	~6-10 ms	~5 ms

Battery life time

A simplified theoretical analysis about battery life time presented in [MET14-D23] showed that ~7-40 times lower energy consumption can be achieved with the proposed frame structure when compared to MTC (machine type communication) optimized LTE-A. These results are presented also in the Figure 6-13 below, including also illustration of the TC11 specific requirement point with two reception intervals, namely one reception per 1 s and 10 s.

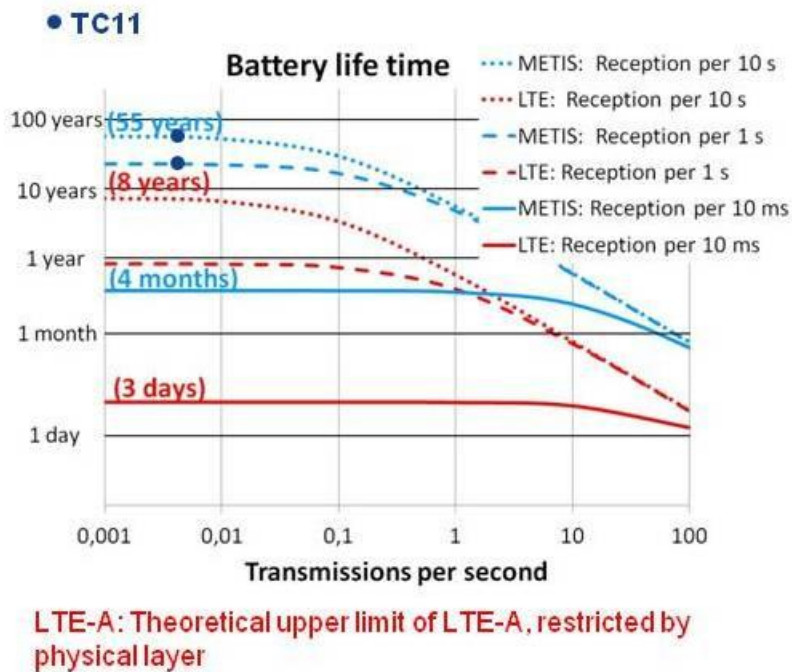


Figure 6-13 Battery life time comparison between MTC optimized LTE and METIS UDN frame structure

Multi-antenna techniques

We evaluated the high gain beamforming effect in terms of channel capacity to see how much SNR can be achieved which is a combined effect of link budget and interference isolation. The floor plan used in the simulation is a small open office with 9 access nodes and 9 active UEs as shown in the left figure below. The used simulation parameters are: Downlink, full buffer, no wireless backhauling, frequency reuse one, one access node serves one UE, number of TX antenna elements 64 with semi-spherical array, number of RX antenna elements 16 with squared array, carrier frequency 60 GHz, transmission power 20e-3 W, thermal noise -174 dBm, noise figure 7 dB, reflection loss of walls 5.6 dB, ray tracing model with maximum number of reflections considered per ray 2, and number of drops 100.

Simulation result showed that a channel capacity of 14-22 Gbits/s per link can be reached with one layer and MRC receiver, as shown in the right figure below. Higher channel capacity is expected to be achieved with e.g. 2 layers by using polarization diversity. As mentioned in chapter 2, the capacities are raw bits, overheads and the transmitter and receiver impairments need to be taken into account to get the actual data throughput which will be for further studies.

Small open office 9x9x3 m³

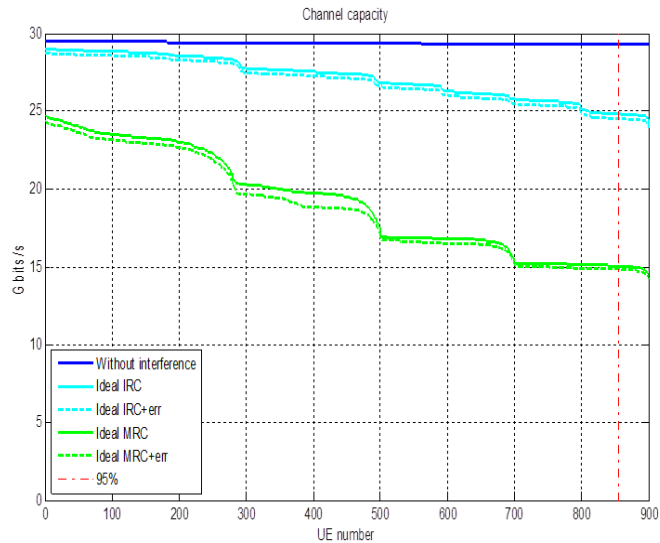
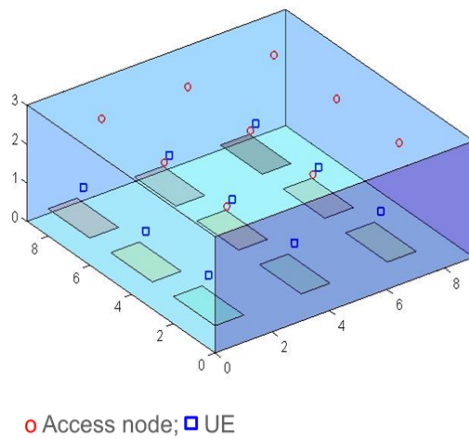


Figure 6-14 NLOS, variant FOM, variant BW, 64QAM, OFDM, BLER

Scalable latency

In TDD systems, the room for traffic adaptation between UL and DL (i.e. the maximum DL/UL ratio) can be limited by the UL coverage, especially by UL control channel, since sufficient amount of time resources are needed for UEs located in the cell edge. This limits the amount of time resources available for DL (and DL throughput accordingly). Problem would be especially noticeable in WA environment, where the eNB transmit power is significantly higher compared to the UE's transmit power. One approach to solve the above described would be to aggregate data via (TDD) METIS UDN link and control via frequency division duplex (FDD) WA link respectively. In the following, we consider an arrangement where a device has possibility to utilize at least two means to transmit information, namely 1) (FDD) WA link type (e.g. LTE, for e.g. UL control) and 2) (TDD) METIS UDN link type. The considered scenario is illustrated in Figure 6-15. The total minimum achievable RTT would depend on both (TDD) METIS UDN and (FDD) WA latencies. As the distance and propagation between UE and the WA FDD eNB increases, also the achievable WA FDD control latency increases (to take into account increased propagation loss, which requires longer TTI). This leads to the concept of scalable latency, with the aim that it should be possible to select the trade-off between RTT and coverage in UE specific manner. Constant and optimal hybrid automatic repeat request (HARQ) RTTs should be supported for different UEs that may have different distances (or path loss) to the (FDD) WA eNB and that may also function on different METIS UDN frequency areas (e.g. cmW, mmW) with different subframe lengths.

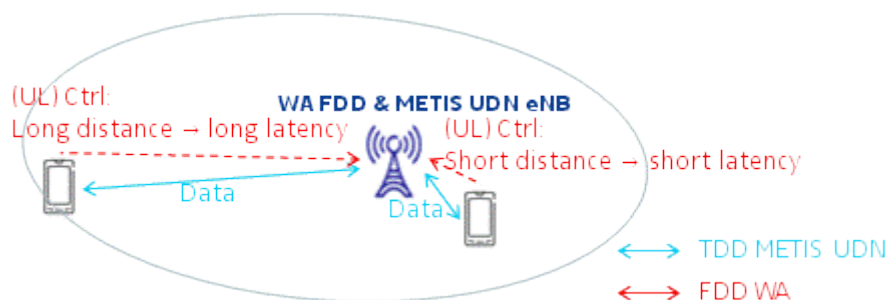


Figure 6-15 Intra-cell TDD-FDD aggregation

Definition of only one static TTI length would evidently set the hard limit for the RTT and for the minimum amount of HARQ processes. Since TTI length plays a significant role in RTT with

TDD-FDD aggregation, scalable TTI length is required to enable scalability in latency. The latency improvement provided by proposed METIS UDN air interface cannot be supported by current LTE control plane.

In order to provide a clean solution and e.g. to support simple and clean HARQ realizations, it would be beneficial to align the TTI timings of TDD and FDD link types. (TDD) METIS UDN subframe lengths could be defined as full multiples of (FDD) WA symbol length. Lengths of the METIS UDN subframes could still be different for different frequency areas, as long as they are aligned to the (FDD) WA symbols. Also, the TTI lengths of these link types could be aligned, such that the M TDD TTIs fit exactly to the same time window as L FDD TTIs ($L < M$).

In addition to aligning the TTI timings between TDD and FDD links, the proposed solution approach includes also providing a common physical resource pool with flexible design and support for variable TTI lengths. This common pool can be shared between different TDD and FDD layers by providing e.g. parallel resources (in frequency and/or code dimension etc.) for different supported TTI lengths (feasible e.g. for HARQ-ACKs) and/or supporting also TDM between channel resources of different TTI lengths (TDM feasible e.g. for CSI). The defined common resource pool supports also flexible switching from one supported TTI length to another. eNB determines the used (e.g. UL control) transmission scheme and the TTI length in device specific manner. This decision may be done based on various aspects, such as UL coverage, control channel type (e.g. CSI, HARQ-ACK) frequency range UE is operating on and required latency. One option could be to utilize LTE as the (FDD) WA link type. In this approach, certain separate physical resources e.g. on top of LTE PUCCH format 2 could be assigned for this arrangement. Alternatively, clean slate approach could be utilized.

Figure 6-16 illustrates an example scheme, where FDD WA link is utilized for transmitting UL control information when the device is not in UDN environment (in UDN environment, the TDD-based METIS UDN air interface would be utilized also for UL control). LTE is used as a basis for the FDD WA arrangement and certain PRBs from PUCCH format 2 are reserved for common resource pool. In this example, parallel resources are utilized to provide support for different TTI lengths and the suitable TTI length is selected based e.g. on the UL coverage.

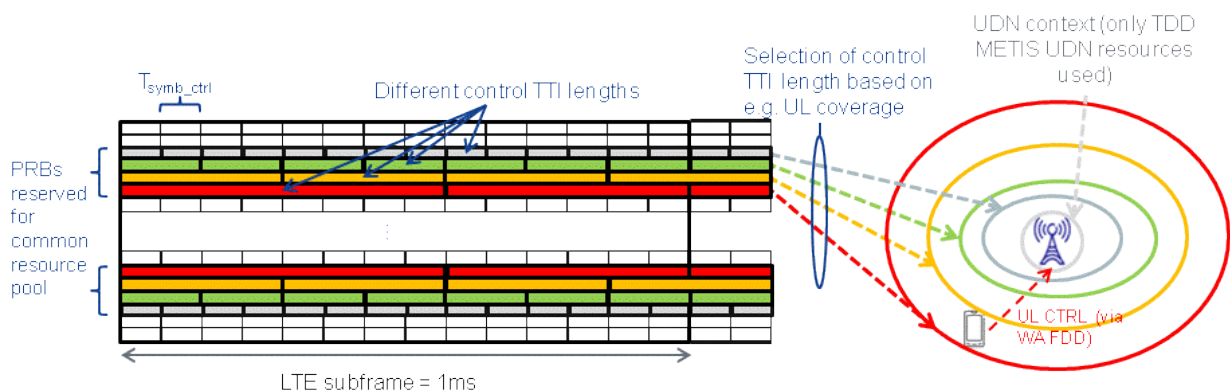


Figure 6-16 of (UL) control TTI length (e.g. based on UL coverage)

TTI length in the common resource pool is selectable in UE specific manner and can be different e.g. for devices with different distance to the (FDD) WA eNB, leading the TTI length to be a trade-off between latency requirements and UL performance (especially coverage). Different TTI lengths may also be assigned for devices functioning on different frequency areas with different subframe lengths. Figure 6-17 illustrates this with an example of the proposed arrangement, where (FDD) WA link type utilized for UL control has been assigned to be LTE ($T_{\text{symbol_WA}} = 71.4 \mu\text{s}$) and (TDD) METIS UDN link types (for DL data) are utilized on both cmW and mmW. Like described before, the TTI timings between different link types should be aligned, and in this example the subframe length of METIS UDN subframes are defined to be 4 and 1 times of $T_{\text{symbol_WA}}$ for cmW and mmW respectively. Two different TTI

lengths are supported by the common resource pool embedded to FDD WA resources. As seen from Figure 6-17, the proposed arrangement enables UE specific clean and flexible HARQ timings with constant RTT and optimal number of HARQ processes. The minimum length for control TTI setting the hard limit for the RTT can be adjusted flexibly with control symbol length resolution.

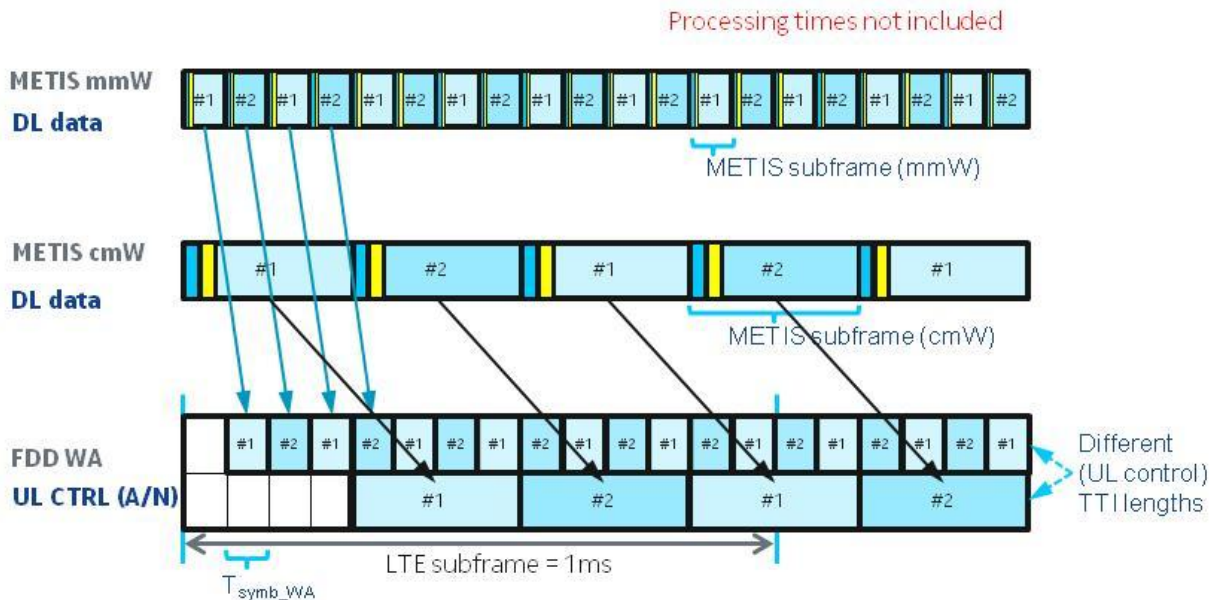


Figure 6-17 Optimal HARQ RTTs for UEs functioning on different frequency ranges

In addition to intra-cell TDD-FDD aggregation (data via METIS UDN link and (UL) control via FDD WA link), the proposed aggregation scheme could also be utilized e.g. for fast backhaul links between small cell and WA base stations (subjected to rapid traffic profile changes) and in Hetnet scenarios.

6.2 Optimized signalling for low-cost MMC devices

6.2.1 MMC Type D2D Links

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM 1 (Throughput) – Does not increase throughput, but instead ensures zero outage for the link with link adaptation. GRM 2 (Supported users) – More than one connection link per resource
Baseline for performance comparison	It has been implemented over simplified LTE Rel.11 physical layer.
Update of the results w.r.t D2.3	Rate bounds that enable Zero-outage link adaption links when interfered by multiple MT-D2D connection with fixed power and rate.
Revolution or evolution	Revolution, as it requires a system redesign to realize the TeC, to add the MTC-D2D functionality.

Targeted test cases (TC)	TeC has not been selected, but could be useful for the following TC(s): 11 due to the focus on MMC (low rate low power transmission devices).
Impacted HTs	TeC has been selected for the HT system concept: D2D, MMC, URC
System aspects addressed	massive MTC & ultra-reliable MTC
Cross-connection to other TeC clusters	Tec#12.1 due to a possible connection with the MMC component.
Contribution to METIS overall goals	
10-100x number of devices	It allows reducing the amount of signalling (CSI-T) required to perform link adaptation, therefore allowing the more efficient scaling of signalling resources with the number of users.
10x longer battery life	Indirectly. MTD can connect to the cellular network via MT-D2D, therefore the MTD will potentially require less power.
Energy efficiency and cost	Indirectly, since MTD can connect to the cellular network via MT-D2D, therefore the MTD will potentially require less power.

Further evaluation results:

In Figure 6-18 is depicted the achievable zero-outage rate in the downlink for the Joint Decoding setting, with a variable number of devices associated with U1. It can be seen that when the MTD rate is very low, a large number of simultaneous MTDs can be supported without a significant impact on the supported zero-outage downlink rate.

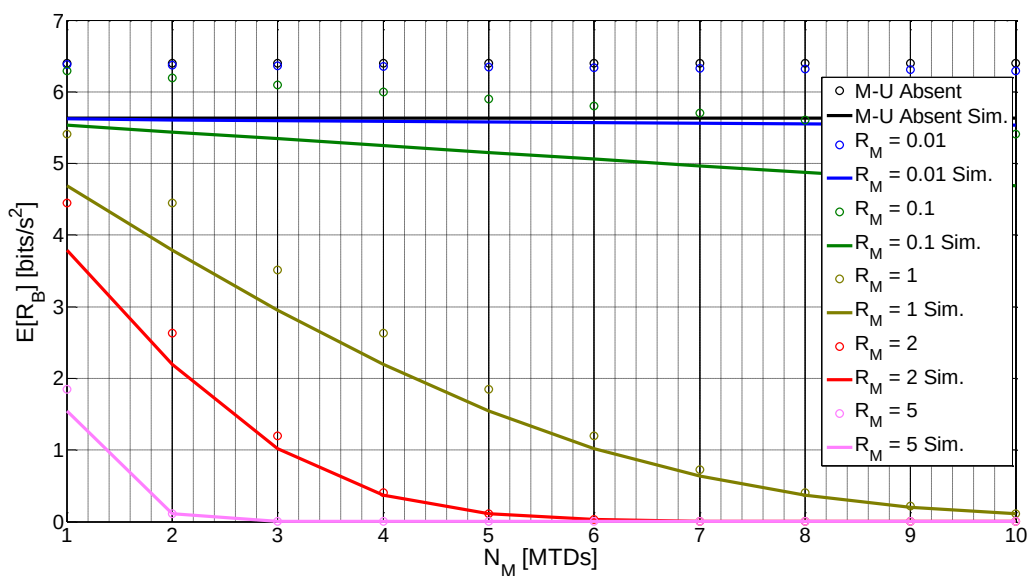


Figure 6-18 GRM1# Throughput and GRM2# Supported Users – Evaluation of the evaluated zero-outage upper bound in the Joint Decoding setting, when a variable number of MTDs are associated with the cellular user U1 [PP+14].

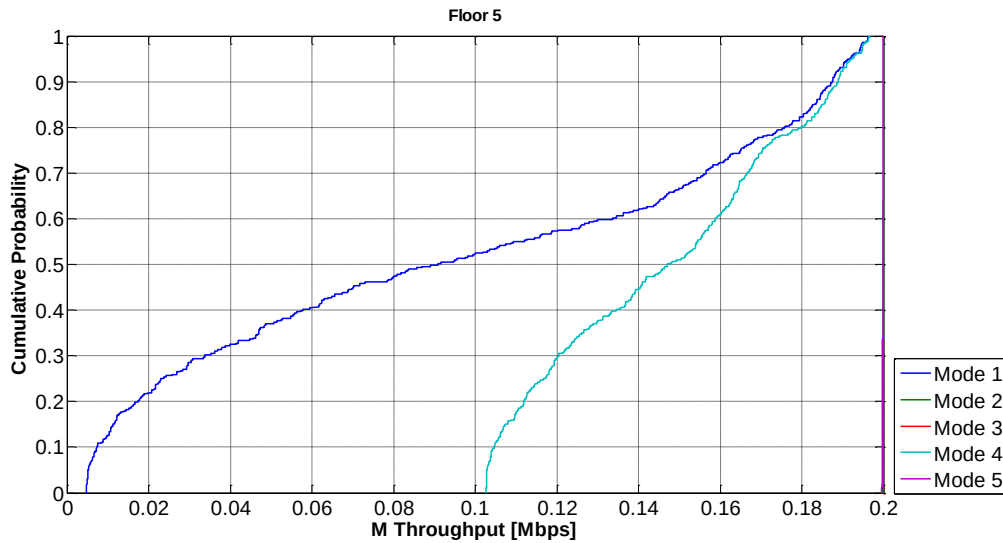


Figure 6-19 – GRM1#Throughput - CDF of the throughput from system level simulation results for link between MTD and cellular device U

The developed zero-outage bound for the Single User Decoding (SUD) setting, was tested using the 3GPP LTE Physical Layer downlink receiver in the METIS TC2. In this scenario the block length is finite and therefore the zero-outage concept no longer holds, but instead it is possible to guarantee link outage probabilities. There were defined five modes to serve as comparison, where Mode 1 denotes the legacy system, Mode 2 SUD based on SINR of B-U1 link, Mode 3 SUD using the zero-outage rule, Mode 4 SUD using SNR of B-U1 link and Mode 5 combining Mode 2 and 3. In Figure 6-19 is shown the CDF of the throughput of the M1-U1 link, where while recalling that the MTD transmits with a fixed rate it can be seen that only Mode 3 and Mode 5 allow to meet the maximum rate the M1-U1 link was designed to. In Figure 3 is shown a correspondent plot but now for the link B-U1. As observed in Figure 6-19, both Mode 3 and Mode 5 have meet the throughput target for the M1-U1 link, while achieving very figures for the throughput in the B-U1 link. This difference in throughput is explained by the amount of information used to select the rate in the B-U1 link. While in Mode 3, only the rate of the MTD transmission is known at B, i.e. there is no CSIT at B from the M1-U1 link, while in Mode 5 it is assumed that CSIT from M1-U1 is available at B.

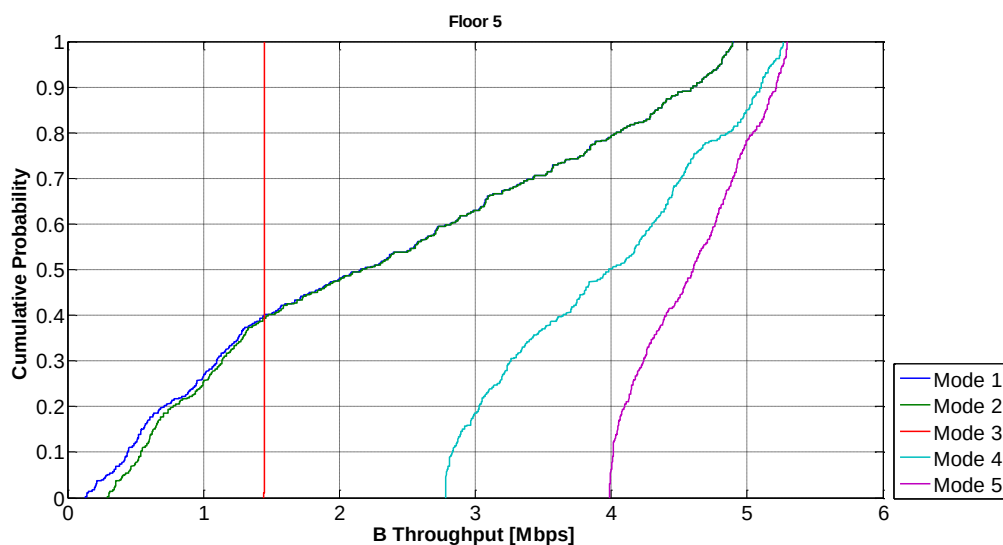


Figure 6-20 - GRM1#Throughput - CDF of the throughput from system level simulation results for link between Base Station and cellular device U

6.2.2 Quasi orthogonal random access

Further details can be found in [NSW14].

6.2.3 Hybrid access scheme for reduced signalling overhead

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM1: Cell throughput (+40%) GRM4: Latency (+38%) GRM5: PER (-20%)
Baseline for performance comparison	Slotted ALOHA for LTE
Update of the results w.r.t D2.3	New TeCC not introduced in D2.3
Revolution or evolution	Evolution as it only adds an extra M2M Access Manager
Targeted test cases (TC)	TC2: Dense Urban Information Society TC11: Massive deployment of sensors and actuators
Impacted HTs	MMC
System aspects addressed	massive MTC
Cross-connection to other TeC clusters	N/A
Contribution to METIS overall goals	
10-100x number of devices	Efficient channel access allows more devices to be connected to the network

6.3 RF Architecture for Dynamic Spectrum Access

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	High-IF architecture is evaluated against GRT2 (Carrier frequency range – CFR) and GRT3 (supported bandwidth – SBW). Prototyped RF front end [ASV+14] demonstrates CFR from 470MHz to 790MHz with system SBW < 80MHz.
Baseline for performance comparison	LTE Rel.11 based Homodyne Architecture
Update of the results w.r.t D2.3	Improvements in out-of-band linearity and noise figure(NF)
Revolution or evolution	Since it brings significant changes to the front end it can be considered as revolution



Targeted test cases (TC)	Can be considered for TC11, if the sensors are supposed to utilize legacy networks with licensed and unlicensed bands
Impacted HTs	TeC has not been selected for the MMC system concept: (can be considered for TC11, if the sensors are supposed to utilize legacy networks with licensed and unlicensed bands)
Cross-connection to other TeC clusters	TeC is driving technology component from TeC#11.2.2: MA using Cognitive radio.
Contribution to METIS overall goals	
Energy efficiency and cost	With an Integrated CMOS high IF front end , the number of external elements can be reduced significantly enabling cost-effective devices.

6.4 Multiple air interface management

6.4.1 Multiple Interface Management in MT-HetNets

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	<u>GRT5: Enabler for Network Heterogeneity (ENHET).</u> GRT5 gains are up to 10-15 dB when considering different IEEE802.11ac modes and switching on mm-wave UWB transmissions <u>GRT2: Carrier frequency range (CFR).</u> The mm-wave UWB system initially designed for SRD applications in UWB bands, allows radio coverage extensions to Hot-Spot into MT-HetNet architectures. <u>GRM5: MAC-packet error rate, reliability and integrity (MAC-PER/REL/INTG).</u> The GLB metric performs the AI selection in choosing the minimum MCM in a given propagation situation which is a quantitative metric of GRM5. MCM provides relative degradations for a targeted PER related to each AI/TMs. The MCM difference between different AI/TMs when selecting the minimum MCM provides gains in dB of AI/TM switching (Table 6-3 and Figure 6-24. Gains are deduced from an initial mode assess <u>GRM7: Energy consumption per bit / area unit (ECON-B/AU).</u> The GLB metric is a MT-LA metric derived from link budget assessments, hence minimizing energy consumption per bit..
Baseline for performance comparison	A reference system is given depending on available air interfaces at the base station and terminal levels in multi-RAT scenarios. IEEE802.11n/ac/ad standards have been considered for Wi-Fi extensions to TC1, TC2 and TC3. The UWB ECMA-368 standard transposed at 60 GHz has been integrated to extend data rates.

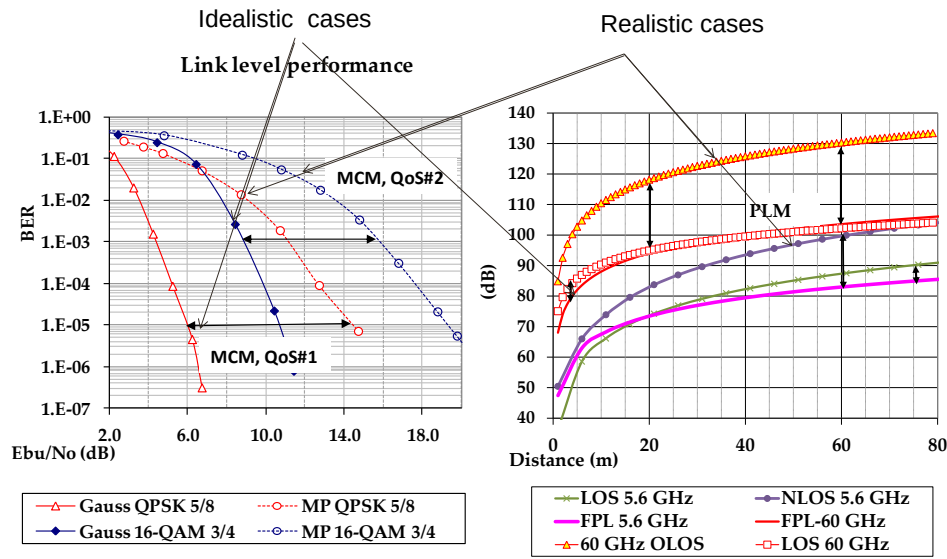


Update of the results w.r.t D2.3	Novel results focused on 5Ghz and 60 GHz switching for hot spot deployments. A new metric has been designed to integrate the time resource allocation of random access systems in MT-LA mechanisms.
Revolution or evolution	The Tec4.1 is an evolution of LA techniques using green criteria to compare independent AIs and TMs.
Targeted test cases (TC)	TC1 : “Virtual Reality Office with mm-wave”, considering evolved ECMA-368 and IEEE802.11 and TM with NLOS/LOS transitions to assess GLB metric gains TC2 and TC3 Dense Urban Information Society” and “Shopping Mall” where IEEE802.11ac/ad Hot Spot extensions are evaluated for small cell deployment in cellular networks
Impacted HTs	UDN, URC
System aspects addressed	Multi-RAT systems
Cross-connection to other TeC clusters	The most relevant connection is to TeCC#1, where different schemes related to unified air interfaces may be evaluated using MT-LA metrics and dynamically select the most outstanding transmission mode and the RF band..
Contribution to METIS overall goals	
1000x data volume	The air interface selection in multi-mode terminals allow a spectrum management which increase the global network throughput and limiting in the same time transmit power requirement to ensure targeted radio coverage.
10-100x number of devices	The GLB metric (α , β sub-metrics) implemented in multi-RAT architectures may increase the number of devices in selecting different air interfaces in several parallel P2P communications in a local hot spot zone. Gain assessments may linearly increase with the number of devices depending on the number involved air interface and transmission modes able to deliver the service, in accordance with the available PHY channels
5x E-E reduced latency	Applicable. Not evaluated yet
Energy efficiency and cost	Energy Efficiency is evaluated using the GLB metric numerical values when considering AI/TM switching involved by LOS/NLOS transitions and spatial variations of the environment. Starting from an initial state, the modification of the AI/TM is quantified using the α -metric which measures related degradations of each interface in a real environment. The β -metric provides the radio coverage of each interface and adapts the excess power to guaranty QoS regarding required power levels (including interference measurements) and targeted radio coverage. Other metrics explicitly considering spectrum management and deployment costs should be deduced from the GLB metric.

Mathematical and detailed description of the GLB metric

In the deliverable D2.2 and [SUM13], a CQI metric devoted to MIM (Multiple Interface Management) has been introduced in order to select, in multi-mode devices and base stations, the air interface and associated transmission mode guaranteeing

- Targeted radio coverage
- The data rate and the QoS of the service
- A dynamic transmit power level control minimization



$$\alpha\text{-metric} = \text{MCM} + \text{PLM}$$

Figure 6-21 the α -sub-metric definition

This metric, denoted Green Link Budget (GLB), is composed of two sub-metrics α and β :

- 1) The α -metric evaluates the *relative* degradations (i.e. the Multipath Channel Margin (MCM)) associated with the air interface/transmission mode to transmit data with a targeted data rate and QoS and the *relative* propagation path-loss degradations involved by the test case of the transmission medium (i.e. the Path-Loss Margin (PLM)). *Relative* values are normalized with idealistic cases represented at the system level by an AWGN channel (MCM) and the free space path-loss level regarding the transmission medium (PLM).

The algebraic expression of MCM and PLM are given by :

$$\begin{aligned} \alpha &= S_M - S_0 + PL_{MFS}(d, fc) - PL_{FS}(d, fc) \\ \alpha &= \underbrace{S_M - S_0}_{\text{MCM}} + \underbrace{ARP_{MFS}(d, fc) - ARP_{FS}(d, fc)}_{\text{PLM}} \quad (6-1) \\ \alpha &= \alpha_{\text{MCM}} + \alpha_{\text{PLM}} \end{aligned}$$

where S_M is the multipath power sensitivity in a multipath context translating the *required power level* to transmit data with a typical BER and data rate D , S_0 is the power sensitivity under AWGN channel related to the same transmission mode and data rate. $PL_{MFS}(d, fc)$ and $PL_{FS}(d, fc)$ are the multipath and free space path-loss models respectively. PLM is illustrated on the Figure 6-21 with the free path-loss attenuation and multipath-path-loss attenuation given at 5 GHz and at 60 GHz.

The air interface and transmission mode which is selected is the mode minimizing degradations quantified by the α - metric.

- 2) The β -metric is a metric dedicated to real-time power control adjustment where the available received power is adjusted to the required power level (i.e. the multipath power sensitivity) necessary to guarantee the QoS and desired radio coverage.

$$\beta = ARP(d) - S_M$$

$$\beta = EIRP + G_R - \alpha - S_0 - PL_{FS}(d, fc) \quad (6-2)$$

A representation of the $\{\alpha, \beta\}$ metric is given below in considering a dedicated test case where 2 Access Points (AP) are able to be connected to a single UE. The AP selection may use the $\{\alpha, \beta\}$ metric values to select the AP and the associated AI and TM to establish the communication. In the example, LOS and NLOS propagation conditions and the distance $d(\text{AP-UE})$ differ with the considered AP. The GLB $\{\alpha, \beta\}$ metric is evaluated for each TM/AI associated to AP1 and AP2 respectively supplying $\{\alpha_1, \beta_1\}$ and $\{\alpha_2, \beta_2\}$ metric variations.

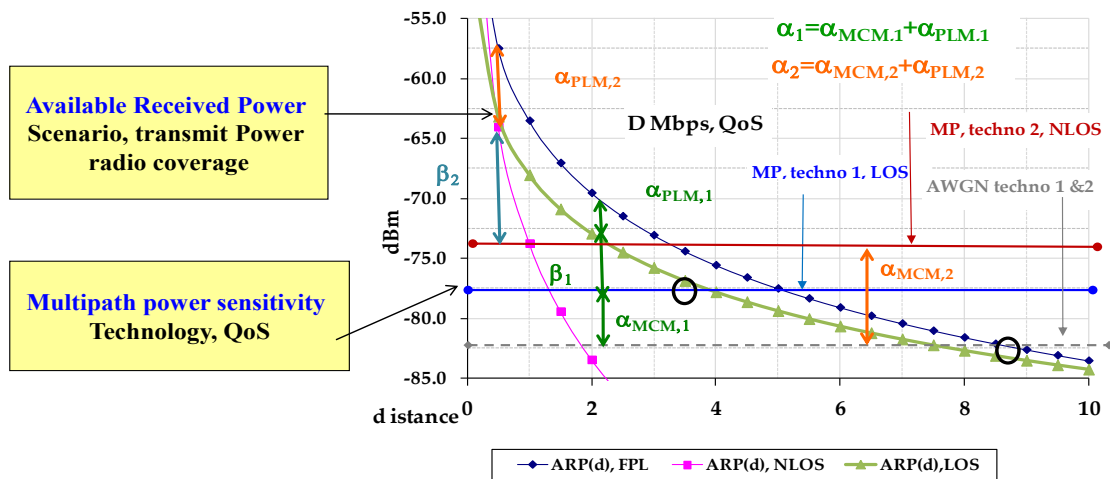


Figure 6-22 The GLB metric representation using 2 AI/TM modes and link budget parameters

The AI and TM selection resort from a α -metric minimization and the β -metric maximization for a given AI and Transmit Mode (TM). An illustration is given for AP selection considering a 60 GHz LOS ECMA-368 transmission and IEEE802.11n TM modes (MIMO STBC and SISO) operating at 5 GHz in a NLOS configuration. These modes are extracted from the Table 6-3.

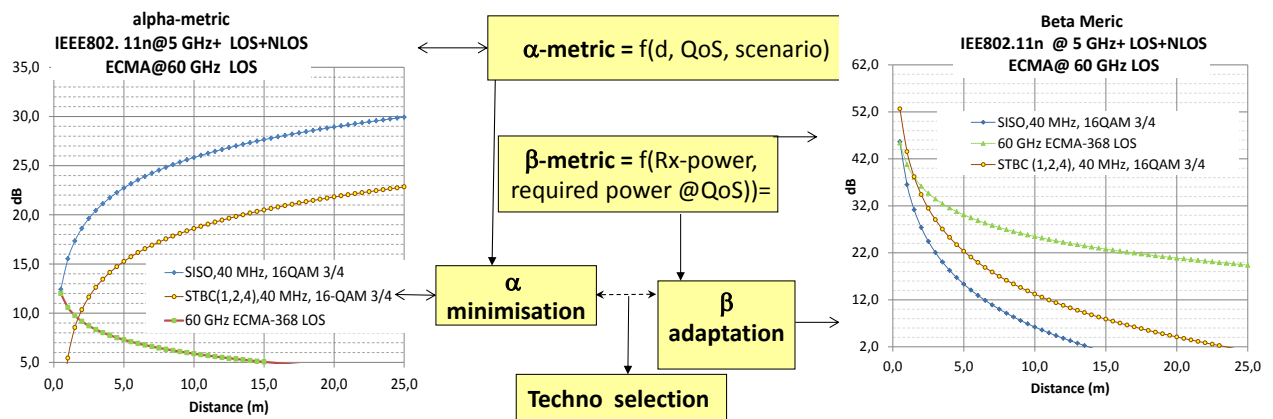


Figure 6-23: The AI/TM selection result for 2 separate AP using the GLB metric

The GLB metric performance for Wi-Fi and mm-wave UWB applied to UDN TC#2

the GLB metric is evaluated in the context of Hot spots and large indoor areas in UDN scenarios. The transmitter and the receiver are able to deliver 2 independent technologies: the IEEE802.11ac/n and the UWB-MB-OFDM (the ECMA-368 standard transposed in mm-wave band). The α -metric representation highlights gains potentially obtained when the most green technology is selected. Link budget and Multipath Channel Margin (MCM) parameters are reported on the Table 6-3. NF is the Noise Figure, L0 is the cable loss.

Table 6-3 IEEE 802.11ac and 60 GHz ECMA-368 link budget parameters

Data rate target : 80 Mbps	IEEE802. Ac				ECMA-368 Transposed At 60 GHz
MCS_{11n}	4	4	26	10	
Transmission mode	16QAM $\frac{3}{4}$ - Nss=1	16QAM $\frac{3}{4}$ - Nss=1	QPSK $\frac{3}{4}$ Nss=4	QPSK $\frac{3}{4}$ Nss=2	QPSK $\frac{1}{2}$ TDS+FDS
	40 MHz	40 MHz	20 MHz	40 MHz	528 MHz
	SISO	MIMO STBC (1,2,4)	MIMO SDM (4,4,4)	MIMO SDM (2,2,4)	QPSK $\frac{1}{2}$ TDS+FDS
Data rate (Mbps)	81	81	78	81	80
Efficient bandwidth(MHz)	35.9	35.9	19.5	35.9	507.37
SNR (dB)	21.5	16	27.5	20	1.5
NF + L0 (dB)	10+2.5	10+2.5	10+2.5	10+2.5	8 (6.6) +2.5
S (dBm) [AWGN]	-71.9	-71.9	-82.05	-78.8	-82.37
S _M (dBm) [MP]	-57.94	-64.94	-56.44	-68.44	-77.89
MCM (dB)	13.96	3.96	25.6	10.36	4.48

Following that access Point and User Equipment (UE) may support different technologies, 4 different transmission modes associated with the IEEE802.11n/ac standard are envisioned and combined with the 60 GHz MB-UWB-OFDM technology of the ECMA-368 standard to ensure typical UDN distance ranges. MIMO Spatial Division Multiplexing SDM(N_{SS}, N_{TX}, N_{RX}) represents the usual SDM technique considering N_{STS} time spatial streams, N_{TX} and N_{RX} antennas at the transmitter and receiver sides respectively. STBC (N_{SS}, N_{TX}, N_{RX}) refers to Alamouti code implementation with a number of spatial stream ($N_{SS}=1$) lower than the number time-spatial stream ($N_{STS}=2$) (notations in the standard [IEEE802.11ad]). Results are represented on the Figure 6-24.

Making the assumption that the terminal and the BTS are in MIMO SDM(2,2,4) mode at the starting of the communication, it appears that with a distance higher than 5 meters it is interesting to switch on 60-GHz ECMA-368 technology. At 15 meters, the gain related to degradation reduction is set to 15 dB. This gain is translated on the β -metric with potential 5 dB transmission power level reduction.

In order to compare MIMO spatial stream increase versus bandwidth increase for the IEEE802.11.ac technology, 2 transmission modes MCS_{11n,10} and MCS_{11n,26} (Table 6-3) having the same modulation are considered. 4 spatial streams combined with a 20 MHz bandwidth size are compared with 2 spatial streams with a 40 MHz bandwidth size. Results show that it is more suitable to increase the bandwidth size rather than doubling the number of spatial stream to guarantee QoS, radio coverage and reduce the transmit power level. The gain is then set to 15 dB reported on the β -metric leading to radio coverage extension.

Complementary results are detailed in [SUM13]. The next step of the work is to evaluate gains when considering Wi-Fi LTE-A inter-RAT, detail the implementation of the metric in the Control and U plane architectures, and investigate MU gains at macro cell level.

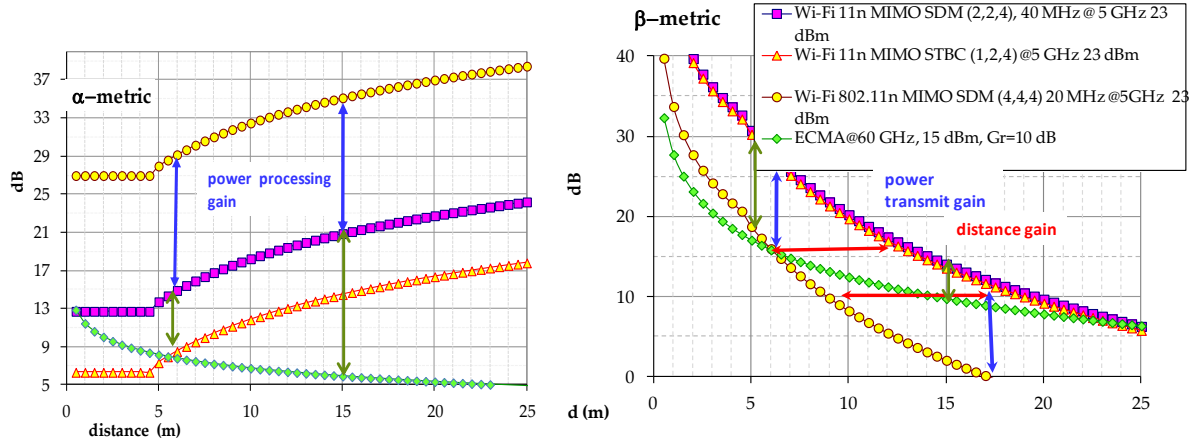


Figure 6-24 The GLB metric results for hot spot deployment

Practical method to evaluate the GLB metric

The practical method to evaluate the α and β -metrics using available parameters extracted from PLCP headers of AIs is a key point of the GLB metric implementation. The method is exposed below and 2 main aspects are addressed:

- How to evaluate the metrics using RSSI and other available signalling protocols and fields?
- How to refresh the decision in time variant and mobile environments?

CQI metrics are usually evaluated using RSSI elements and BER look up tables supplying the correspondence between the QoS target and the required Signal power to Noise ratio (SNR). The method is similar in providing, for each AI and TM, evolved look up tables and LOS/NLOS radio link identification. The problem is then to identify LOS/NLOS propagation conditions deduced from Channel estimation process and deduce from this CSI, PLM and MCM parameters.

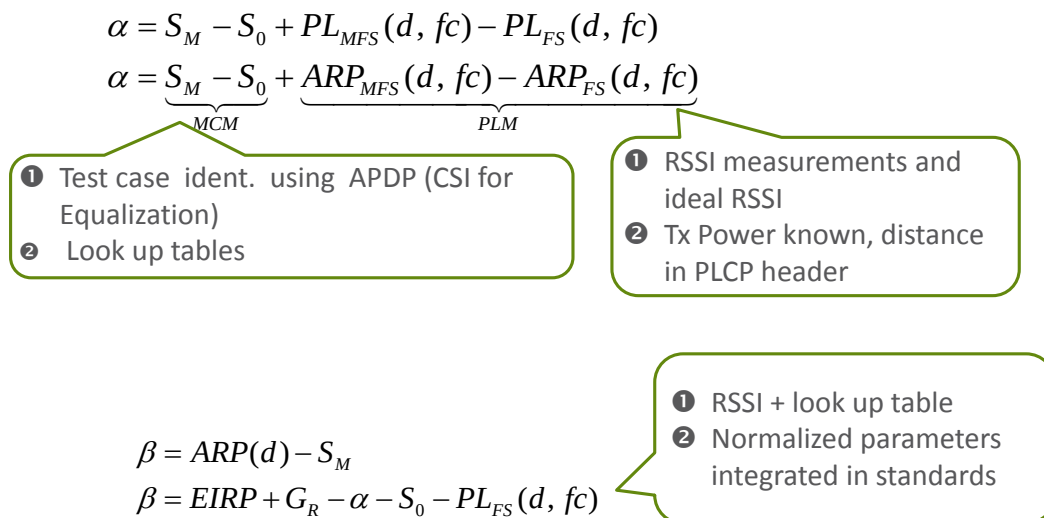


Figure 6-25 Practical method to evaluate MCM and PLM parameters

MCM estimation

The MCM assessment resorts from :

- The propagation scenario identification using the Average Power Delay Profile (APDP) of the propagation channel derived from the equalization process and normalized propagation selectivity parameters allowing a LOS/NLOS discrimination.
- The Rice factor assessment based on power distribution estimation within the time window of the frame as proposed in DAA mechanisms [LA07], refining the first decision using normalized propagation parameters
- LL Look up tables pre-established for each scenario and available AI (terminal and BS) with LOS/NLOS separation.

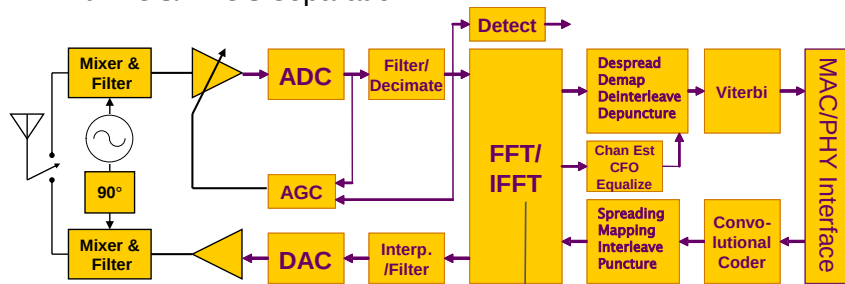


Figure from J. Lansford, "Detect and Avoid for MB-OFDM", IEEE802.19-05.034r0, Sept 2005

$$O_{DF} = \frac{Bw}{B_{c-y}}$$

$$B_{nc-y} = \frac{Bc-y}{\Delta f_{FFT}}$$

Mm-wave UWB system : MG-ECMA-368						
60GHz MP channels	LOS		NLOS/OLOS		Severe NLOS	
Bandwidth	528 MHz	1056 MHz	528 MHz	1056 MHz	528 MHz	1056MHz
ODS= T_{SYM}/S_{DS}	132.38	74.39	45.02	38.71	26.25	25.09
B _{nc} -0.9	9	7	2	2	1	1
B _{nc} -0.5	28	38	15	14	9	3
ODF-0.9	13	36	54	126	122	252
ODF-0.5	4	6	8	17	14	81

Figure 6-26 The Normalized selectivity parameters to classify propagation conditions

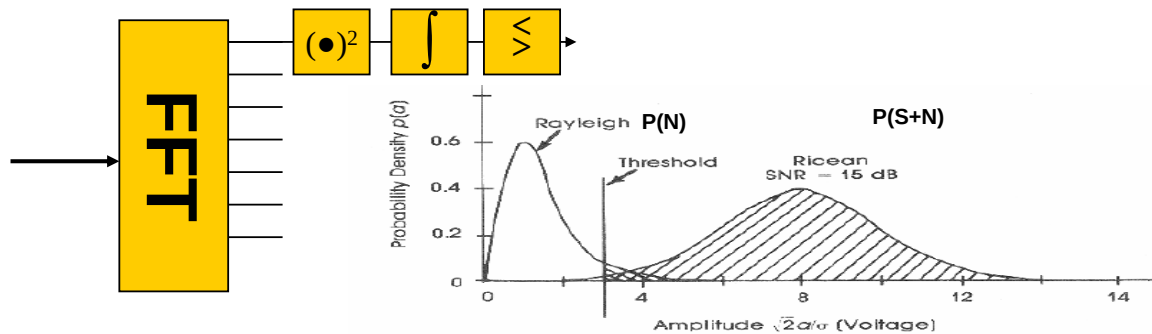


Figure 6-27 The Rice factor estimation to confirm LOS/NLOS conditions

PLM estimation

The PLM estimation is deduced from RSSI measurements and calculation of the ideal RSSI by considering Free Space path Loss model in the theoretical equation of the RSSI.. Measured SNR and ideal SNR shall be used as well.

RF Multi-Band processing for HetNet applied to Wi-Fi and mm-wave UWB

The proposed RF MB processing is based on scalable aggregated sub-channels owing to backward compatibility between UWB and Wi-Fi systems. IEEE802.11n and IEEE802.11ac standards use an elementary sub-channel size multiple of 5 MHz to build several transmission channel sizes {20, 40, 80, 160 MHz} IEEE802.15.3c and IEEE802.11ad implement a 2160 MHz transmission channel size multiple of 540 MHz which results from 27 aggregated @20 MHz channels. A 540 MHz channel size integrates a 528 MHz sub-channel associated with the UWB ECMA-368 standard channelization whilst allowing channel scalability with 60 GHz and Wi-Fi standards.

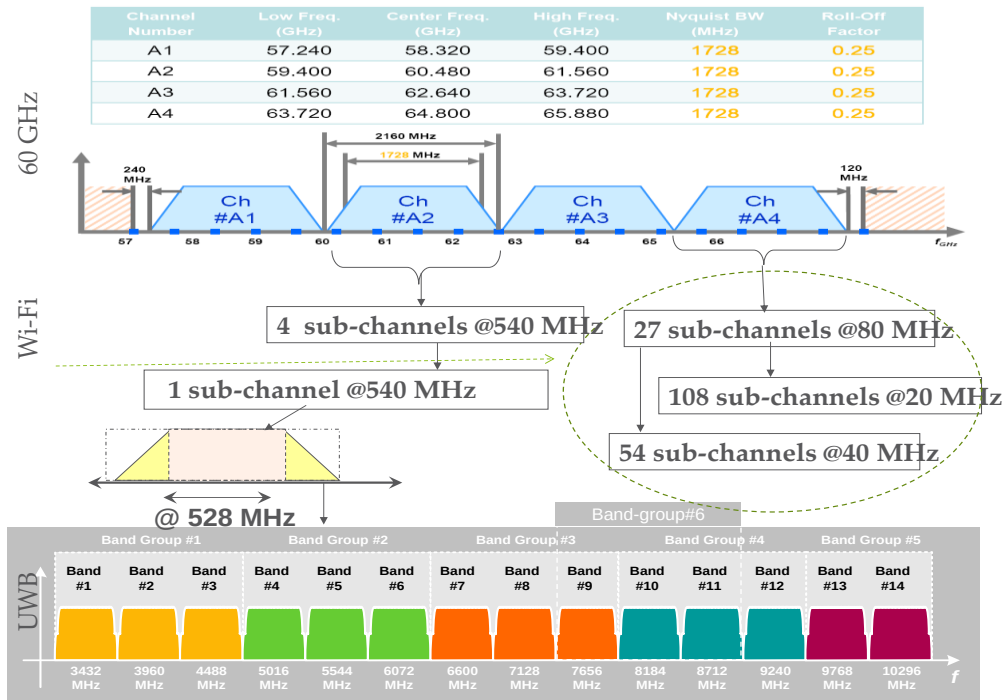


Figure 6-28 Multi-Band processing and RF band allocation in multi-RAT scenarios

6.4.2 Software Configurable Air Interface

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	This is a framework component to enable flexible air interface. Please refer to the different TeC's for the achieved gains.
Baseline for performance comparison	Baseline LTE Rel.11
Update of the results w.r.t D2.3	Provided a case study of adaptation of some SoftAI components to address the MMC HT and estimation of potential gain. (See T4.2 TeC#16 of [MET14-D43])
Revolution or evolution	Revolution: a system redesign required to realize this TeC
Targeted test cases (TC)	This TeC is a framework component. As such please refer to other TeC's for targeted test cases.
Impacted HTs	All HTs

System aspects addressed	xMBB (extreme mobile broadband) massive MTC ultra-reliable MTC
Cross-connection to other TeC clusters	SoftAI can be the framework for realizing various TeCCs. For example a link between this TeC and TeCC#1 UDN can be made such that the UDN configuration can be enabled/disabled.
Contribution to METIS overall goals	
1000x data volume	SoftAI addresses the goal of higher data volume (e.g. via configurable waveforms, multiple access schemes and protocols)
10-100 user data rate	SoftAI addresses the goal of higher user data rate (e.g. via configurable waveforms and modulation and coding)
10-100x number of devices	SoftAI addresses the goal of massive connectivity (e.g. via configurable multiple access schemes)
10x longer battery life	SoftAI addresses the goal of longer battery life (e.g. via configurable frame structures, multiple access schemes and modulation and coding schemes)
5x E-E reduced latency	SoftAI addresses the goal of latency reduction (e.g. via configurable frame structures and multiple access schemes)
Energy efficiency and cost	SoftAI addresses the goal of energy efficiency and cost (e.g. via configurable frame structures, multiple access schemes and modulation and coding schemes)

6.5 Advanced signalling concepts

Aspects of advanced signalling concepts for non-orthogonal multiple access (NOMA) are discussed in section 3.11.1. Since the former was the only TeC in this TeC cluster, it has been closed during the project.

6.6 Air interface for moving networks

6.6.1 Framework for URC

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM5: MAC-PER/REL/INTG Target reliability of 99.999% can be achieved if accurate availability estimation is available. In case of subpar availability estimation, the target reliability is achieved at the expense of availability.
Baseline for performance comparison	LTE Rel.11.
Update of the results w.r.t D2.3	Performance evaluation of the proposed framework by means of system level simulations and analysis of results.
Revolution or evolution	The framework for URC could be introduced in future release of LTE as an evolution since it has been defined in a universal manner.
Targeted test cases (TC)	TeC has not been selected, but could be useful for TC8 and TC12, since both test cases encompass dependable services with strict end-to-end latency and reliability constraints.
Impacted HTs	TeC has been selected for the HT system concept of Moving Networks as an enabler for the provision of reliability in the context of clusters MN-M and MN-V. The TeC is as well relevant for URC.
System aspects addressed	The Tec addresses xMBB and U-MTC services as an enabler to provide dependability to high-demand services and applications.
Cross-connection to other TeC clusters	There is an important connection to TeC#6.2 "Modelling and Predicting the Reliability of a Link", as this is fundamental for the computation of the availability indication that is included in the framework.
Contribution to METIS overall goals	
5x E-E reduced latency	The TeC aims at improving the reliability in the case of dependable services such as V2X. In this sense, the TeC ensures the E-E latency as required by the application.

6.6.2 Modelling and predicting the reliability of a link

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM5 (reliability) GRM3 (Availability)
Baseline for performance comparison	No baseline for comparison
Update of the results w.r.t D2.3	D2.3 only outlines method for AI computation for non-repairable systems only. The present work is a further extension along with evaluation results.
Revolution or evolution	Revolution since new computation methods need to be integrated either in the terminal or the network. Moreover, the applications/services should also be configured to specify their URC requirements accordingly.
Targeted test cases (TC)	NA (Not evaluated on any specific test case since it is more likely a general concept)
Impacted HTs	TeC has been selected for the HT system concept: URC, MN, D2D
System aspects addressed	ultra-reliable MTC
Cross-connection to other TeC clusters	NA
Contribution to METIS overall goals	
10-100x number of devices	Indirectly, since enabling reliable applications will bring new range of domains such as automotive and industrial under 3GPP.
5x E-E reduced latency	Will enable safety critical V2V communication over D2D links, which will reduce latency
Energy efficiency and cost	Indirectly, since V2V over D2D links can be more energy efficient compared to V2V over traditional cellular links.

Further evaluation results:

The transmission of a Mobile UE, moving in space & time is characterized by

- Negative effects (Failure factors) such as Channel degradation, Interference from other UE's, Service Blockage due to Multiple Access etc.
- Positive effects (Repair factors) such as Channel gain, Link adaptation via Tx power control & Adaptive MCS

For reliability analysis, processes with continuous time and discrete state space are important. Hence, in order to proceed any further, we assume that we are dealing with fixed state spaces. This also makes sense since the AI is computed for each URC transmission instance and the continuous time behaviour over this time (t) can be discretized into regular interval states. (For example, at time t, RTL is available with (AI=1), at time t+1, the RTL degrades with a failure rate (λ_1) into state 2 (AI<1)). Moreover there should also exist a termination condition (for the present computation) so that the next URC transmission can be scheduled.

Failure rate $\lambda(t)$ and repair rate $\mu(t)$

Failure rate $\lambda(t)$ is the limit, if it exists, of the conditional probability that the failure occurs within the time interval $(t, t + \delta t]$, to δt when $\delta t \rightarrow 0$, given that the RTL was available at $t = 0$, and did not fail in the interval $(0, t]$. It is the frequency at which RTL becomes unavailable (RTL=0)

Repair rate $\mu(t)$ is similar to $\lambda(t)$ and gives the conditional probability per unit of time that the restoration of the RTL ends between t and $t + \delta t$ provided that it was not finished over $[0, t]$. Table 6-4 lists the available methods for reliability analysis of regenerative systems. A non-regenerative process (highlighted in **Green**) with arbitrary $\lambda(t)$ and $\mu(t)$ describes a URC communication the most accurately but it involves AI computation using partial differential equations with a very high complexity. Hence, we downscale the assumptions and try to evaluate the URC using a semi Markov process (**Orange**) with arbitrary failure and constant repair rates. The failure rates can be derived from the respective life distributions of the channel (the n-dimensional distribution function). The repair rates are dependent on the link adaptation and can be determined with empirical measurements

Table 6-4: List of Stochastic processes for regenerative systems

Stochastic Process	Can be used for	Background	Difficulty
Renewal Process	Arbitrary $\lambda(t)$ & negligible $\mu(t)$, new after repair	Renewal Theory	Medium
Alternating Renewal Process	Arbitrary $\lambda(t)$ & $\mu(t)$, new after repair	Renewal Theory	Medium
Markov Process	Constant $\lambda(t)$ & $\mu(t)$ during stay times in every state	Differential or Integral Equations	Low
Semi Markov Process	Arbitrary $\lambda(t)$ & Constant $\mu(t)$	Integral Equations	Medium
Semi Regenerative Process	Constant $\lambda(t)$ & Arbitrary $\mu(t)$	Integral Equations	High
Non Regenerative Process	Arbitrary $\lambda(t)$ & Arbitrary $\mu(t)$	Partial Differential Equations	High to Very High

Evaluation methodology

In order to evaluate the validity of the semi-Markov models and the KPI's as defined (PA and Sojourn times), we try to implement the Availability calculation in the context of wireless communication. For this we use Markov models directly from a higher symbolic language (Mathematica) that provides default toolboxes to evaluate the KPI's. However, for such evaluation to happen, we need to construct the states and the state transition matrix.

States

In order to derive the reliability states, respective life distributions of the failure factors were used in order to compute the system reliability of the channel [RS14]. Three degrading factors were taken into account – large scale path loss represented with an exponential distribution, Fast fading represented by a Rayleigh distribution and shadowing with a log normal distribution. The mean and standard deviations of the considered distributions are.

- 1) Path loss – Exponential – Mean **(1)** – Variance **(1)**
- 2) Fading – Rayleigh – Mean **(2 π)** – Variance **(4(2 – $\frac{\pi}{2}$))**
- 3) Shadowing – Log Normal – Mean **(e^3)** – Variance **($e^6(-1 + e^4)$)**

The system reliability (Figure 6-29) is calculated by considering each of the distribution as a life distribution of an individual component and connecting them in series ($R(t) = R_{pathloss} \wedge R_{fading} \wedge R_{shadowing}$). Mathematica was used in order to compute this system reliability over time. For simplification purposes and the need to have finite state spaces, 100 uniformly distributed Random variables were generated out of this distribution and they serve as the reliability states for our evaluation

State Transition Matrix

The State Transition matrix can be derived from the distribution of failure and repair rates. This was computed using the Hazard Function (also known as the force of mortality). This gives us the failure rate at which the state transitions occur (Figure 6-29). Efforts are presently on to use a probability matrix instead of the hazard functions since in the former case, we will have a complete matrix.

Since no prior information about repair rates were available, we assumed it to be a constant value of 2 spread over irregular intervals depicting the link adaptation process carried on a RTL.

The PA is calculated as the probability of RTL=1 over time t [RPS14]. Since, we depend upon accurate distributions for our evaluation, the time scale and interval can only be defined once we have the mobility model in place. Hence, for a start, we assume time to be dimensionless. The rationale behind taking three different classes of applications is to show how the PA varies over time for each of the service. The services are classified into high medium or low based upon their UP states. An application with high reliability constraints will have very few Up states and conversely a low reliability application can still be able to work in more states. Moreover, such partitioning of RTL reliability into multiple decrement states also enables reliable service composition (METIS) i.e. graceful service degradation

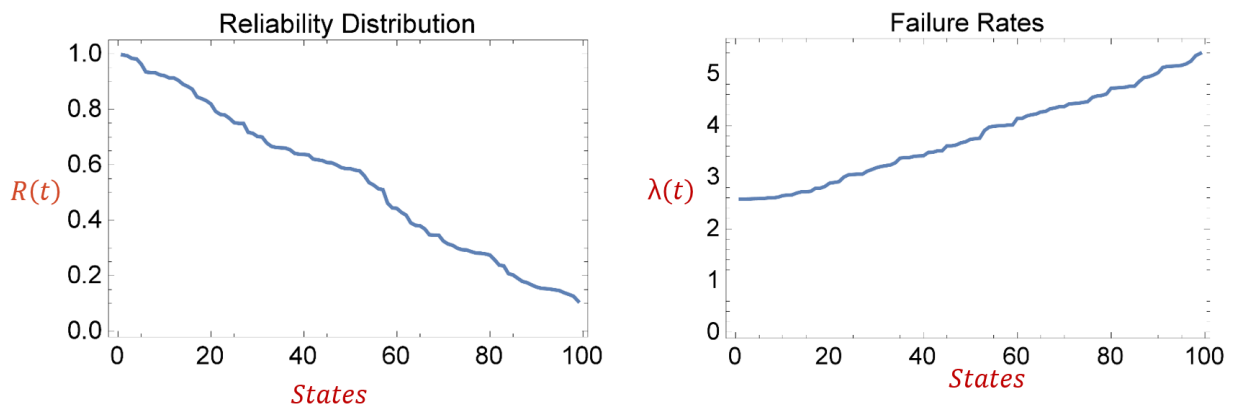


Figure 6-29 : States & State Transition Matrix

6.6.3 Channel estimation for V2V

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	Performance is measured in terms of SNR required to reach a certain mean-squared channel estimation error, which is not a GRM or GRT. The gains are significant, approx. 10-15 dB.
Baseline for performance comparison	Regularized least squares estimator.
Update of the results w.r.t D2.3	More refined estimators have been developed and simulated.
Revolution or evolution	Revolution: since the method (currently) requires preamble signals and is tailored to the direct V2V channel, this constitutes a significant change compared to LTE.
Targeted test cases (TC)	TeC has been selected for TC evaluation of the following TCs: TC12
Impacted HTs	TeC has been selected for the HT system concept: D2D, MN, URC
System aspects addressed	Ultra-reliable MTC
Cross-connection to other TeC clusters	Channel estimation is performed for all practical receivers. Hence, this TeC is related to TeC#6.5, which operates over V2V channels.
Contribution to METIS overall goals	
10-100 user data rate	Indirectly, since V2V links can increase user data rates between vehicles in some scenarios.
5x E-E reduced latency	Will enable direct V2V communication over D2D links, which will reduce latency
Energy efficiency and cost	Indirectly, since V2V over D2D links can be more energy efficient compared to V2V over traditional cellular links.

Further details on research results of TeC#6.3 “Channel Estimation for V2V Links” can be found in [BMS14, BSM14].

6.6.4 Channel prediction

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	At least 10 times longer prediction horizon compared to legacy prediction schemes for a given NMSE requirement.
Baseline for performance comparison	Wiener or Kalman prediction with no predictor antenna.
Update of the results w.r.t D2.3	Attainable NMSE with noisy measurements in a realistic case with measured channels to two antennas on a vehicle.



Revolution or evolution	Revolution: enables closed loop CSIT based schemes also at high vehicular velocities.
Targeted test cases (TC)	TC8, TC2.
Impacted HTs	Focus is on mobile broadband to/from moving networks, but robust backhauling of moving networks is also an enabler for network-assisted ultra-reliable communication between vehicles.
System aspects addressed	Targets enabling solution for advanced backhaul links to fast moving vehicles, which is essential for efficient support of moving small base stations that are well integrated in a heterogeneous small cell system.
Cross-connection to other TeC clusters	TeCC#1 UDN air interface, since the MAC design impacts the required prediction horizon.
Contribution to METIS overall goals	
1000x data volume	Closed loop transmission schemes to fast moving vehicles will enable densification with vehicle mounted small cell base stations that will substantially improve the aggregated capacity in the macro-network, as shown by results in [MET14-D43].
10-100 user data rate	Closed loop transmission schemes enabled also for vehicular users will enable substantially higher data rates for the onboard users.
10-100x number of devices	Closed loop transmission schemes in the backhaul to vehicle mounted moving small base stations will enable group based user mobility, thus potentially substantially lowering the signalling requirements and supporting more users.
10x longer battery life	Closed loop transmission schemes in the backhaul to vehicle mounted moving small base stations will enable substantially lower uplink path loss for the users, thus saving battery on user terminals.
5x E-E reduced latency	Closed loop transmission to vehicles enables QoS-aware resource allocation that can control and shorten the delay.
Energy efficiency and cost	Our evaluations in [MET14-D33] using the predictor antenna scheme with massive MIMO beamforming shows substantial energy savings, at the cost of additional antennas on the vehicle.

System parameters and additional results for the Predictor Antenna scheme

The measurement setting and equipment used for data collection is briefly summarized in Table 6-5 and further details are described in [SGA+12] and also in [JBG+14].

Table 6-5 Experimental conditions and parameters [SGA+12]

Parameter	Description
Geometry	Line-of sight channel with significant multipath components
Carrier frequency	2.68 GHz
Bandwidth	20 MHz
Number of subcarriers	2048
Transmit antenna	Standard base station antennae (KATHREIN 80010541), using +45° polarization, at height 55 m
Receive antennas	Two thin $\lambda/4$ monopole antennas, placed in-line at the centre of a flat metal plate on the vehicle roof.
SNR at receivers	25-27 dB
Measurement interval	200 ms at 45-50 km/h, corresponding to 2.8 m distance.
Number of measurements	3
Bandwidth	20 MHz

One key observation of the NMSE performance in Figure 3-17 is the over-all accuracy: The NMSE is on average better than -13 dB with antenna spacing of one wavelength or larger. This level of CSI quality is useful for coherent multipoint transmission using robust zero-forcing and for various adaptive beamforming schemes. Similar accuracy levels have in [AS14] and [A14] been shown to enable good performance gains (up to 54% as compared to cellular transmission) in coordinated multipoint transmission that uses coherent joint downlink transmission with zero-forcing precoding. In that investigation, three single-antenna base stations were used. The results were obtained for measured 20 MHz wide OFDM channels based on channel sounding measurements.

Another key observation in Figure 3-17 is that the measured prediction accuracy does not decrease with the antenna spacing, up to the highest antenna spacing investigated, three wavelengths. This bodes well for the use of predictor antenna schemes for vehicles at very high vehicle velocities and/or high carrier frequencies: A given required prediction horizon in time would correspond to more wavelengths in space when the velocity or the carrier wavelength is increased. The predictor antenna scheme far outperforms the use of conventional prediction, which is based on the measured channel statistics: A prediction horizon of 3 wavelengths in space is *an order of magnitude* longer than the prediction horizon normally attainable by e.g. Kalman prediction [AS14, A14, A11].

The performance of the predictor antenna scheme in the presence of noisy channel estimates has also been investigated. We have here found interpolation over subcarriers and over space/time samples to be efficient in reducing the effect of noise. Table 6-6 shows an example, where a predictor antenna placed 2 wavelengths in front of the main antenna is used, and the prediction horizon is one wavelength (11 cm in space, corresponding to 8 ms in time at 50 km/h). Measured channels and parameters are as described in Table 6-5 but the data is for a subset of the data used in Figure 3-17. We have added noise to the received measured signals that have SNR 20-27 dB, to also generate artificial measurements with SNRs 15 dB and 5 dB. Using interpolation over time reduces the influence of the noise. Smoothing splines were used for interpolation in these experiments. Other smoothing /interpolation techniques such as Kalman smoothing or Wiener interpolation could be used.

Table 6-6 Channel prediction based on noisy predictor antenna estimates

NMSE when predicting channel of rearward antenna:	No added noise (SNR ~25 dB)	SNR 15 dB	SNR 5 dB
Based on raw pred. antenna estimates	-13.94	-11.55	-5.42
With smoothing spline interpolation over time (1D interpolation)	-14.08	-13.68	-10.82

It was noted in [MET14-D23] that for a short antenna spacing, 0.5 wavelengths or shorter, the prediction accuracy is reduced because of a decorrelation effect due to mutual couplings

between the antennas. As outlined in [JBG+14] and illustrated in [MET14-D23], the antenna coupling and the resulting decorrelation can be reduced by a linear compensation scheme that operates on the antenna signals. This compensation has not been utilized in Figure 3-17. Its use would improve the prediction accuracy for half-wavelength spaced antennas.

6.6.5 Ad-hoc MAC for V2V

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	Channel load at a given packet error rate. The channel load is defined as $G = K/N$, where K is the number of users and N is the number of slots in a frame. Hence, K can be interpreted as GRM2 maximum number of supported connections (MAX#-CONN). The achieved gain in GRMA2: MAX#-CONN is 100% at a packet error rate (GRM5: MAC-PER) of 10^{-3}
Baseline for performance comparison	802.11p MAC (CSMA) and default data rate for traffic safety applications (6 Mbit/s).
Update of the results w.r.t D2.3	This TeC has not been presented in D2.3. Hence the results presented here are novel.
Revolution or evolution	Revolution: there is no pure ad-hoc MAC mode in LTE and it is unlikely that this will be introduced in the future.
Targeted test cases (TC)	TeC has been selected for TC evaluation of the following TCs: TC12
Impacted HTs	D2D, MMC, MN, URC
System aspects addressed	Ultra-reliable MTC
Cross-connection to other TeC clusters	There are connections to V2V channel estimation (TeC#6.3), since the TeC#6.5 need accurate channel estimation for V2V links when performing interference cancellation.
Contribution to METIS overall goals	
10-100 user data rate	V2V links can increase user data rates between vehicles in some scenarios.
5x E-E reduced latency	Will enable direct V2V communication over D2D links, which will reduce latency.
Energy efficiency and cost	V2V over D2D links can be more energy efficient compared to V2V over traditional cellular links.

Further details on research results of TeC#6.5 “Ad-hoc MAC for V2V” can be found in [JBG+14].

6.7 Faster than Nyquist (FTN)

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM1: Increase link spectral eff., only link level
Baseline for performance comparison	Comparison with baseline: Nyquist in AWGN channel
Update of the results w.r.t D2.3	Realization scheme for FTN-FBMC/OQAM
Revolution or evolution	Evolution, since flexible FBMC (including FMT) architectures can be designed including FTN features.
Targeted test cases (TC)	TC1: Virtual reality office where high data rates are required in a short range transmission and low mobility context. TC9: Open air festival. FTN can improve the spectral efficiency that has to be shared by many users in a favourable transmission environment.
Impacted HTs	D2D and MMC.
System aspects addressed	xMBB
Cross-connection to other TeC clusters	TeC#8.1 FBMC
Contribution to METIS overall goals	
1000x data volume	FTN can increase the throughput of single link up by a factor of 2
10-100 user data rate	FTN can increase the throughput of single link up by a factor of 2

6.8 Filtered and filterbank based multi-carrier

As targets of both waveform approaches FBMC (TeC#8.1) and UPMC (TeC8.2) are the same, a comprehensive table for addressing the METIS goals is given for the overall TeC cluster.

Contribution to METIS overall goals	
1000x data volume	Thanks to the excellent spectrum sharing properties, spectral efficiency in multi-service scenarios sharing the same spectrum can be significantly increased.
10-100 user data rate	Efficient spectrum sharing without the need for synchronization enables significant increases of user data rates in corresponding scenarios. Individual adaptation of the system configuration per transmission link also increases the achievable rates per link.



10-100x number of devices	Due to the relaxation of the required degree of synchronization, low end devices may potentially require less PRACH interactions increasing the number of supported devices for a given PRACH capacity.
10x longer battery life	Reduced sensitivity to relaxed synchronization requirements helps reducing the on-time of devices.
5x E-E reduced latency	By allowing different subcarrier spacing, a system is able to concurrently support standard modes and low-latency modes (catchword lower frame length through shorter symbols) concurrently in a highly scalable manner. The effective reduction depends on the actual parameterization (i.e. implemented spacing and number of symbols per TTI).
Energy efficiency and cost	Same as with longer battery life. Additionally, as required accuracy of local oscillators may be relaxed, low end devices may benefit in terms of cost.

6.8.1 FBMC based waveform and transceiver design

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM1 (MAC spectral efficiency), addressed in terms of the PHY metrics, SINR, PAPR, ACLR GRM5 (MAC packet error rate), addressed in terms of the PHY metrics: BER, MSE. High robustness against sync errors enables high performance in asynchronous multi-user systems. GRT3: Supported bandwidth: FBMC can flexibly adapt to any bandwidth / spectrum mask by switching off subcarriers GRT5: Enabler for network heterogeneity: Different links can easily be supported with individual configurations thanks to spectrally contained signals (even asynchronous).
Baseline for performance comparison	OFDM / LTE settings
Update of the results w.r.t D2.3	Comprehensive summary of METIS work in the two fields: - Key aspects of FBMC as an enabler for a flexible air interface design - Solutions for practical challenges
Revolution or evolution	Revolution. Introduction of a new waveform requires redesign of the air interface and corresponding signalling procedures.
Targeted test cases (TC)	5, 6, 8, 9, 10, 11, 12
Impacted HTs	MMC, UDN, MN, D2D, URC
System aspects addressed	xMBB: eased spectrum sharing, sub-band configuration to match link conditions per user

	mMTC: support of asynchronous access, ease of power boosting thanks to spectral containment uMTC: synchronization robustness, no timing advance required in asynchronous access
Cross-connection to other TeC clusters	TeC#11.2.1

6.8.1.1 Pulse shape adaptation vs. dynamic subcarrier spacing

The FBMC modulation scheme offers a new degree of freedom in designing mobile communication systems: The selection of the pulse shape, which can be designed to meet any desired system requirements. We therefore investigated the influence of selecting different pulse shapes in FBMC systems and evaluated the performance gains attainable by adapting the pulse shape used in a mobile communication system according to current channel conditions.

The main idea behind channel adaptive modulation is to adjust the energy spread of the applied pulse shape in the time-frequency grid according to the delay and/or Doppler spread of the current channel to minimize the loss of energy which is induced by mismatched filtering at the receiver. To tune the energy distribution of the pulse according to the channel spreads, three different methods can be applied:

1. Tune the applied pulse shape in case it has a parameter which allows to adjust its energy distribution in the time/frequency grid, e.g. the enhanced Gaussian function (EGF) pulse (Figure 6-30)
2. Select a different type of pulse shape (EGF, Phydias, Hermite, etc.) Tune the system parameters subcarrier spacing and symbol duration, respectively.

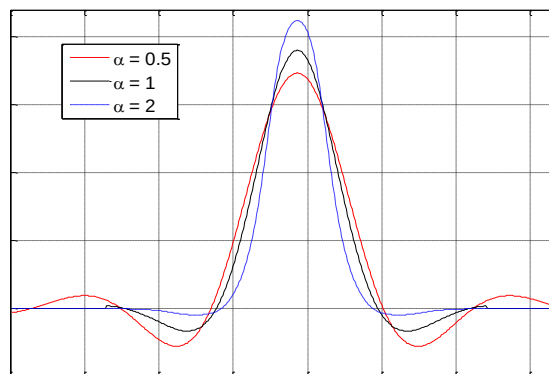


Figure 6-30 Time domain plot of EGF pulse shape utilizing different values for spreading parameter α

The first two methods are different solutions of channel adaptive pulse shaping, which adapts the mobile communication system to the channel by exchanging/modifying the applied pulse shape without changing basic system parameters. In contrast to channel adaptive pulse shaping, dynamic subcarrier spacing adapts the whole system to the channel by utilizing similar rules for the adjustment. An illustration of the achievable reconstruction performance in terms of SIR for 2D channels (with delay and Doppler spread) depending on the selected pulse shape as well as the selected subcarrier spacing is given in Figure 6-31: While for Doppler dominated channels the adjusted EGF pulse shape provides the best performance, the PHYDYAS pulse is the favourable choice for delay dominated channels, whereas the black line in the figure represents the pulse shape switching region. For dynamic subcarrier spacing, the dashed diagonal lines represent the trajectories of operating points: For a fixed

channel scenario, adjusting the subcarrier spacing will move the operation point along this line, enabling the system to realize the desired (minimal) SIR.

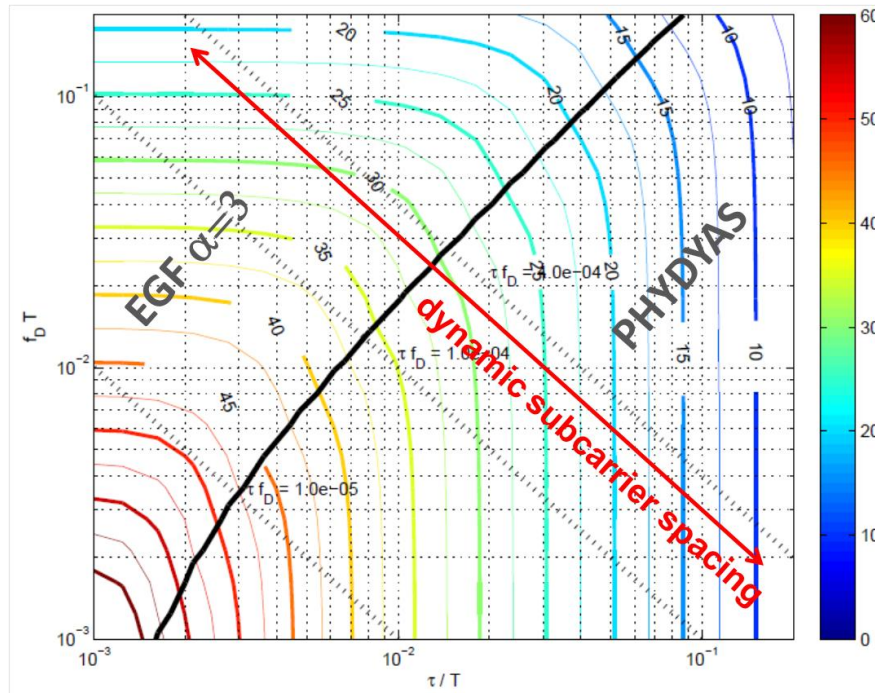


Figure 6-31 Reconstruction performance (SIR, given by the colour code) for EGF pulse with $\alpha=3$ (upper left region) and Phydias (lower right region) in dependence of the normalized delay spread τ/T and normalized Doppler spread $f_D T$.

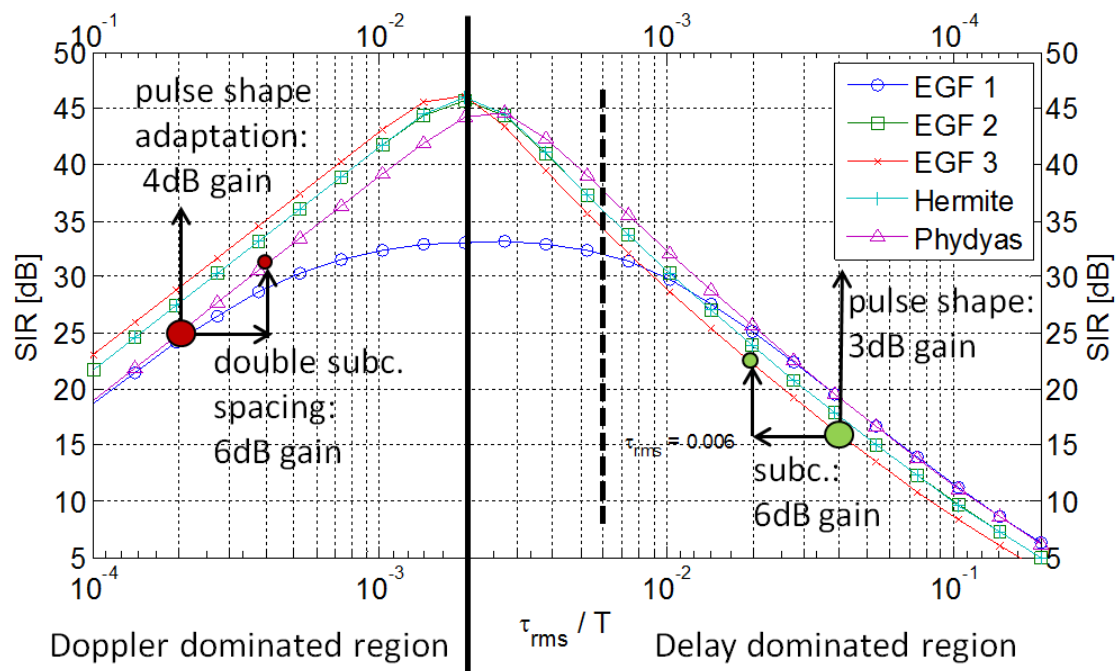


Figure 6-32 Exemplary reconstruction performance of different pulse shapes, subcarrier spacing and symbol durations, respectively, for a specific channel scenario

Figure 6-32 depicts the reconstruction performance of different pulse shapes and subcarrier spacing for an exemplary channel environment (this is equivalent to one of the dashed trajectories shown in Figure 6-31). As suggested by the matched filter approach, pulse shapes with a larger energy spread in frequency domain (e.g. EGF with $\alpha=3$) perform better in

Doppler spread dominated system configurations (large symbol duration/small subcarrier spacing; left side of the figure), and pulse shapes with a larger energy spread in time domain (e.g. Phydys and EGF with $\alpha=1$) are better in case of delay spread dominated ones (small symbol duration/large subcarrier spacing; right side of the figure). From the figure, the quantitative SIR gains attainable from pulse shape adaptation and subcarrier spacing, respectively, can easily be read.

Additional to the SIR performance, a critical system parameter for multi-user scenarios, where different users may be allocated to adjacent sub-bands with individual pulse shape and/or subcarrier spacing configuration, is the required amount of in-band guard bands for proper interference isolation. The performance gains achievable by channel adaptive modulation in combination with the required in-band guard bands for those multi-user scenarios are summarized in Table 6-7. Further investigations showed that by utilizing the in-band guard bands stated in Table 6-7, the different user signals in adjacent sub-bands are well-isolated with respect to mutual interference, so that those different user signals do not require to be synchronized in time.

Table 6-7 SIR gains and required amount of guard carriers to provide an interference isolation of 20 dB at the edge carrier between adjacent sub-bands allocated to different users

Dominating Channel Parameter	Pulse shape adaptation (Phydys $\leftrightarrow$ EGF 3)		Subcarrier spacing adaptation (factor = 2, 4 for Phydys)	
Multipath	3 dB	1 guard carrier	6.5, 14 dB	2, 4 guard carrier
Doppler	4 dB	1 guard carrier	6, 12 dB	2, 4 guard carrier

We can summarize that channel adaptive pulse shaping offers a suitable performance gain in combination with a low requirement for guard bands of only one subcarrier. Subcarrier spacing adaptation, on the other hand, can provide a much larger performance improvement at the price of an increased need for guard bands (measured in carriers based on smallest subcarrier spacing used). Reading from Table 6-7, it can be stated as a rule of thumb that a robustness increase by 3 dB requires approximately one additional subcarrier guard band.

Further details on channel adaptive pulse shaping can be found in [MET14-D23] as well as in [FPS14] and [FPS15].

6.8.1.2 Sub-band isolation in unsynchronized multi-user uplink

One of the major benefits of FBMC compared to OFDM is its capability to partition the transmission band into sub-bands that can be perfectly isolated from each other. If adjacent sub-bands are assigned to different users, they can operate independently in these bands without posing any perceptible distortions to the adjacent user's signal, even if they are not well synchronized. To prove this perfect isolation, we have carried out an investigation, where two adjacent sub-bands are assigned to different users in the unsynchronized uplink. We assume that user 1 uses the sub-band adjacent to a sub-band occupied by user 2. Both users are unsynchronized in time. Assuming that we are perfectly time synchronized with user 2, we measure the leakage power of the signal of user 1 into the sub-band of user 2 for different timing offsets, covering the range of ± 0.5 symbol durations T , which is the maximum misalignment in time shift in an OQAM-OFDM system. Results are shown in Figure 6-33. We clearly observe that by using a single guard carrier at the sub-band edge, we can perfectly isolate the signals in an FBMC system, so that the energy leakage falls below -60 dB. Compared to that, OFDM is not capable to conveniently isolate the signals; the use of guard carriers can bring the SIR down slightly below -20 dB, which may be not sufficient in most practical cases.

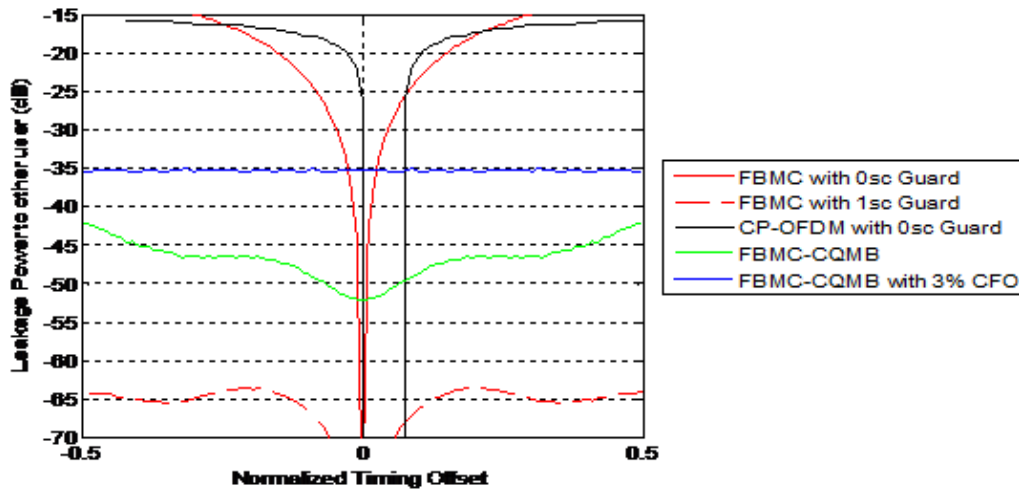


Figure 6-33 Average leakage power from adjacent sub-band occupied by user 1 (9 RBs) to the sub-band occupied by user 2 (3 RBs) in unsynchronized uplink with timing offset between user signals

6.8.1.3 MIMO FBMC

We consider a multicarrier MIMO-FBMC system with M subcarriers, with a single transmitter and a receiver, each equipped with N antennas. The bit error rate (BER) performance of a (MIMO) FBMC system is analyzed comprehensively and it is shown that the system can achieve low error performance [SRL14b] comparable to MIMO-OFDM. The system model and further details are given in [MET14-D23]. We consider a time-invariant Rayleigh fading channel, where the channel impulse response spans Λ sampling intervals. The punctured Tomlinson Harashima precoding (THP) technique is proposed which shows feasibility of FBMC. The punctured ZF also seems promising. For the precoding methods discussed, we assume perfect CSI is available at both the transmitter and the receiver. The approach is to remove the error floor, at a slight loss of throughput, by not using one or more antennas for some symbols in some subcarriers. The error floor can be removed which is quite a significant result. A general example is shown below in Figure 6-34 with energy/antenna as unity and SNR defined for a single antenna. The modulation scheme is 4-QAM.

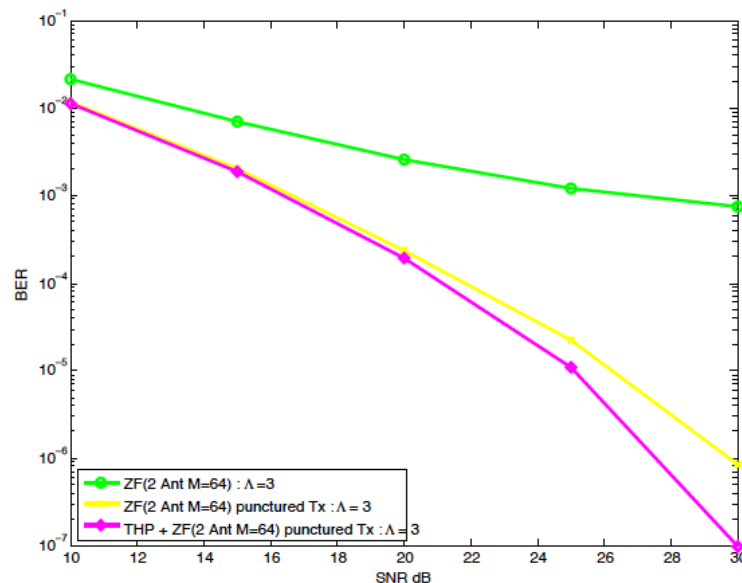


Figure 6-34 Removing the error floor by ZF precoding and puncturing (No. of subcarriers=64, Antennas=2, No. of channel taps=3)

Further results on the THP precoded system are provided below in Figure 6-35. The number of transmit and receive antennas is the same in this configuration. The total channel energy in this case is 1 with uniform distribution. The modulation scheme is 4-OQAM. The difference between THP and punctured ZF becomes higher at high SNR region. The energy/antenna as unity and SNR defined for a single antenna.

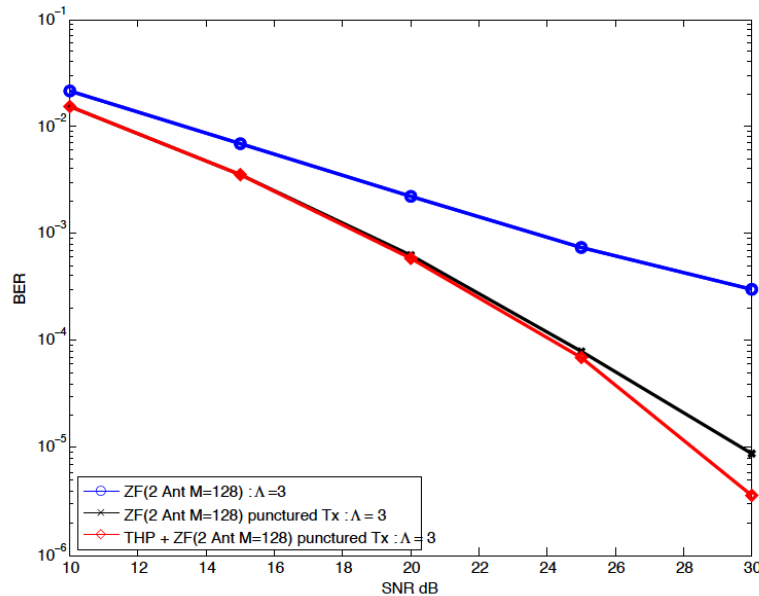


Figure 6-35 No. of subcarriers 128, No. of antennas=2, No. of taps in channel (Λ)=3

Next we consider a system with 64 subcarriers, higher channel energy channel 1.5 with three taps, first tap variance 1 and the other two equal at 0.25 each with four antennas. Here we see some difference with zero forcing as observed earlier though with punctured zero forcing the performance is quite close to that of THP. It is also seen that differences emerge as we go higher in the SNR region. The total energy of all antennas normalized to unity. The modulation scheme is 4-OQAM.

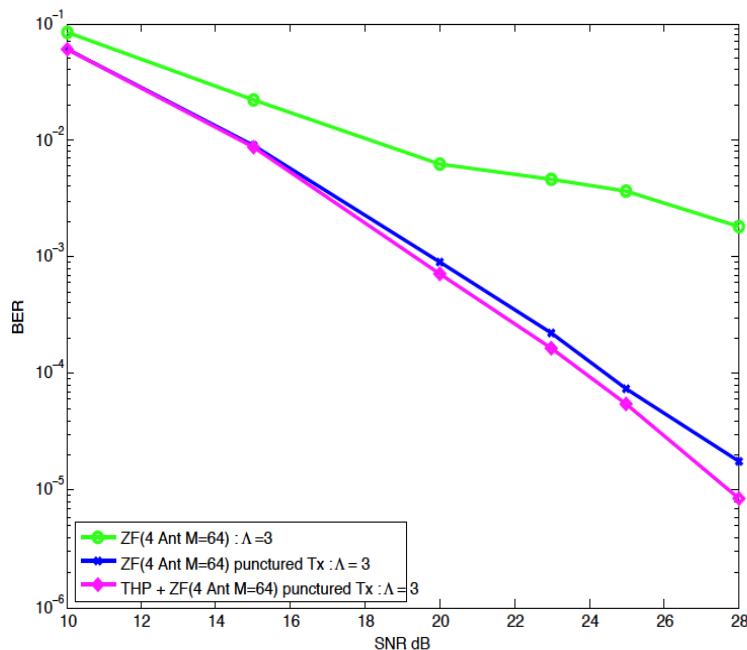


Figure 6-36 No. of subcarriers 64, No. of antennas=4, No. of taps in channel (Λ)=3

6.8.1.4 Nonlinearly Amplified FBMC Signals

In this topic, the purpose was to study the harmful effects arising in FBMC/OQAM system due to nonlinear amplification at the transmitter. Two independent campaigns were carried out. One is based on the PSD measurements on the On-the-Shelf commercial Radio-Head-Unit. The other is based on the numerical simulation with the state-of-the-art PA modelling from literature.

[Campaign 1] PSD measurement on a 1.8GHz LTE FDD radio-head-unit with output power ca. 45 dBm, including the processing effects of I/RF filtering, CFR, and DPD.

Test signal is generated according to the LTE-R8 FDD downlink signal with 5MHz bandwidth settings and subcarrier nulling of 900 KHz (i.e. 5 RB). The signal PSD is measured at the output port of the Radio-Head-Unit with RF attenuator and a spectral analyzer. From Figure 6-37 one can see that the linearity of a current practical RF unit is sufficient to sustain the PSD advantages of FBMC signal with up to 50dB in-band attenuation. Meanwhile for the in-band notching cases, OFDM without additional filtering procedure will lead to a rather poor PSD performance. This concludes the FBMC signal advantage for flexibly utilizing the spectrum resources .

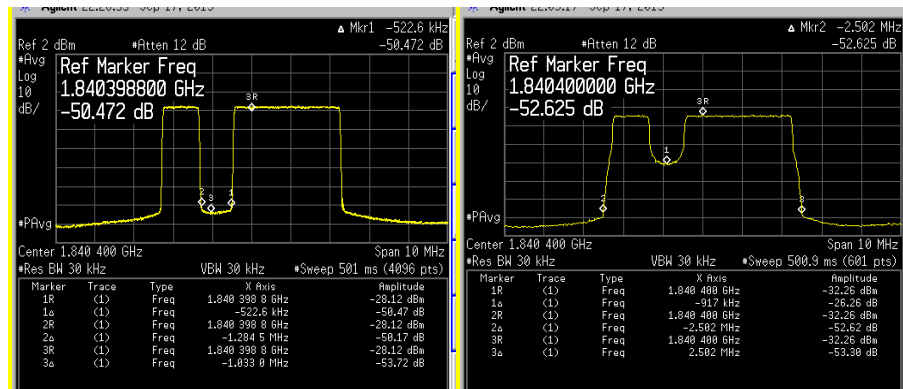


Figure 6-37 PSD measurements by (left FBMC signal) and (right OFDM signal)

[Campaign 2] For realistic performance estimation, we build the model of power amplifier (PA) based on the published measurement results of an actual PA designed for 3GPP Long Term Evolution (LTE) application [Fra12]. Using this model, we numerically analyse and compare the performance of the FBMC/OQAM system under nonlinear amplification conditions taking LTE as a reference case. We use Error Vector Magnitude (EVM) and Adjacent Channel Leakage Ratio (ACLR) for performance evaluation, since these metrics are accepted as a measure of signal quality in LTE standard i.e., EVM < 17.5% for QPSK, < 12.5% for 16-QAM and ACLR > 30 dB.

In Figure 6-38, we investigate the effect of oversampling rate (OSR), subcarrier number and constellation size on the complementary cumulative distribution function (CCDF). We can conclude that:

- Oversampled FBMC signals exhibit higher PAPR than critically sampled ones but for OSR more than 4, PAPR doesn't grow significantly.
- PAPR directly depend on the number of subcarriers in the system, so that quadrupling of the subcarriers number gives roughly 1 dB of PAPR re-growth.
- PAPR does not depend on the constellation size.

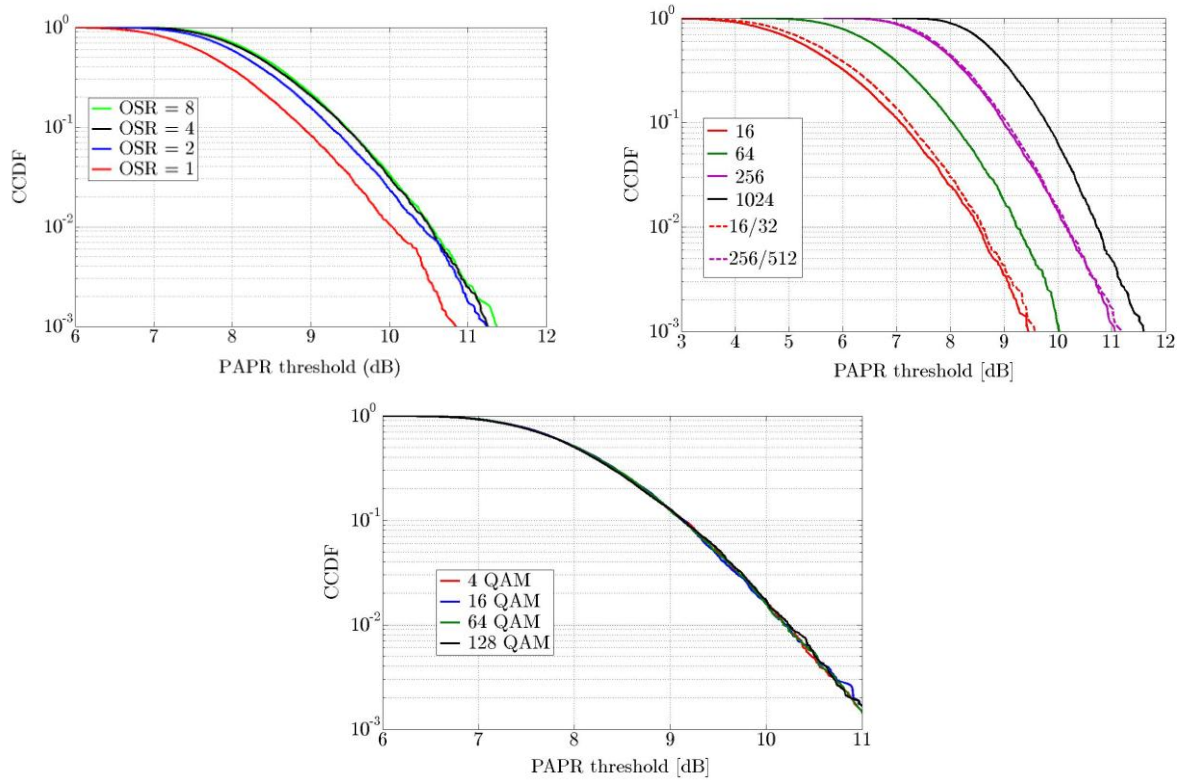


Figure 6-38 CCDF of FBMC signal vs. OSR, # of subcarriers and constellation size (Notation $\frac{M}{N}$ means M out of N subcarriers are used)

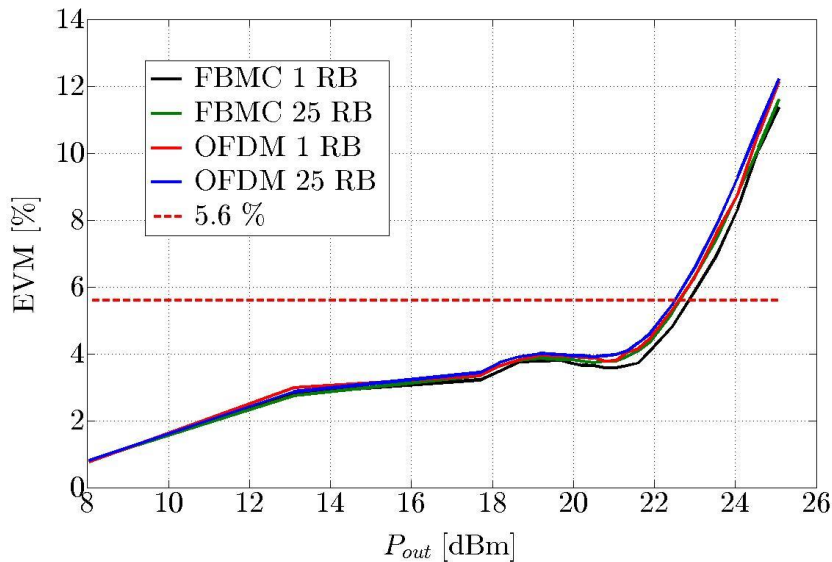


Figure 6-39 EVM vs. Pout analysis for FBMC and OFDM systems with different configurations

Figure 6-39 shows the plot of EVM vs. average output power (P_{out}) for FBMC/OQAM and OFDM/QAM signals. EVM for both systems rapidly grows once the level of the output power approaches the saturation level. It can be noticed from the plot that OFDM/QAM and FBMC/OQAM signals exhibit almost the same in-band distortions under the nonlinear amplification, although it was theoretically and experimentally approved in [Bou14] that FBMC/OQAM system is more vulnerable to the AM-PM transformations than OFDM/QAM.

Worth noting, however, is that results presented in [Bou14] were for the case of very high and unrealistic level of AM-PM distortions (approx. 60°).

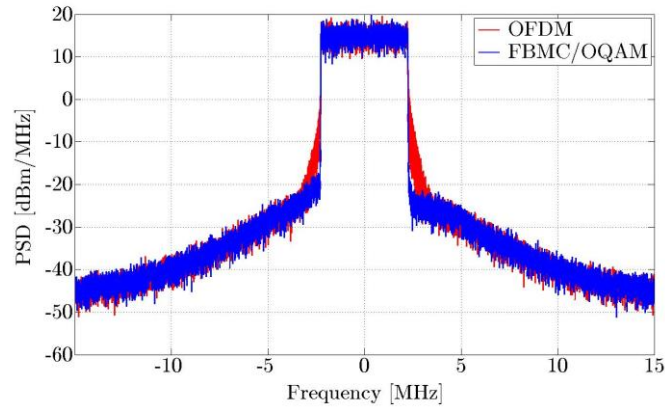


Figure 6-40 PSD of TX signal with full allocation amplified with nonlinear PA

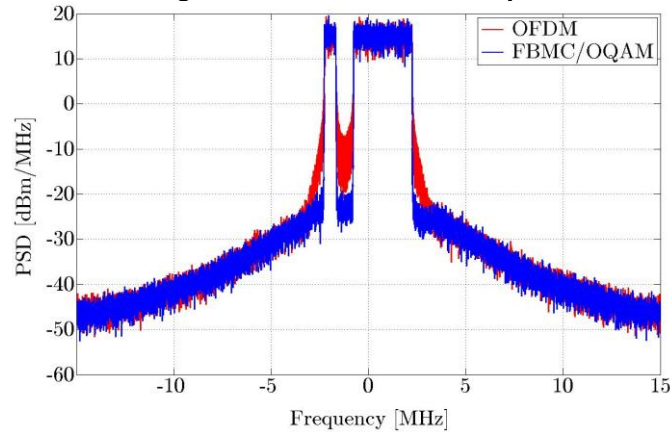


Figure 6-41 PSD of TX signal with 60 out of 300 subcarriers set to zero and later the modulated signal is amplified with nonlinear PA

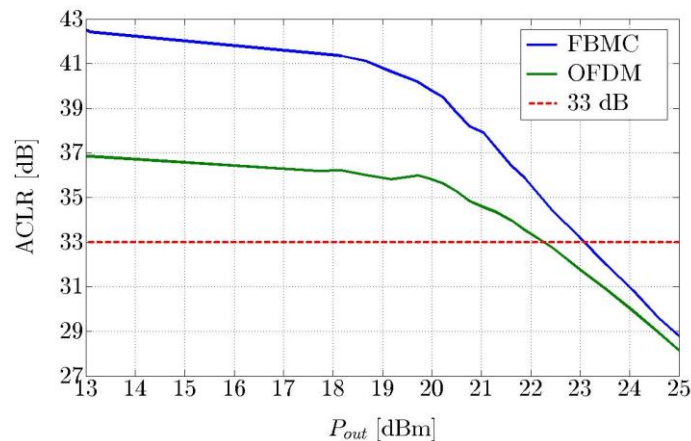


Figure 6-42 ACLR vs. P_{out} for FBMC and OFDM signals

The out-of-band (OOB) radiation and ACLR performance are investigated in Figure 6-40 to Figure 6-42. For full-allocation, one important observation is that, although original FBMC signal (not shown here) exhibits very low side lobes, the nonlinearity causes high OOB radiation that raises the side lobes by approximately 30 dBm. Nevertheless, even visually it can be noticed that FBMC system exhibits lower out-of-band radiation level even in the presence of nonlinear PA. When a contiguous set of subcarriers are unused, we can see that the decay in PSD for FBMC is sharper than OFDM that is important for coexistence scenarios

(see Figure 6-41). Finally, in ACLR experiments we have shown that FBMC exhibits significantly lower OOB radiation under the moderate nonlinear distortions. For the average output power above 22 dBm (OFDM) and 23 dBm (FBMC), both of the systems fail to satisfy the ACLR requirements. Worth noting that the slope of the FBMC line is higher than one of the OFDM line that means that the same degree of nonlinearity causes higher ACLR loss in FBMC system.

Consistent with the state-of-the-art and practices in RF design, we intend to develop a unified approach that satisfies our target requirements, namely:

1. No loss in spectral resources,
2. Less PAPR than the original signal (at least 3 dB improvement),
3. ACLR > 33 dB (keeping margin for non-PA related emissions) @ 25 dBm average output power and,
4. EVM < 5.6 % (keeping margin for other frontend imperfections).

One promising PAPR reduction technique that can satisfy requirement 1 and 2 is companding [Del14]. However, it introduces strong distortions that do not allow its use as a standalone technique for reducing effects of nonlinear distortions. Therefore, we propose zero-phase filtering to suppress out-of-band radiation and Bussgang noise cancellation (BNC) to mitigate demodulation error. Note that the same technique is also applicable to conventional OFDM systems, but the filters used to reduce out-of-band leakage do not have to be necessarily zero-phase to preserve in-band performance.

Proposed scheme allows reduction up to 4.5 dB of PAPR (Figure 6-43) and increases the average power of the transmitted signal by 2 dBm without significant degradation of EVM and ACLR (see Figure 6-45). We then show in Figure 6-46 that the proposed scheme is able to reduce BER in presence of frequency selective channel and channel estimation error (-20 dB). We also consider modifications of the proposed scheme exploiting knowledge of the amplitude characteristic of the PA and concluded that applying digital-pre-distortion (DPD) at the transmitter side allows the best performance in terms of ACLR (see Figure 6-43), while exploiting the amplitude characteristic in the BNC at the receiver significantly reduces EVM (see Figure 6-44).

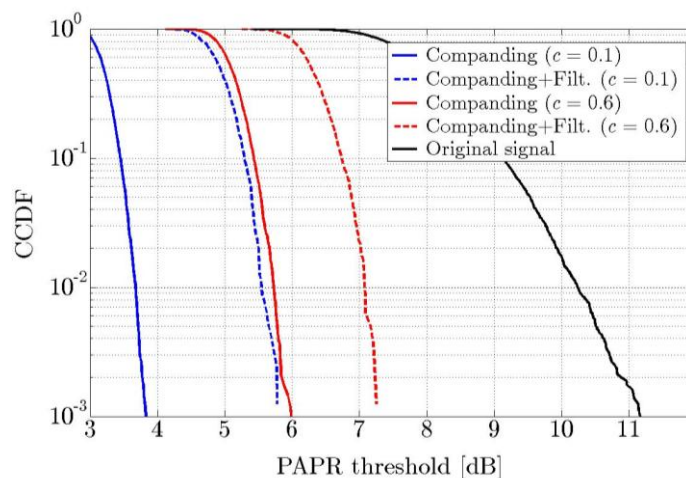


Figure 6-43 CCDF vs. PAPR threshold for companding and filtering method: Filtering uses 5th order elliptic filter with minimum 15 dB stopband attenuation.

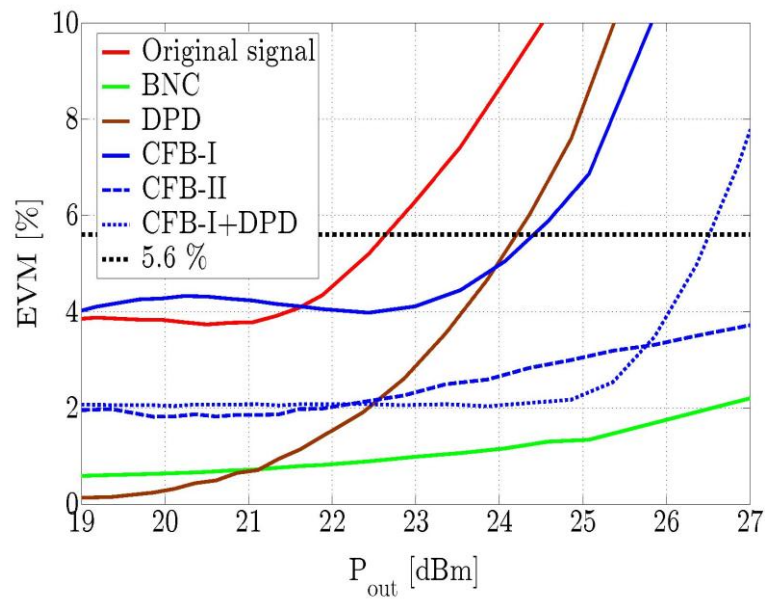


Figure 6-44 EVM vs. P_{out} improvement achieved with CFB-I (Comp.+Filt.+BNC using companding model, 1 iteration), CFB-II (Comp.+Filt.+BNC using companding & PA models, 1 iteration) and DPD methods: $c = 0.1$, 1024 subcarriers, 64-QAM symbols, AWGN channel and filtering uses 5th order elliptic filter with minimum 15 dB stopband attenuation.

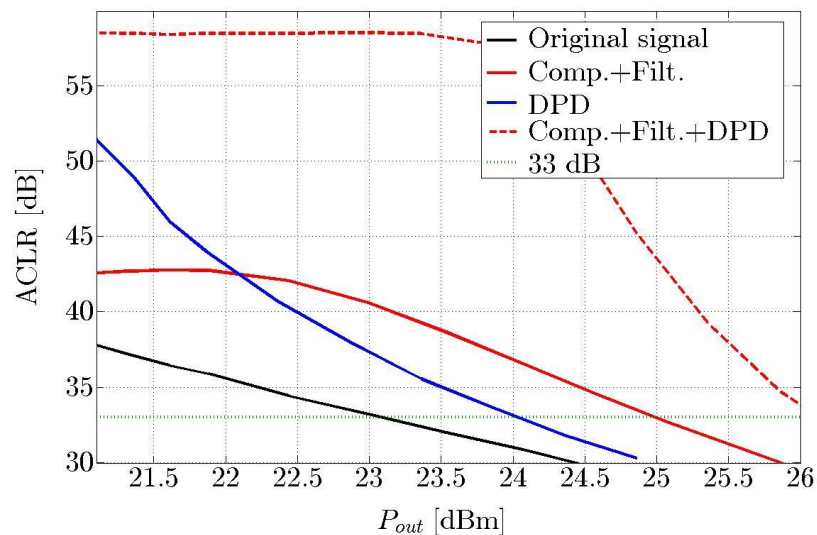


Figure 6-45 ACLR vs. P_{out} indicating improvement with the proposed filtering technique: $c = 0.1$, 1024 subcarriers, 64-QAM symbols and filtering uses 5th order elliptic filter a with minimum 15 dB stopband attenuation.

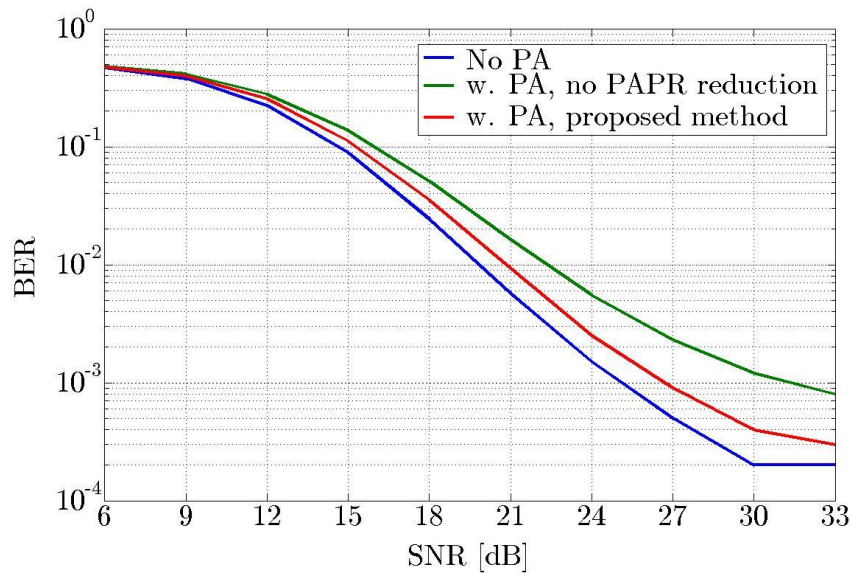


Figure 6-46 Coded BER vs. SNR for the proposed method: $c = 0.1$, 1024 subcarriers, 64-QAM symbols, time-varying Rayleigh fading channel (3GPP Typical Urban, Doppler spread = 100 Hz, CSI error variance = -20 dB) and filtering uses 5th order elliptic filter with a minimum 15 dB stopband attenuation.

6.8.1.5 Compensation of I/Q imbalance in FBMC systems

It has been reported that FBMC/OQAM is sensitive to hardware imperfections, among them I/Q imbalance being a key issue [Ish13]. In this work, we have developed widely-linear equalizer (WLE) based schemes for I/Q imbalance compensation jointly with frequency-selective channel equalization. Due to its generality, it can suppress I/Q imbalance at TX and RX side simultaneously.

A comprehensive analysis of the FBMC/OQAM widely-linear (WL) receiver in coherent Rayleigh-fading channel with TX/RX I/Q imbalance is provided in terms of the un-coded (Figure 6-47) and coded (Figure 6-48) error rates. In the simulations, the total number of subchannels is $K = 64$, extended Gaussian filters (EGF) are used with $\lambda = 2$ and they overlap over four consecutive symbols. A 10-tap channel model is assumed with i.i.d complex Gaussian random variables and generated from exponential power delay profile. Relatively higher I/Q imbalance is considered in RX w.r.t TX to mimic a downlink scenario with 4% and 8% mismatch at TX and RX respectively (for more details on I/Q imbalance model, please refer to [AG14]). The bit interleaved coded modulation structure has a rate $1/2$ – convolutional code with generator polynomial $[171,133]_8$ combined with a random interleaver.

Results presented in Figure 6-47 show the effectiveness of a time-domain (TD) based WLE methods where the image interference due to I/Q imbalance was almost fully mitigated. It is obvious that FBMC/OQAM has nearly similar performance as CP-OFDM/QAM using frequency-domain equalization (FDE) and provides substantial improvement over strictly-linear equalizers (SLEs).

In Figure 6-48, we show a comparison of the BER performance among the strictly-linear equalizer (SLE) and proposed pairwise equalizer (PWE). The PWE with residual interference cancellation (RIC) gains around 3 dB at a BER of 10^{-4} in the case of 16-QAM signals. This is because higher constellations suffer more from intrinsic contamination. It is also obvious that exploiting RIC with decoding iterations helps improves WL-PWE performance. On the other hand, for small constellations, PWE without RIC is enough for near-optimal performance.

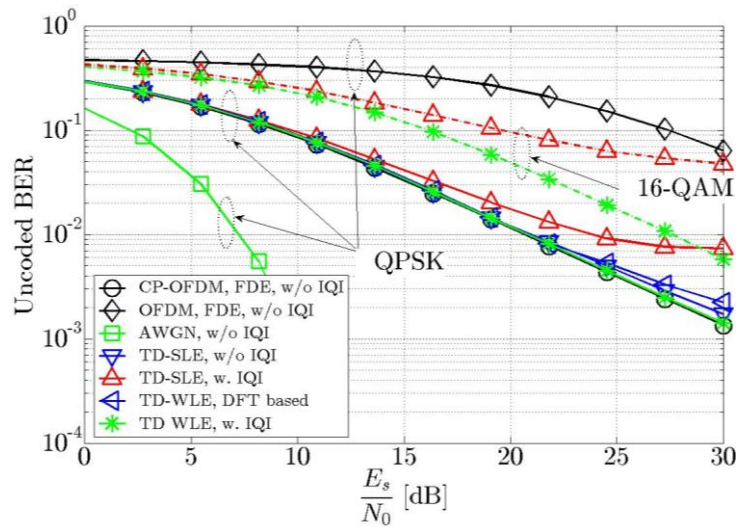


Figure 6-47 Uncoded BER versus symbol SNR of FBMC/OQAM system with QPSK or 16-QAM modulation symbols, channel taps =10, and normalized delay spread $\tau_{rms}= 5$.

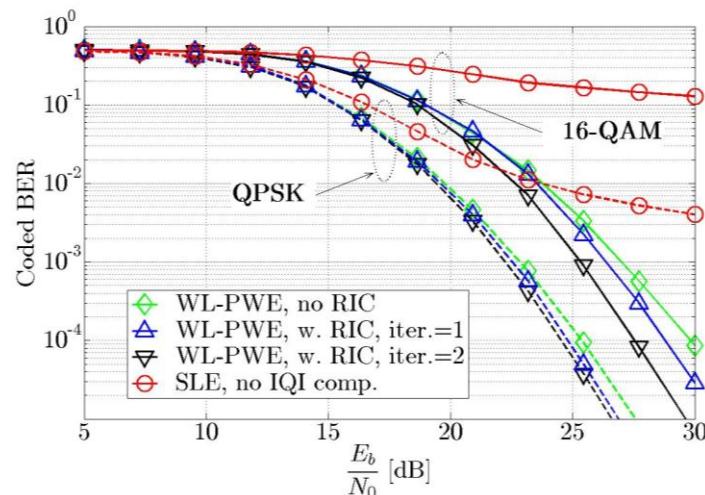


Figure 6-48 BER versus bit energy per noise spectral density in a coded FBMC/OQAM system with optimized RIC size

6.8.1.6 Impact of phase noise on FBMC performance

Multicarrier systems are more sensitive to the phase noise introduced by the oscillators in the RF frontend, in the sense that they give rise to inter-carrier interference (ICI) instead of self-rotation in single carrier systems. This translates into either more stringent requirements on RF frontend or higher complexity of baseband processing for digital ICI compensation. Although some degree of phase noise compensation is performed in any practical digital frontend, the relative sensitivity of future waveforms to phase noise is important, since it defines the starting point of an iterative decision-directed compensation approach.

We consider a zero-mean stationary phase process that resembles practical oscillators in terms of its power spectral density (PSD) shape. One of the PSD shape of choice is the so-called linear decay spectrum, for which the frequency-domain characterization of the phase noise process is given by:

$$S_{\theta}(f) = 10^{-c} + \begin{cases} 10^{-a} & |f| \leq f_l \\ 10^{-a-b\frac{f-f_l}{f_h-f_l}} & f > f_l \\ 10^{-a+b\frac{f+f_l}{f_h-f_l}} & f < -f_l \end{cases}$$

where f_l is the 3 dB bandwidth of the reference phase noise, the parameters a and c denote the noise floors at frequencies lower than f_l and higher than f_h respectively, whereas b represents spectral decay in between the two white noise regions. In numerical simulations, the phase noise process $\theta(n)$ has PSD shape with parameters set to $f_l = 10$ KHz, $f_h = 100$ KHz, $a = 7, b = 4, c = 11$ (shown in Figure 6-49) and having variance $\sigma_{\theta}^2 \approx 4 \times 10^{-3}$ @ 10 MHz sampling rate. For more details, refer to [Ish15].

In Figure 6-50, we have computed ICI as a function of the subcarrier offset. It shows that ISI in FBMC/OQAM ($k = 0$) is at par with common phase error in OFDM/QAM and the first subcarrier in FBMC/OQAM is the major contributor to ICI. Also, as the localization factor of EGF filter λ is reduced, higher frequency confinement of the filter caused steeper ICI fall. One can easily conclude that in the absence of phase noise compensator, an ISI resilient pulse ($\lambda > 2$) is preferable, whereas, with compensation scheme, the low-pass behaviour of FBMC/OQAM ($\lambda < 2$) essentially requires lower-order ICI mitigation than OFDM/QAM.

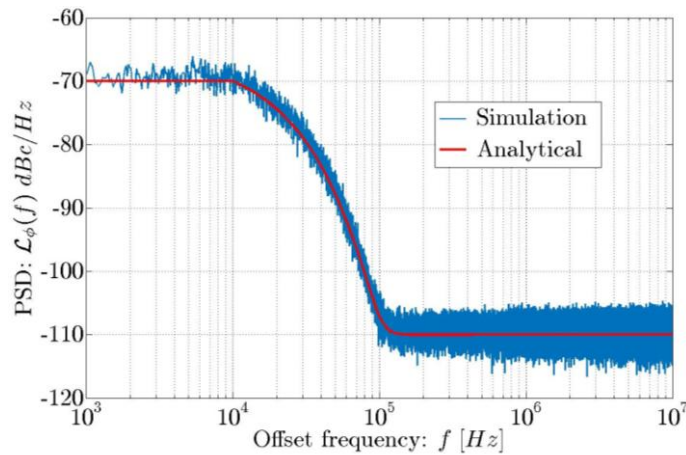


Figure 6-49 Power spectral density of the phase noise process.

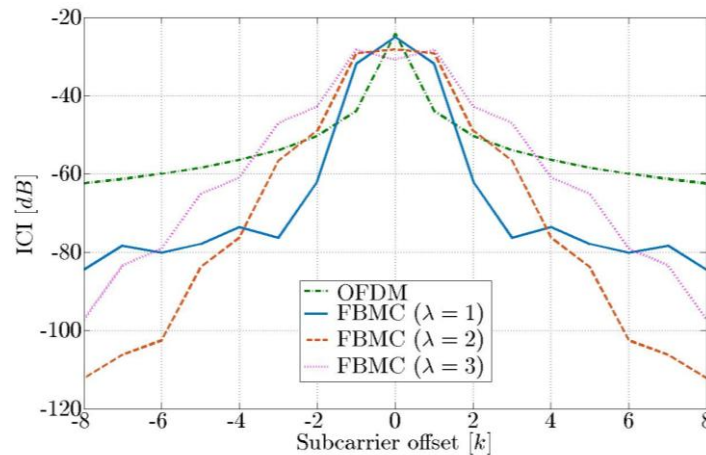


Figure 6-50 ICI extension in FBMC/OQAM and OFDM/QAM systems without channel fading: K=128 and B=10 MHz

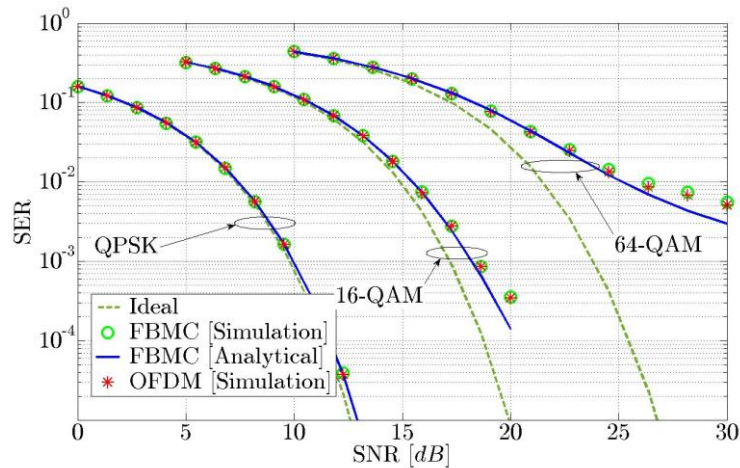


Figure 6-51 SER versus SNR for various constellation using FBMC/OQAM and OFDM/QAM systems under AWGN channel

In Figure 6-51, we show the sensitivity of FBMC/OQAM in terms of SER using hard-decisions. Analytical SER approximation based on Gaussian ICI assumption is also plotted and seems to fit well with the numerical trials for all constellations. It is clear that higher constellations suffer more from phase noise impact while FBMC/OQAM and OFDM/QAM have equivalent SER performance.

In Figure 6-52 a) and b), we study the effect of subcarrier spacing f_{sub} relative to the corner frequency f_1 on the SIR performance. A decrease in distortion with $\frac{f_{\text{sub}}}{f_1}$ is justifiable because smaller f_{sub} relative to f_1 will cause ICI components to assemble in the vicinity of the decay region in Figure 6-49. Moreover, decreasing f_{sub} will decorrelate phase noise samples. Intuitively, a more noise-like phase noise process increases sensitivity to phase noise distortion and as a result, makes the phase noise estimation a challenge. A trade-off in FBMC systems is that the reduction of f_{sub} w.r.t channel coherence bandwidth reduces the impact of channel frequency-selectivity. It is also shown that the EGF pulse with larger λ outperforms smaller values by reducing ISI contribution (see Figure 6-50). In fact, OFDM forms a tight upper bound on the FBMC performance and is achievable only for larger λ or smaller $\frac{f_{\text{sub}}}{f_1}$.

When the transmission suffers from channel fading, we can deduce from Figure 6-52 b) that a gain of up to 2 dB is expected from FBMC for the defined channel conditions due to better frequency confinement. The loss in wider spacing can be attributed to FBMC's non-orthogonality in frequency-selective channels. The regime $f_{\text{sub}} \ll f_1$ is dominated by phase noise and Doppler effects. Due to orthogonality against multipath channel ensured by cyclic-prefix (CP), the interference power in OFDM case is independent of channel frequency-selectivity.

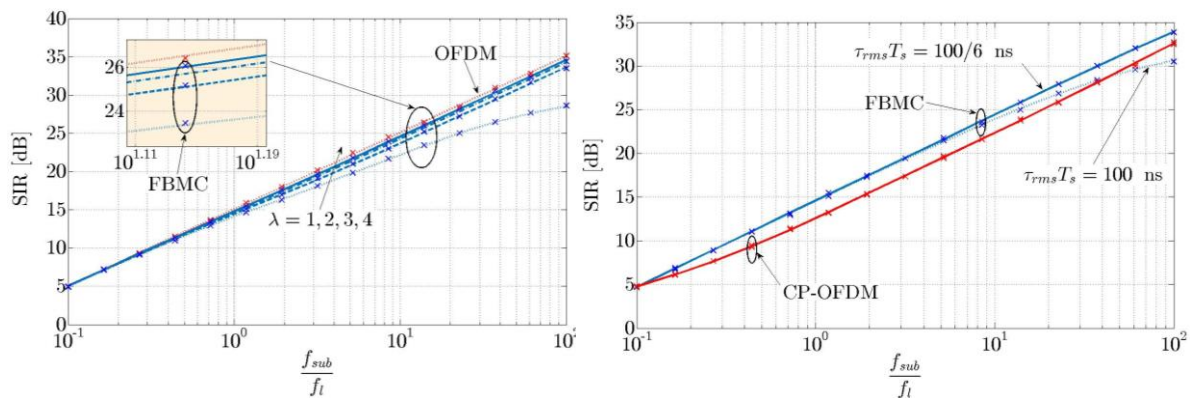


Figure 6-52 Signal-to-interference ratio versus subcarrier-spacing-to-corner-frequency ratio $\frac{f_{sub}}{f_l}$ for a fixed subcarrier number $K=128$. Left figure a) AWGN channel and right figure b) Rayleigh fading ($\lambda=2$, channel taps=3 and Doppler shift=600 Hz).

6.8.2 Universal Filtered Multi-Carrier (UFMC)

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	- inter-user interference reduced in scenarios with simplified access protocols by $\sim 10\times$ with LTE-like settings (increased tolerance against relaxed synchronism). With dedicated parameter optimizations higher performance gains possible. - overhead scaling with burst length - per service/user adaptation enabled - Modular service/user-specific L1/L2 treatment enabled
Baseline for performance comparison	LTE settings
Update of the results w.r.t D2.3	Improved support of high velocities (see appendix)
Revolution or evolution	It replaces the basic waveform (revolution), but it is able to reuse most of the concepts/schemes of 4G (evolution)
Targeted test cases (TC)	2, 5, 8, 9, 10, 11, 12
Impacted HTs	TeC has been selected for the HT system concept: MN, URC, D2D, MMC, UDN
System aspects addressed	Focus in METIS has been in enabling - protocol simplification for massive MTC (MMC) - low latency modes for low latency MTC (MCC) - per sub-band adaptation for MBB
Cross-connection to other TeC clusters	none

6.8.2.1 Further evaluation results

5G is foreseen to have to support both low-speed/stationary users and high speed users (e.g. within high-speed trains), concurrently. So, the system has to cope with time varying channels with highly differing correlation properties. This has some important consequences we have to take care for when designing the system.

The coherence time of the wireless channel between device and base station is depending on the relative speed between those two nodes (the other characterizing parameter is the carrier frequency). Figure 6-53 depicts the channel correlation coefficient c defined as follows

$$c(\Delta_t) = \frac{\text{cov}(H_n, H_{n+\Delta t})}{\sqrt{\text{var}(H_n) \text{var}(H_{n+\Delta t})}}$$

for different velocities in [km/h] depending on the time distance between the complex channel realisations H_n and $H_{n+\Delta t}$ (carrier frequency: 2.5 GHz):

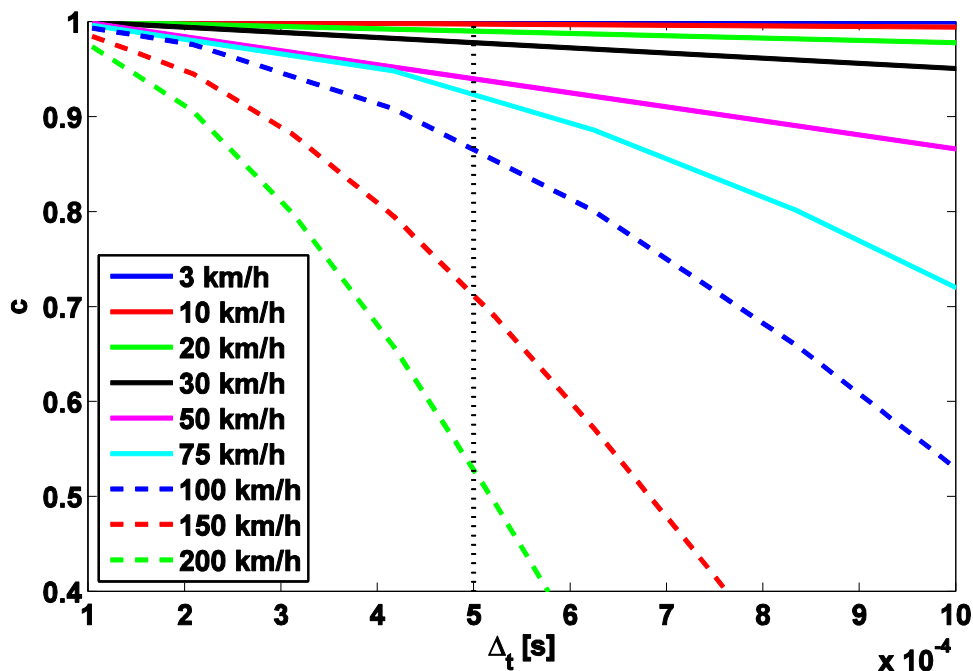


Figure 6-53 Correlation coefficient between channel realizations depending on time distance between the measurements

As expected the higher the velocity the faster the correlation between two channel realisations drops. The vertical line depicts $\Delta t = 0.5$ ms – the distance between the pilots in LTE uplink. So, while for low speeds (up to 75 km/h, here) c stays above 0.9, it quickly shrinks for even higher speeds. Naturally, this behaviour depends on the carrier frequency. The results presented here are with 2.5 GHz.

In general, channels being time variant has several impacts. With increasing speed:

- Doppler spread is increasing, too, leading to stronger ICI (covered in METIS D2.3)
- coherence time of the channel shrinks (see above) having the following consequences:
 - Gains from frequency selective scheduling are shrinking (not treated, here)
 - Channel estimation gets worse (assuming a given pilot pattern, e.g. LTE UL)

Here, we concentrate on the last element. Typically, a system applies a given pilot pattern leading to a given distance between pilots (e.g. 0.5 ms with LTE uplink). To investigate the impact of this, we have performed link level simulations measuring the post detection estimation error (mean squared error in dB between modulation symbols s having been transmitted and symbols $\tilde{s}$ at the receiver after equalization):

$$MSE = 10 \log_{10} E \left[|s - \tilde{s}|^2 \right]$$

For estimating the channel we so far use a rather simple channel estimator:

1. Estimate the channel at the pilot positions
2. Apply noise smoothing in frequency direction (sliding window size: 7 subcarriers)
3. Linear interpolation in time direction

The resulting estimation error is as follows (LTE UL settings, QPSK, channel model: eVEHA according to 3GPP TS 36.101 V10.5.0):

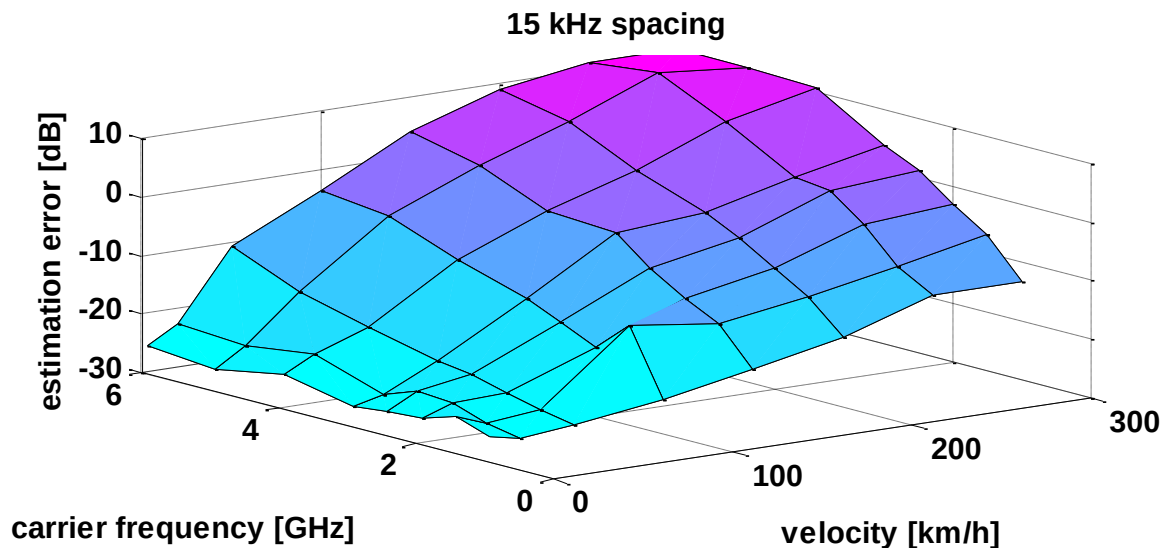


Figure 6-54 Estimation error depending on velocity and carrier frequency (15 kHz spacing)

As predicted, while being reasonable for low speeds the estimation error grows fast with growing speed. Additionally, the higher the carrier frequency, the higher the estimation error gets.

There are two options to improve this:

- Increase the pilot density by inserting more pilots
- Increase the sub-carrier spacing for users with high velocities

The former one is disadvantageous, as it increases the pilot overhead. With applying the second option the pilots (while keeping the PRB definition fixed, i.e. number of sub-carriers, number of multi-carrier symbols, pilot positions) are moving closer in absolute time. Though, with increasing spacing the single symbols become shorter increasing the sensitivity to delay spread, accordingly. Naturally, this has to be taken into account. Ways to combat this are using zero-tail DFT spreading as described in [BTS+13] or to adapt the filter coefficients to increase protection against delay spread. The following figure depicts the respective PRB settings (green rectangles: pilot positions, white areas: data):

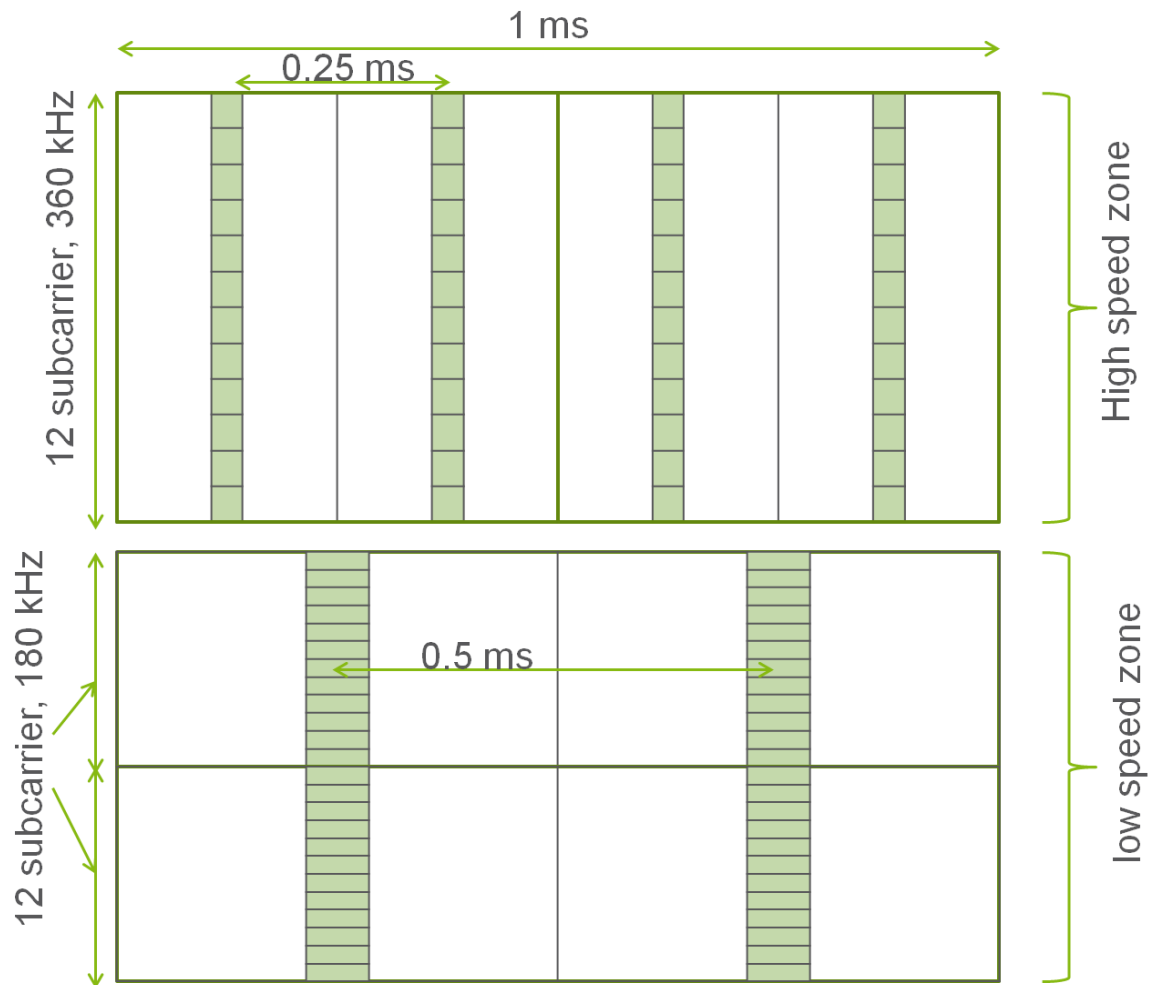


Figure 6-55 PRB definitions

With applying this, pilot overhead is unchanged. Channel estimation is improved, as the absolute distance between the pilots is reduced. It is not reasonable to make this adaptation continuously. Instead, a code-book based approach as indicated in the following is preferable:

- Low speed users with reasonable delay spread $\rightarrow x$ kHz spacing, w/o zero-tails
- Low speed users with high delay spread $\rightarrow x$ kHz spacing, w/ zero-tails
- High speed users with reasonable delay spread $\rightarrow 2 \times x$ kHz spacing, w/o zero-tail
- High speed users with high delay spread $\rightarrow 2 \times x$ kHz spacing, w/ zero-tail

To indicate the potential gain we have measured the estimation error when applying 30 kHz spacing instead of 15 kHz (as applied above):

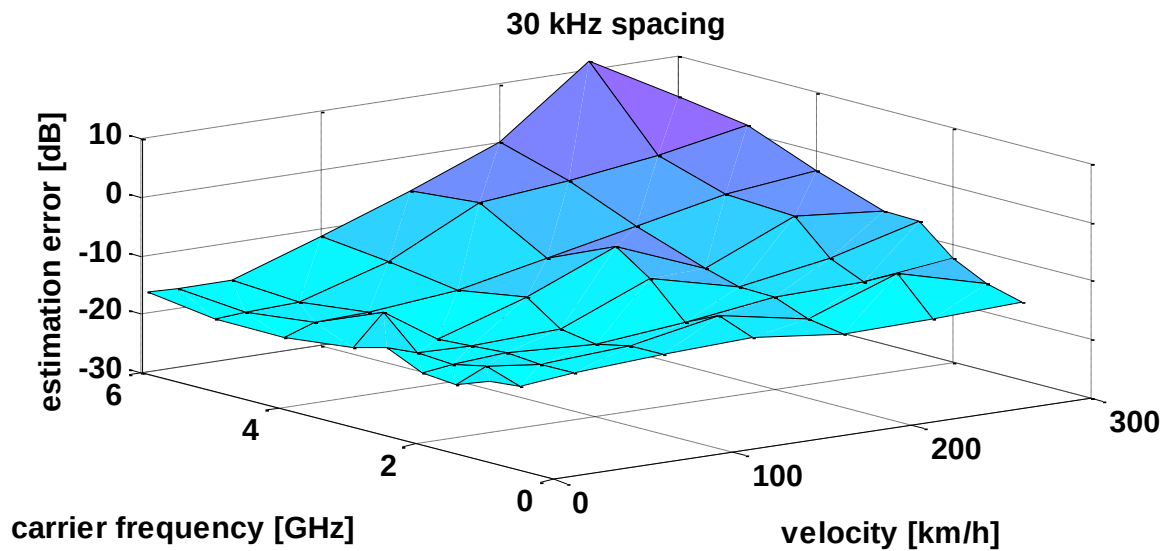


Figure 6-56 Estimation error depending on velocity and carrier frequency (30 kHz spacing)

Comparing Figure 6-54 and Figure 6-56, we see as expected, that for low speeds 15 kHz is preferable, as here the delay spread effect is dominating, while for high speeds the estimation error due to unmatched pilot distances becomes dominant. So far, no means to deal with delay spread is applied.

The following figures compare the two variants for given carrier frequencies to better visualize the impacts:

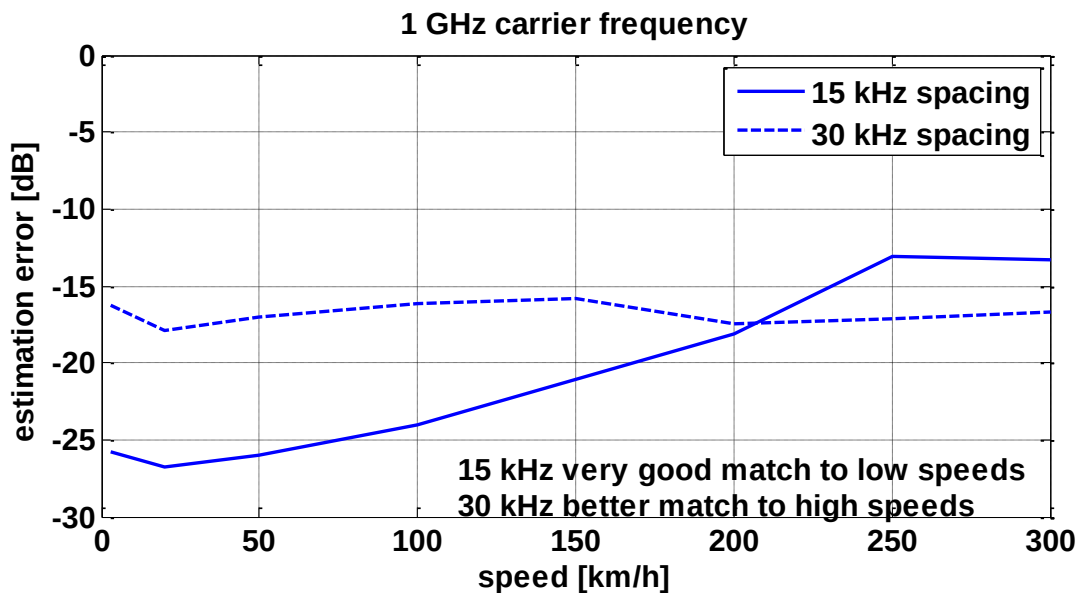


Figure 6-57 Estimation error depending on the velocity (1 GHz carrier spacing)

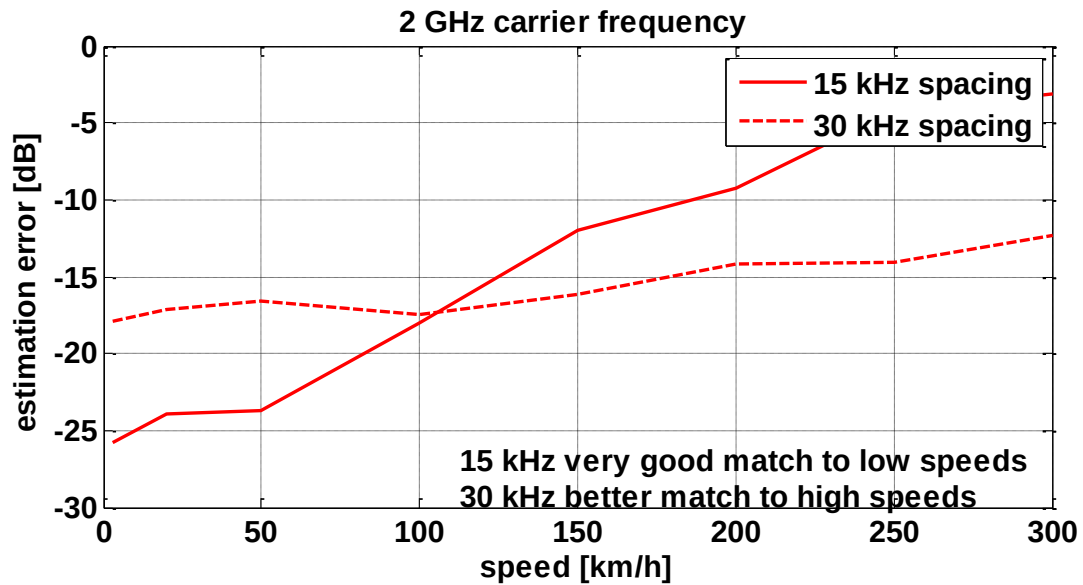


Figure 6-58 Estimation error depending on the velocity (2 GHz carrier spacing)

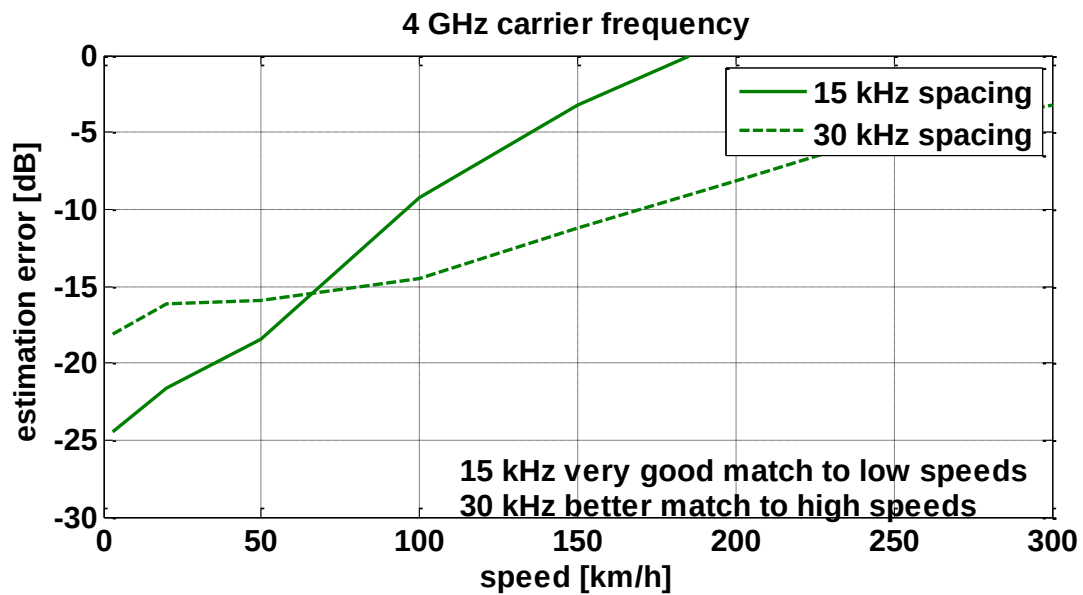


Figure 6-59 Estimation error depending on the velocity (4 GHz carrier spacing)

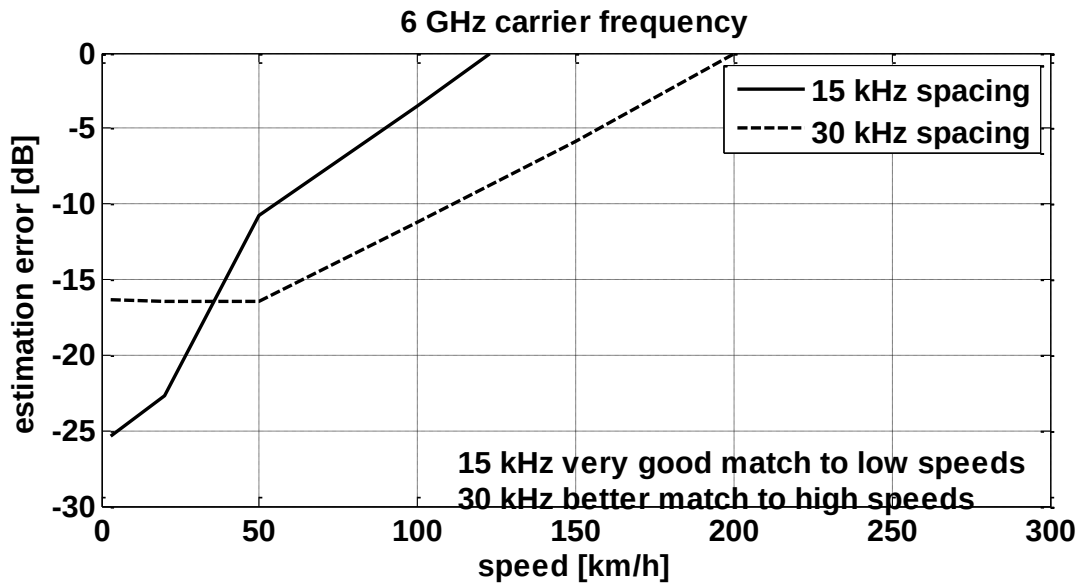


Figure 6-60 Estimation error depending on the velocity (6 GHz carrier spacing)

As expected, for low speeds 15 kHz spacing is preferable, while for high speeds 30 kHz performs better. The actual switching point differs for different carrier frequencies. Additionally, we expect to see an additional impact with applying a more sophisticated estimator (future work), though, the general principle still holds.

This principle is applicable to any multi-carrier format. Though, for scaling reasons it might be beneficial to enable the system to mix different spacing within a single TTI (i.e., in a FDMA manner within a single band) instead of applying different spacing between different TTIs (i.e., in a TDMA manner). Then, a multi-carrier format applying a filtering functionality (as e.g. UFMC aka UF-OFDM) is preferable, as the two zones are more easily separated. Naturally, when using different spacing in adjacent sub-bands special means have to be applied to minimize the inter-sub-band interference. Though, this is not treated, here.

Future work:

- More sophisticated channel estimator
- Means to reduce interference between bands applying different spacing
- Means to deal with delay spread
- Rate improvements by applying the proposed scheme.

6.8.2.2 Prove: orthogonality between subcarriers with FFT based detection

In the following we show that with using the appropriate receiver, orthogonality between the subcarriers is maintained. For doing so we investigate the system in back-2-back mode, i.e. receiver directly connected to the transmitter without channel and noise.

We denote with $\mathbf{y} \in \mathbb{C}^{(N+\tilde{L}) \times 1}$ the vector of $N + \tilde{L}$ complex receive samples of a single multi-carrier symbol, where N is the length of the main body of the symbol in samples (e.g. $N = 1024$ for a system with 10 MHz following LTE settings) and $\tilde{L}$ is the time domain overhead in samples ($\tilde{L} = L - 1$ with L being the filter length). So, we have:

$$\mathbf{y} = \mathbf{T}\mathbf{s}$$

where $\mathbf{s} \in \mathbb{C}^{Q \times 1}$ represents the data symbol vector sent on a subband of Q subcarriers drawn from an complex modulation alphabet (e.g. QPSK). $\mathbf{T} \in \mathbb{C}^{(N+\tilde{L}) \times Q}$ is the multi-carrier modulation matrix, with $\mathbf{T} = [\mathbf{t}_0, \mathbf{t}_1, \dots, \mathbf{t}_{Q-1}]$. The n -th column vector $\mathbf{t}_n$, with $n = 0, 1, \dots, Q-1$, is responsible for modulating the k -th subcarrier with $k = n+K$, where the *offset* integer K is responsible for the relative subband position within the available subcarriers. In fact, index n addresses the subcarrier within the subband, whereas index k addresses the subcarrier position in the overall band.

For UF-OFDM the (not yet normalized) multi-carrier modulation matrix $\tilde{\mathbf{T}}_{UF} \in \mathbb{C}^{(N+\tilde{L}) \times Q}$ is

$$\tilde{\mathbf{T}}_{UF} = \mathbf{F}\mathbf{V} = [\tilde{\mathbf{t}}_{UF,0}, \tilde{\mathbf{t}}_{UF,1}, \dots, \tilde{\mathbf{t}}_{UF,Q-1}]$$

where $\mathbf{F} \in \mathbb{C}^{(N+\tilde{L}) \times N}$ is the Toeplitz matrix for carrying out the discrete linear convolution with the subband band-pass FIR filter with coefficients $\mathbf{f} = [f_0, f_1, \dots, f_{L-1}]^T \in \mathbb{C}^{L \times 1}$. The first column of $\mathbf{F}$ consists of $\mathbf{f}$ appended with a $(N-1) \times 1$ zero vector. $\mathbf{V} = [\mathbf{v}_0, \mathbf{v}_1, \dots, \mathbf{v}_{Q-1}]$, with $\mathbf{V} \in \mathbb{C}^{N \times Q}$, is carrying out a N -point inverse DFT (IDFT) on the data symbol vector $\mathbf{s}$ with $[\mathbf{V}]_{m,n} = \exp(j2\pi m(n+K)/N)$, $m=0,1,\dots,N-1$ ($[\mathbf{V}]_{m,n}$ denotes the entry on row m and column n of matrix $\mathbf{V}$).

We consider two options for normalizing the matrix $\tilde{\mathbf{T}}_{UF}$:

1. With the first option we only constrain the overall radiated energy per subband to be Q . After denoting with $\mathbf{T}_{sb}$ the normalized multi-carrier modulation matrix with this first option is ($\text{tr}(\mathbf{X})$ is the trace of matrix $\mathbf{X}$)

$$\mathbf{T}_{sb} = \sqrt{\frac{Q}{\text{tr}(\tilde{\mathbf{T}}_{UF}^H \tilde{\mathbf{T}}_{UF})}} \tilde{\mathbf{T}}_{UF}.$$

2. With the second option we force the overall radiated energy per subband to be Q and additionally we also constrain the subcarrier symbol energy to be unitary. After denoting with $\mathbf{T}_{sc} = [\mathbf{t}_{[sc],0}, \mathbf{t}_{[sc],1}, \dots, \mathbf{t}_{[sc],Q-1}]$ the normalized multi-carrier modulation matrix is:

$$\mathbf{t}_{[sc],n} = \frac{1}{\sqrt{\tilde{\mathbf{t}}_{UF,n}^H \tilde{\mathbf{t}}_{UF,n}}} \tilde{\mathbf{t}}_{UF,n}.$$

Let us denote with $\mathbf{G} = [\mathbf{g}_0, \mathbf{g}_1, \dots, \mathbf{g}_{Q-1}] \in \mathbb{C}^{(N+\tilde{L}) \times Q}$ an arbitrary demodulation matrix. As the equations are the same for both normalization methods, we just drop the subscripts here for readability and consider a general modulation matrix $\mathbf{T} = [\mathbf{t}_0, \mathbf{t}_1, \dots, \mathbf{t}_{Q-1}]$. The demodulated subcarrier symbol vector $\mathbf{r} \in \mathbb{C}^{Q \times 1}$ is thus $\mathbf{r} = \mathbf{G}^H \mathbf{y}$ and the n -th demodulated subcarrier symbol can be written as

$$r_n = \mathbf{g}_n^H \mathbf{y} = \underbrace{\mathbf{g}_n^H \mathbf{t}_n s_n}_{r_n^{[u]}} + \underbrace{\sum_{v=0, v \neq n}^{Q-1} \mathbf{g}_n^H \mathbf{t}_v s_v}_{r_n^{[i]}}$$

where $r_n^{[u]}$ is the useful signal and $r_n^{[i]}$ is the inter-carrier interference. For proving subcarrier orthogonality we require $r_n^{[i]}$ to vanish.

The simple full-band low complex baseline demodulator for UF-OFDM is based on a $2N$ -FFT [MET14-D23]: It pads the receive vector $\mathbf{y}$ with zeros to a total length of $2N$, executes a $2N$ -FFT, and picks each second output as the demodulated subcarrier. Let us denote with $\mathbf{G}_{FFT} =$

$[\mathbf{g}_{\text{FFT},0}, \mathbf{g}_{\text{FFT},1}, \dots, \mathbf{g}_{\text{FFT},Q-1}]$ the combiner applied by the receiver to vector $\mathbf{y}$. After denoting with $\tilde{\mathbf{V}}$ the $2N$ -size IDFT matrix, i.e., $[\tilde{\mathbf{V}}]_{a,b} = \exp(j2\pi ab/2N)$, $a, b = 0, 1, \dots, 2N-1$, matrix $\mathbf{G}_{\text{FFT}}$ is obtained by simply selecting from matrix $\tilde{\mathbf{V}}$ the first $N + \tilde{L}$ rows and the Q even columns corresponding to the considered subband position, i.e., we can write

$$[\mathbf{G}_{\text{FFT}}]_{m,n} = [\tilde{\mathbf{V}}]_{m,2n+2K} = e^{j\frac{2\pi}{N}m(n+K)}, \quad m = 0, 1, \dots, N + \tilde{L} - 1, n = 0, 1, \dots, Q - 1. \quad (*)$$

To achieve orthogonality between subcarriers we require:

$$\mathbf{g}_{\text{FFT},\nu}^H \mathbf{t}_n = 0, \quad \nu \neq n, \quad \nu, n = 0, 1, \dots, Q - 1.$$

As $\mathbf{t}_n = \mathbf{F}\mathbf{v}_n$, where $\mathbf{F} \in \mathbb{C}^{(N+\tilde{L}) \times N}$ is a lower triangular Toeplitz matrix with vector $\mathbf{f}$ on the first L entries of its first column, we can write

$$\mathbf{t}_n = \mathbf{\Psi}_n \mathbf{f} \quad (**)$$

where $\mathbf{\Psi}_n \in \mathbb{C}^{(N+\tilde{L}) \times L}$ is also a lower triangular Toeplitz matrix, but with vector $\mathbf{v}_n$ on the first N entries of its first column. Then, by performing the matrix multiplication in (**), we obtain

$$\mathbf{t}_n = \sum_{l=0}^{L-1} f_l [\mathbf{0}_{1 \times l}, \mathbf{v}_n^T, \mathbf{0}_{1 \times (\tilde{L}-l)}]^T \quad (***)$$

$\mathbf{0}_{1 \times b}$ is a vector of length b with all zero entries.

Then, by observing from (*) that

$$\mathbf{g}_{\text{FFT},\nu} = [\mathbf{v}_\nu^T, [\mathbf{v}_\nu]_{0:(\tilde{L}-1)}^T]^T$$

with $[\mathbf{x}]_{0:b}$ denoting the vector obtained by extracting the first b entries from vector $\mathbf{x}$ and by using (***) we can write:

$$\begin{aligned} \mathbf{g}_{\text{FFT},\nu}^H \mathbf{t}_n &= \sum_{l=0}^{L-1} f_l \mathbf{g}_{\text{FFT},\nu}^H [\mathbf{0}_{1 \times l}, \mathbf{v}_n^T, \mathbf{0}_{1 \times (\tilde{L}-l)}]^T \\ &= \sum_{l=0}^{L-1} f_l e^{-j\frac{2\pi}{N}l(n+K)} \mathbf{v}_\nu^T \mathbf{v}_n \end{aligned}$$

Finally, due to DFT orthogonality $\mathbf{v}_n^H \mathbf{v}_\nu = 0, \nu \neq n$, we obtain

$$\mathbf{g}_{\text{FFT},\nu}^H \mathbf{t}_n = \begin{cases} N \sum_{l=0}^{L-1} f_l e^{-j\frac{2\pi}{N}l(n+K)}, & \nu = n \\ 0, & \nu \neq n \end{cases}.$$

6.9 Modulation & coding and new channel coding concepts

6.9.1 Constrained envelope coded modulation

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	KPI is energy efficiency per bit with low cost hardware. Corresponding GRM/GRTs are GRM7, GRM6, GRT1 and the robustness relates to GRM5.
Baseline for performance comparison	An LTE-like performance benchmark, using rate $\frac{1}{2}$ convolutionally encoded SC-FDMA.
Update of the results w.r.t D2.3	The work on this TeC since D2.3 has been focused on its implications on and potential integration in the METIS system concept.
Revolution or evolution	Evolution
Targeted test cases (TC)	TC11, TC2
Impacted HTs	MMC, UDN
System aspects addressed	Energy efficiency, coverage, robustness
Cross-connection to other TeC clusters	There are several cross-connections identified in the project, relating to e.g. MAC design and synchronization, but resources in the project were not enough to implement such joint studies.
Contribution to METIS overall goals	
10x longer battery life	Yes, potential gain seems to be up to 4 dB, but actual values depends a lot on the high power amplifier used in the terminal, as well as overall MAC design.
5x E-E reduced latency	Indirectly yes, by utilizing a large system bandwidth for the frequency interleaving of the users. That would lower the outage due to fading and narrowband interference, and thus lower the need for retransmissions that contribute to the E-E latency.
Energy efficiency and cost	The scheme has a focus on improving the terminal energy efficiency by using open loop communication with amplifier-friendly transmit signals. Overall gains on system-level energy efficiency has not been evaluated.

Further details on TeC9.1 “Constrained envelope coded modulation” can be found in [MET14-D23].

6.9.2 Advanced coding and decoding

Further details on research results of TeC cluster 9.2 “Advanced coding and decoding” can be found in [MET14-D23], [AG15].

6.9.2.1 Adaptive complexity flexible baseband

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	Related metrics from WP2 are GRM7 ECON-B/AU and GRT1 COST-HW. The improvement is measured in terms of reduction in computational complexity when compared to other receiver structures with respect to a target error rate for a particular system configuration.
Baseline for performance comparison	The compared techniques are applied at the receiver side for a LTE system. The target is to provide manufacturers with the performance/complexity comparison of different receiver algorithms under particular system parameters to set a sort of a baseline when a target error rate is set. Indeed, results show that conclusions vary depending on system parameters and channel conditions.
Update of the results w.r.t D2.3	Minor updates w.r.t. D2.3 as our contribution to this activity has been ended by Nov. 2013.
Revolution or evolution	Evolution
Targeted test cases (TC)	All test cases where MIMO can be of relevance.
Impacted HTs	All HTs where MIMO can be of relevance
System aspects addressed	Complexity, latency
Cross-connection to other TeC clusters	Cross-connection to other TeCs has not been addressed as this activity ended by Nov. 2013.
Contribution to METIS overall goals	
10x longer battery life	Yes. Concrete values for achievable gains depend on specific scenarios and cases.
5x E-E reduced latency	Yes. Concrete values for achievable gains depend on specific scenarios and cases.
Energy efficiency and cost	Proposed multi-mode adaptive-complexity MIMO detector allows to dynamically tuning energy consumption depending on system parameters and channel conditions.

6.9.2.2 Practical lattice codes

Technical component profile	
KPIs, GRMs and GRTs considered and achieved gain	SNR, Error rate, spectral efficiency partly encompassed in GRM1 and GRM4 (only link level).
Baseline for performance comparison	None
Update of the results w.r.t D2.3	Improved coding gain for lower spectral efficiencies, Rayleigh fading channel performance
Revolution or evolution	Revolution if channel coding is replaced by lattice codes. Evolution if lattice codes are concatenated with the baseline channel coding.
Targeted test cases (TC)	N/A
Impacted HTs	N/A
System aspects addressed	N/A
Cross-connection to other TeC clusters	N/A
Contribution to METIS overall goals	
1000x data volume	Addressed by the increased spectral efficiency
10-100 user data rate	Addressed by the increased spectral efficiency

6.10 Advanced transceiver design

6.10.1 Full Duplex communications

Technical component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM4: Compared to TDD systems, full-duplex doesn't need guard intervals and waiting for transmission times. So latency can be reduced. GRM6: Maximum coverage area can be calculated considering the maximum transmit power and the QoS requirements. GRM1: 100% gain for the D2D link possible
Baseline for performance comparison	Rate comparison to HD is considered. Double rate is achieved for same bandwidth for the D2D link.
Update of the results w.r.t D2.3	New results with more practical parameters, comparison to half duplex
Revolution or evolution	Revolution due to changes necessary in TX/RX
Targeted test cases (TC)	TeC has not been selected, but could be useful for the following TC(s): TC2, TC8

Impacted HTs	TeC has not been selected, but could be useful for the following HT(s): MN, D2D
System aspects addressed	- xMBB (extreme mobile broadband)
Cross-connection to other TeC clusters	
Contribution to METIS overall goals	
1000x data volume	Data volume of the entire system will be doubled compared to conventional half-duplex system
10-100x number of devices	For the same rate, number of devices can be doubled because of spectral efficiency that full-duplex provides.
5x E-E reduced latency	Latency is reduced in TDD systems.

Further evaluation results:

Additional results from [ARL14] are given in Figure 6-61. We consider a single-cell scenario where 30 users are randomly placed in the cell, and two users separated by less than 25 meters are selected as a D2D pair. It is assumed that each user has two LTE resource blocks to communicate on the uplink, and the D2D users have to use sub-channels of other cellular users. In Figure 6-61, we present the ratio of the FD D2D link rate to the HD D2D link rate for the scenario, where the D2D pair is sharing resources with a single cellular user. The curves are parameterized by the SI cancellation factor in dB. As the SNR target increases, the cellular user can transmit with a larger power, as can the D2D users. However, this will also lead to an increase in residual SI, and so FD performance will worsen.

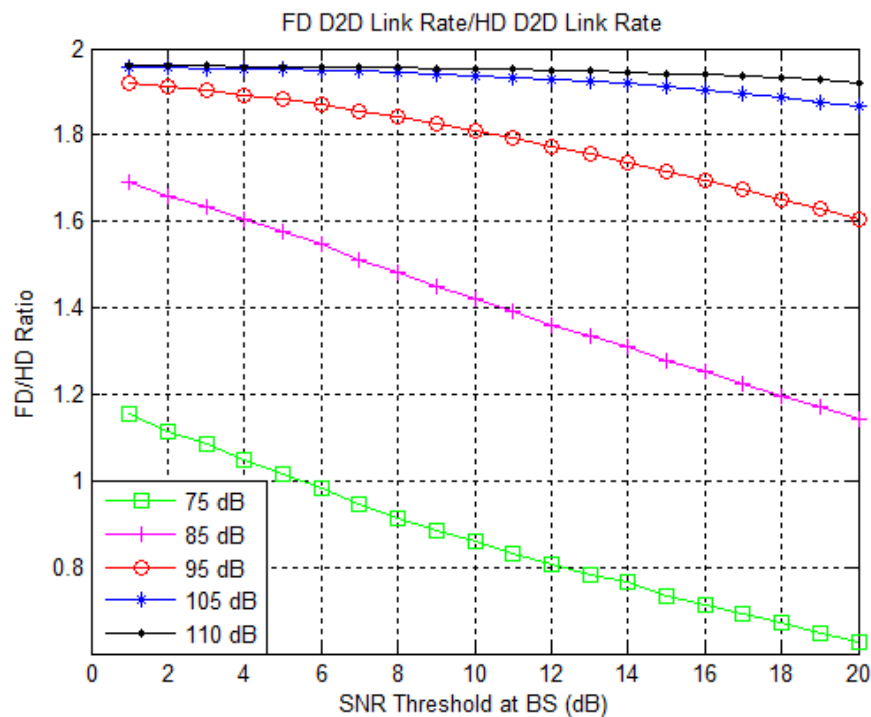


Figure 6-61 FD/HD Rate Ratio as a function of minimum SNR at BS for one user resource sharing

6.10.2 Multi-rate equalizer for single-carrier communications

Technical component profile	
KPIs, GRMs and GRTs considered and achieved gain	Energy consumption, experienced user throughput, energy consumption, modem hardware cost
Baseline for performance comparison	DFT precoded OFDM specified in LTE
Update of the results w.r.t D2.3	New section on FTN
Revolution or evolution	Simple changes are necessary to LTE channel encoding
Targeted test cases (TC)	TC2
Impacted HTs	D2D and MMC: MRE allows using single carrier effectively.
System aspects addressed	xMBB (extreme mobile broadband)
Cross-connection to other TeC clusters	MRE inspires a practical FTN scheme (TeC#7), which provides 30% improvement in throughput.
Contribution to METIS overall goals	
10-100 user data rate	If a SC transmission is employed provides a 50% throughput increase
10x longer battery life	SC reduces PAPR, thus, improves power amplifier efficiency
Energy efficiency and cost	SC allows using cheaper amplifier, synchronizers.

Further evaluation results:

The proposed MRE based FTN transmission is simulated with QPSK modulation and a family of multiple rate LDPC codebooks [VWW04] of length 4096. This family contains codebooks of rates varying from 0 to 0.89. The codebook which yields error-free transmission over one million bits is determined via simulations. In a multiple rate LDPC codebook family, lower rate codebook is always a subset of the higher rate codebook. This feature of multiple rate LDPC codes ensures fair simulation. The results of this simulation are shown in Figure 6-62 Average throughput per channel use for. These results confirm that R_1 can indeed achieve the throughput with Nyquist rate communications and R_0 provides the extra throughput. Furthermore, the extra throughput provided by R_0 is always less than R_1 which is expected since the rate R_0 is affected by both interference and noise while R_1 is affected by noise only. Overall gain in throughput varies between 30% and 50% depending on the SNR. However, in the region of interest it is around 30%.

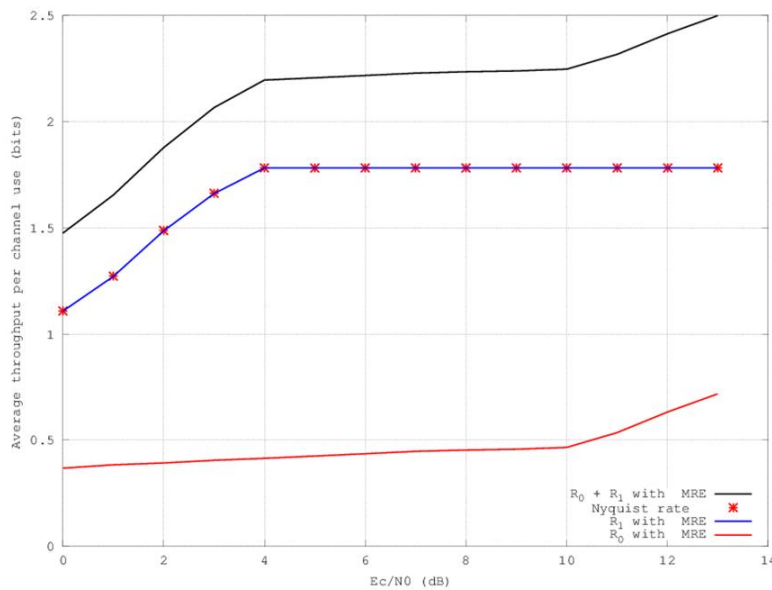


Figure 6-62 Average throughput per channel use for FTN transmission equalized with MRE with QPSK

6.11 Multiple Access

6.11.1 Non- and quasi-orthogonal multiple access allowing spectrum overload

6.11.1.1 NOMA

Technical component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM1: Multi-user/cell throughput (MU/C-TP) and MAC spectral efficiency (MAC-SE) - The system throughput can be improved by up to 70% compared to OFDMA (LTE rel. 8). - More gains are obtained as we set the target cell-edge user throughput higher. - With proportional fairness (PF) scheduling, NOMA can further increase the geometric mean of user throughput which can be translated to capacity gains or fairness gains GRM2: Max supported number of connections (MAX#-CONN) - The number of users transmitting at the same time can be doubled. - Almost 90% of users are using NOMA and less than 10% are using OFDMA. GRT1: Modem hardware cost (COST-HW) - SIC receiver is required at the receiver side. GRT2: Carrier frequency range (CFR) - Not dependent on frequency band but can be more attractive in lower frequency bands with more coverage.

Baseline for performance comparison	OFDMA (LTE Rel. 8 parameters)
Update of the results w.r.t D2.3	Proposal and evaluation of for NOMA combined with SU-MIMO
Revolution or evolution	It can be introduced both as an evolution and as a revolution
Targeted test cases (TC)	TC2: Dense environment with very large number of users TC8: High mobility environment
Impacted HTs	NOMA is a fundamental technology to improve spectrum efficiency. Therefore, it can be basically applied to all HTs. - For NOMA, user multiplexing is conducted in the power-domain by exploitation of SNR difference among multiple links or devices; therefore, multi-user transmission is enabled with no need to increase the number of transmit antennas - NOMA does not rely much on CSI feedback from the receiver for user multiplexing at the transmitter; therefore, it has good robustness to mobility. Examples of impacted HTs are: MN, UDN, D2D
System aspects addressed	xMBB (extreme mobile broadband)
Cross-connection to other TeC clusters	The design of CSI feedback for NOMA combined with SU-MIMO is also studied under WP3 TeC#8
Contribution to METIS overall goals	
1000x data volume	Up to 1.5x
10-100 user data rate	Up to 1.5x (even at high mobility)
10-100x number of devices	Up to 2x

Further evaluation results:

NOMA gains for small cells: 2x2, ISD=200m, 20MHz, 10UEs, 44dBm, 3km/h, ITU UMi channel model, TM3 (Open-loop MIMO)

- For a given target cell edge user throughput the achievable cell throughput can be significantly increased using NOMA. The cell throughput gains increase more (up to 70%) as the cell-edge user throughput target is set higher.

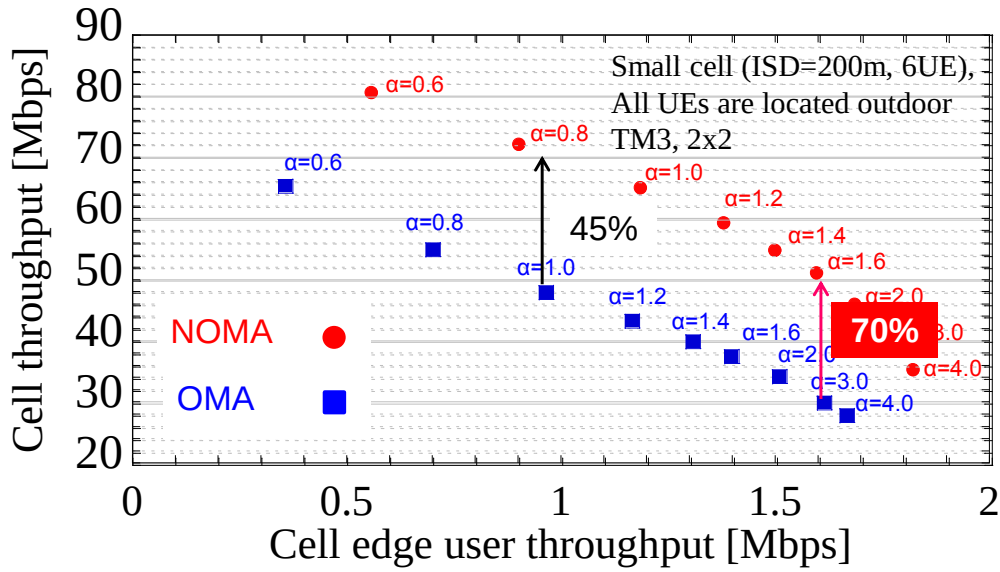


Figure 6-63 Cell throughput vs. cell-edge user throughput

For a given target cell throughput the achievable cell edge user throughput can be significantly increased using NOMA. The cell-edge user throughput gains increase more (up to 234%) as the cell throughput target is set higher.

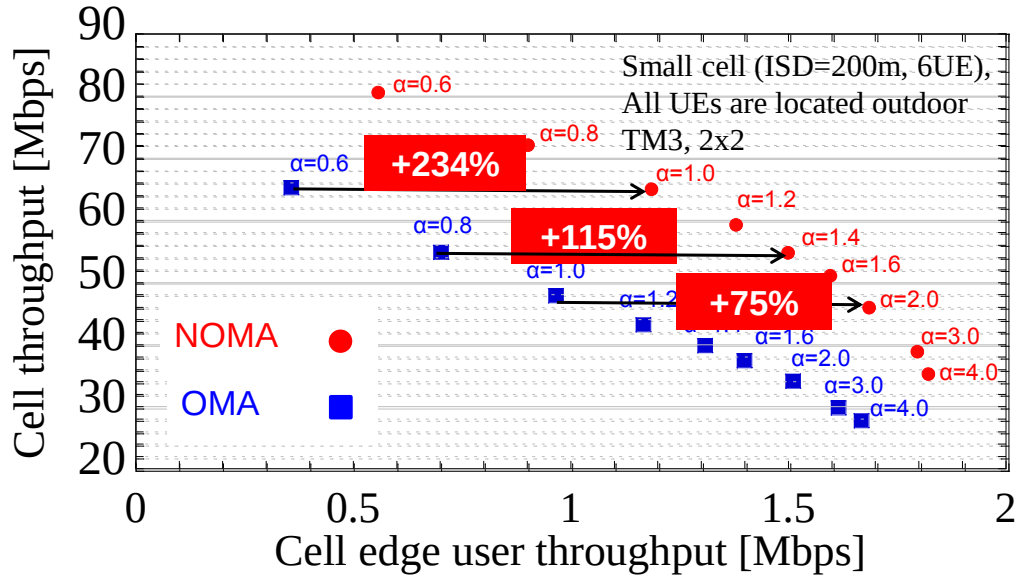


Figure 6-64 Cell throughput vs. cell-edge user throughput

where a multi-user proportional fair (PF) scheduler is used to select the NOMA user set S^* that maximizes the following scheduling metric:

$$S^* = \arg \max_S \sum_{k \in S} \frac{R_k(t)}{T_k^\alpha(t)}$$

$R_k(t)$ is the instantaneous throughput of user k at time instance t (the time index of a subframe), whereas $T_k(t)$ is the average throughput of user k , S indicates a candidate user set and α is a weighting parameter. Note here that the higher the value of the parameter α is, the fairer the scheduling becomes, since the scheduler gives more priority to scheduling cell-edge users (which have lower average throughput) than scheduling cell-centre users.

6.11.1.2 Sparse Code Multiple Access (SCMA)

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	DL: Cell throughput, 5% cell coverage, perceived throughput UL: Number of supported users for random-access
Baseline for performance comparison	DL: LTE OFDM MIMO system (Release 11) UL: LTE Release 11 grant-based UL transmission
Update of the results w.r.t D2.3	SCMA codebook design with low number of projections SCMA blind detection for UL random access New results for DL MIMO MU-SCMA and MU-SCMA/CoMP for full and non-full buffer traffic scenarios UL contention based SCMA results are updated with number of devices supported, using a LTE R11 baseline
Revolution or evolution	This is a new modulation and multiple access scheme but it can easily be implemented on top of the current OFDM-based LTE structure.
Targeted test cases (TC)	DL: TC2, TC6, and TC8 UL: TC5, TC9, and TC11
Impacted HTs	Selected for MN, UDN, MMC. Can be useful for D2D
System aspects addressed	DL: xMBB UL: mMTC, uMTC
Cross-connection to other TeC clusters	TeC#12.1: Contention based massive access
Contribution to METIS overall goals	
1000x data volume	MU-SCMA and SCMA with open-loop CoMP can be applied to address the goal of higher data volume
10-100 user data rate	MU-SCMA and SCMA with open-loop CoMP can be applied to address the goal of higher user data rate
10-100x number of devices	Contention based UL SCMA mechanism is able to support up to 10+ number of devices
10x longer battery life	Contention based UL SCMA mechanism can save up to 25 times active time reduction
5x E-E reduced latency	MU-SCMA can reduce the latency Contention based UL SCMA mechanism without the dynamic request and grant procedure is able to reduce significantly the air-link time and support the transmission with ultra low latency, as low as 1ms.

Energy efficiency and cost

Contention based UL SCMA mechanism can save up to 25 times active time reduction, thus increasing energy efficiency

MIMO SCMA

To further improve the spectral efficiency, SCMA can be combined with MIMO techniques. In this section, we propose some open-loop MIMO transmission schemes for SCMA OFDM.

MIMO for multiplexing: Open-loop single-user and multi-user multiplexing exploiting both space and code domain to multiple layers of data stream. This combined technique is called code-space multiplexing (CSM) which is the extension of the spatial multiplexing (SM). Ignoring the complexity issue, one can apply ML detector over all combined SCMA layers of all MIMO streams. However to enjoy the sparsity feature of SCMA and hence limit the complexity of reception, the MIMO streams are first separated by a MIMO detector and then SCMA layers of each MIMO stream are decoded separately following the MPA reception technique.

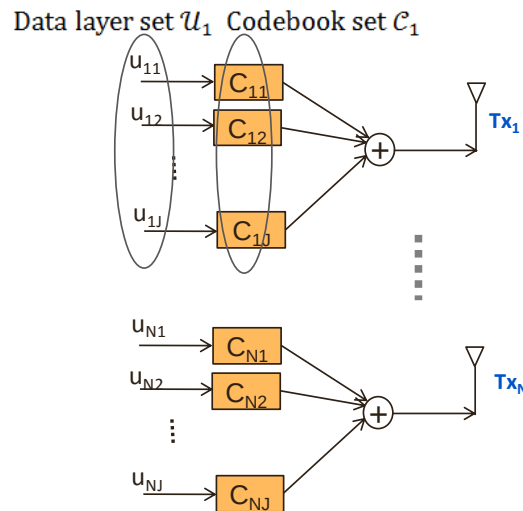


Figure 6-65 MIMO CSM for SCMA

MIMO for transmit diversity: Block-wise space-time/frequency coding (BWST) is introduced to apply space-time coding techniques such as Alamouti over SCMA blocks to achieve further diversity. Figure 6-66 illustrates the block-wise Alamouti code for SCMA. A block-wise Alamouti detector is used at the receiver to recover the SCMA block before feeding into the MPA detector. Notably, the columns of the block can be arranged to control the inter-block interference caused by the channel variation.

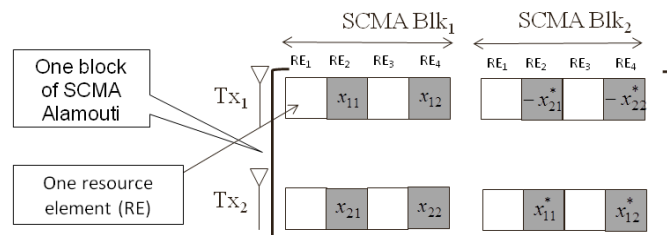


Figure 6-66 Block-wise Alamouti encoder for SCMA

BF-CSM: As shown in Figure 6-67, BF-CSM is a closed loop scheme combining CSM and MIMO beamforming to separate the MIMO streams at the transmit side. The beamforming is UE group based with a lower sensitivity to the precision of the precoder. The UE groups are separated with beamforming and then SCMA layers of each group are separated by using MPA detection at every UE belonging to a group.

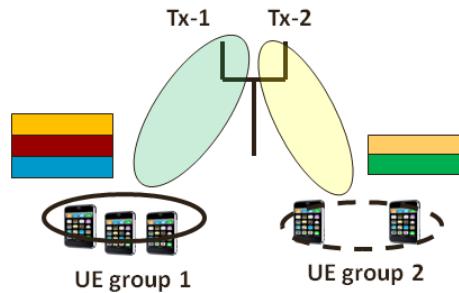


Figure 6-67 BF-CSM for SCMA

DL MU-SCMA

DL MU-SCMA enables open-loop user multiplexing with robustness to mobility and low rate of feedback requirement. With a very limited need for channel knowledge in terms of CQI, a TP simply pairs users together while the transmit downlink power is properly shared among multiplexed layers in code domain. Code domain pairing along with near optimal message passing algorithm (MPA) detector provides a flexible multiplexing and UE pairing solution. Multiplexed layers are separated and decode at the terminals using MPA receiver. Compared to 3GPP closed-loop MU-MIMO, DL MU-SCMA as an open-loop multiplexing scheme is more robust against dynamic channel variations in high speed scenarios and less sensitive to the error in channel information feedback. DL MU-SCMA is evaluated to show the advantage of SCMA user pairing to increase throughput of downlink for a highly loaded network for both low and high speed users.

DL MU-SCMA/CoMP

SCMA can provide an open-loop CoMP solution without knowledge of short term multi-TP CSI. It can bring two main advantages to the system: i) dramatic reduction of the overhead caused by dynamic multi-TP CSI feedback, and ii) significant increase of cell average throughput and cell edge throughput even for vehicular users.

In open-loop inter-TP and intra-TP SCMA CoMP, different SCMA codebook sets are assigned to different TP antennas. Each transmit antenna uses a specific codebook set to multiplex UEs. Terminals jointly detect the signals from multiple TPs within their CoMP coordination cluster. The cluster size depends on the network topology. A terminal may suffer interference from a bunch of neighbouring TPs. This makes inter-TP interference management more challenging. SCMA CoMP provides a flexible and simple multi-TP joint transmission solution to manage interference and improve cell edge throughput.

As shown in Figure 6-68, a neighbouring TP can be either cooperating or interfering from a UE point of view. In the cooperating case, as illustrated in Figure 6-68(a), the strong signal from a neighbouring TP can target the same UE. Open-loop joint-transmission CoMP is implemented if the serving and cooperating TPs serve the same UE. The aim is to improve the coverage for cell edge users. The target UE jointly detects the SCMA layers receiving from multiple serving TPs. In this type of CoMP, the data of a UE needs to be available at multiple serving TPs. Hence, it needs further backhaul traffic requirement which may not be available in every network infrastructure. However, this scheme can facilitate handover which can be useful for TC2 and TC8 scenario in which fast and more frequent handover is required for very dense or high speed networks.

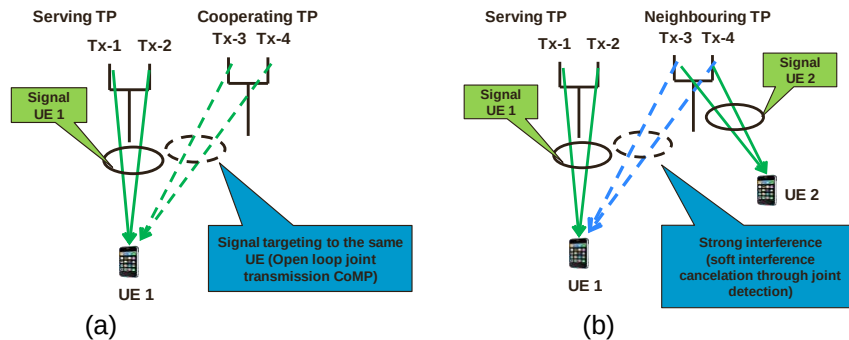


Figure 6-68 (a) Open-loop joint-transmission using SCMA CoMP, (b) Interference cancellation through joint detection using SCMA CoMP

As an alternative SCMA-CoMP solution, the target of the neighbour TP can be a different UE as shown in Figure 6-68(b). TP 1 and 2 serve their corresponding users UE1 and UE2, respectively. UE1 receives a strong interference from its neighbouring TP. In this scenario, UE1 conducts joint detection over its receiving signal and strong interference. The outcome is soft cancellation of interference and better coverage and rate for UE1.

As an enhanced mode of SCMA CoMP, joint transmission and multi-user SCMA can be combined to improve cell edge and cell average throughput both together as illustrated in Figure 6-69.

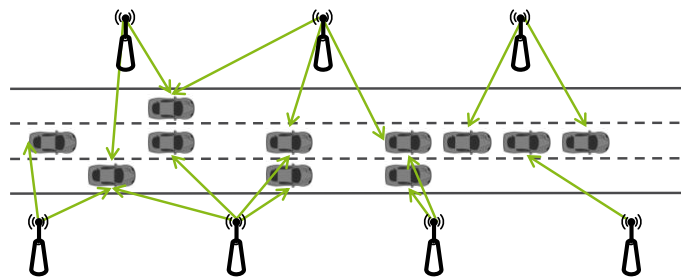


Figure 6-69 SCMA CoMP and inter-TP MU-SCMA layer sharing

One of the application scenarios for SCMA CoMP is in UDN as shown in Figure 6-70. In UDN, multiple TPs serve a UE through multiple SCMA links while SCMA layers are co-ordinately allocated to TPs. A TP may serve multiple users if they have overlapped CoMP clusters. It enables UE-centric CoMP via SCMA layer allocation across multiple TPs. In addition, multiple links to a UE can facilitate soft handover for very high speed users whose frequent handover is a technical challenge. Also, flexible SCMA UE pairing in conjunction with CoMP can provide load balancing gain when the network loading is not balanced among neighbouring cells.

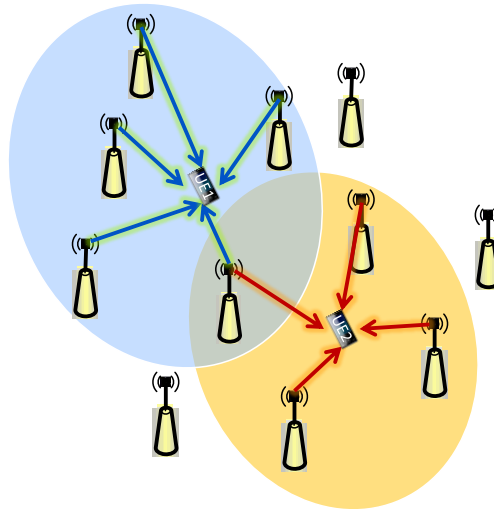


Figure 6-70 SCMA CoMP and inter-TP MU-SCMA layer sharing for UDN

Simulations for DL MU-SCMA and DL MU-SCMA/CoMP

System-level simulation results are provided based on simplified Madrid grid. The evaluation results for DL OFDMA, DL MU-SCMA and DL MU-SCMA/CoMP are provided according to METIS TC2 and TC8 scenarios. Further details on simulation setup, assumptions and results can be found in [METIS D6.5].

The relative cell average throughput and cell-edge throughput gains of MU-SCMA over OFDMA are shown in Figure 6-71 for low and high speed cases for full buffer traffic. As illustrated in this figure, the relative gain is stable regardless of the user speed. The relative gain is between 23-39% for SM mode and 48-72% for Alamouti mode. These results confirm the capability of MU-SCMA to provide high throughput and high quality of user experience independent of the user mobility status and their speeds.

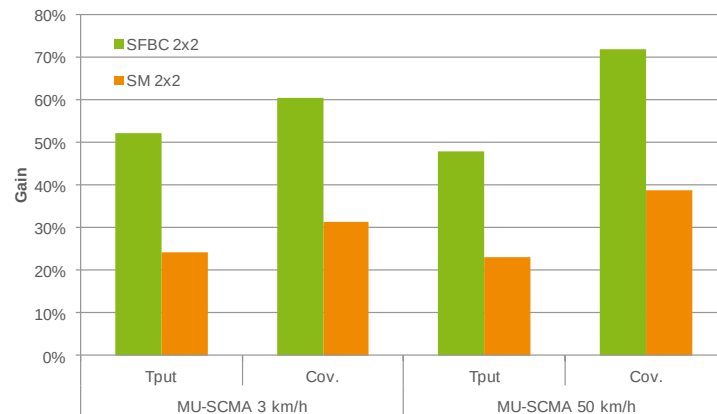


Figure 6-71 Performance gain of DL MU-SCMA over DL OFDMA for low and high speed scenarios with full buffer traffic

For the bursty non-full buffer traffic case, the relative gain of DL MU-SCMA over DL OFDMA is shown in Figure 6-72 in terms of the total supported network load for the required percentage of successful packet delivery with 0.5s delay constraint. As illustrated in this figure, assuming 0.5s delay requirement for 95% of packets, MU-SCMA can support more than 21% higher load compared to OFDMA.

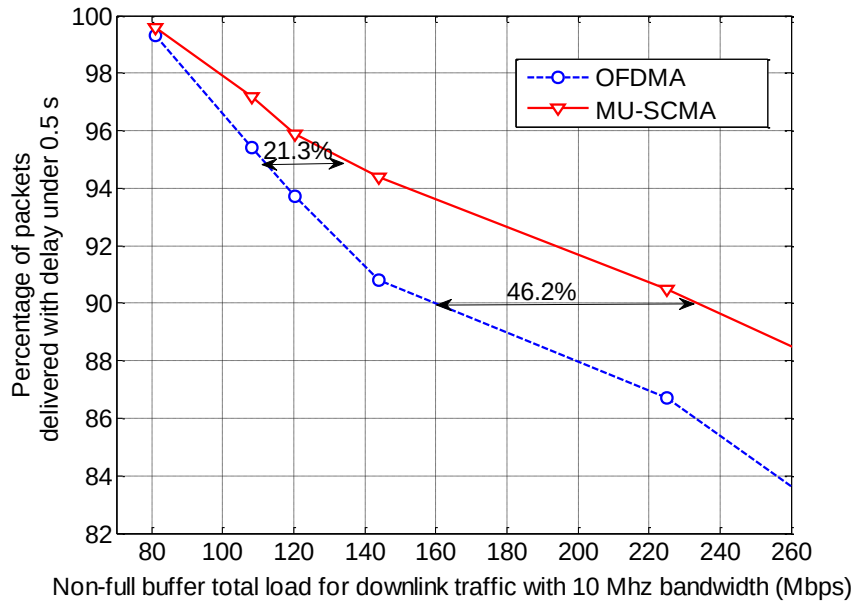


Figure 6-72 Performance gain of MU-SCMA over OFDMA for low speed scenario and SM mode with non-full buffer traffic

To see the benefit of SCMA in CoMP scenario, DL MU-SCMA/CoMP is evaluated against OFDMA baselines. The evaluated CoMP scenarios are illustrated in Figure 6-73.

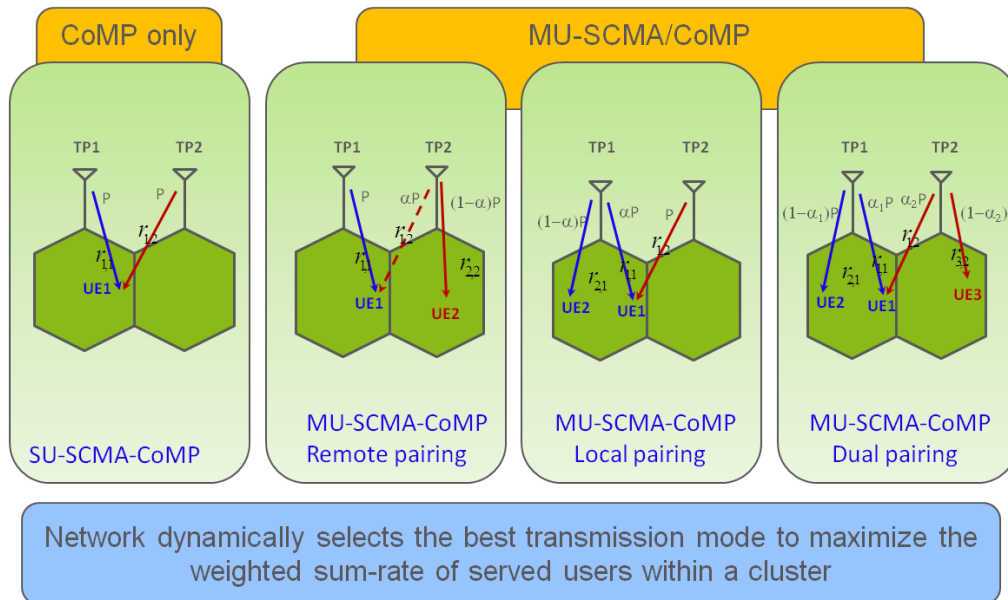


Figure 6-73 CoMP scenarios considered for performance evaluation

Simulation results in terms of cell average throughput and cell-edge throughput gains are illustrated in Figure 6-74 for full buffer traffic with MIMO transmission mode for low speed (3km/h) and high speed (120 km/h) scenarios. It can be observed that the gain of DL MU-SCMA/CoMP over OFDMA and OFDMA-CoMP is high (30-35% throughput gain and 65-75% coverage gain), for both low and high speed users. Furthermore, the gain of DL MU-SCMA/CoMP over OFDMA baselines is even higher for high speed scenario which implies MU-SCMA/CoMP is more robust against the user mobility compared to OFDMA.

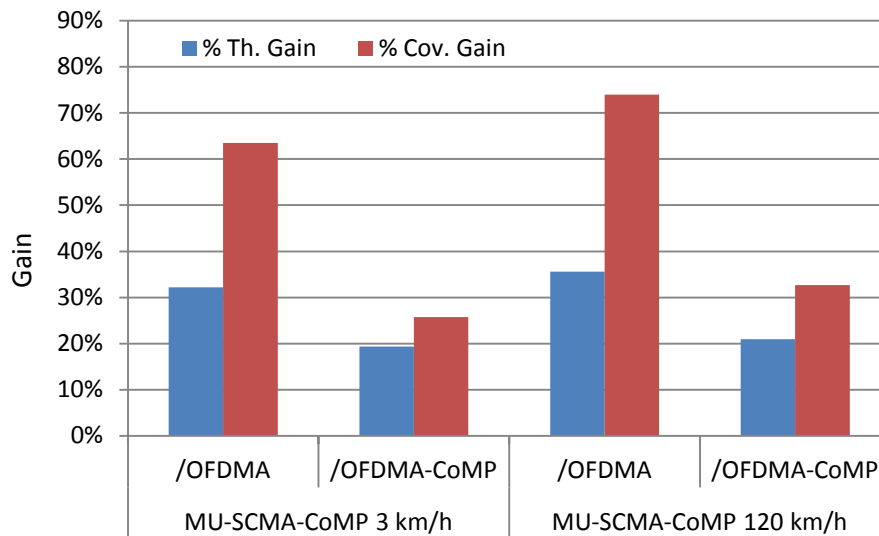


Figure 6-74. Gain of MU-SCMA CoMP over OFDMA baselines for low speed and high speed scenarios

6.11.2 FBMC based multiple access and Cognitive Radio

6.11.2.1 Multiple Access for MIMO FBMC systems

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM1 (MAC spectral efficiency), addressed in terms of the PHY metrics: SNR GRM5 (MAC packet error rate), addressed in terms of the PHY metrics: BER
Baseline for performance comparison	ZF MIMO
Update of the results w.r.t D2.3	New results
Revolution or evolution	Revolution, as FBMC is revolutionary (see TeC#8.1)
Targeted test cases (TC)	TeC has not been selected, but could be useful for the following TC(s): TC2 (targeting high data rate for multi-user)
Impacted HTs	TeC has not been selected, but could be useful for the following HT(s): MN, UDN, MMC
System aspects addressed	xMBB, as high data throughput in multi-user environment is targeted
Cross-connection to other TeC clusters	TeC#8.1 - FBMC, as the assumed waveform is FBMC.

Contribution to METIS overall goals	
1000x data volume	Multi-user FBMC can be applied to address the goal of higher data volume
10-100 user data rate	Multi-user FBMC can address higher data rate per user.
10-100x number of devices	Multi-user FBMC can increase the number of supported devices.

Further evaluation results:

The Figure 6-71 shows the performance for the downlink with a different no. of users, N_u with a single transmitter [SRL14a]. The number of subcarriers is 2M. For the precoding methods in the downlink, perfect CSI is considered to be available at both the transmitter and the receiving users. The average BER of the MIMO FBMC system with THP precoding to cancel the interference is presented. The performance without precoding is plotted for comparison. We note that the precoding technique has reduced the error floor in the MIMO FBMC system. However, this does not offer significant performance improvement over ZF. This is because the precoding technique cannot completely remove the interference. Further, we can see that the increase in the number of users has not affected the performance significantly, which was the case for the uplink. This is because in the downlink, at the receiver, the time domain fading of the channel is the same in all subcarriers, irrespective of the number of users sharing the subcarriers. Hence, the level of inter-carrier interference is unaffected by number of users- N_u , unlike in the uplink. We can consider THP and ZF with channel allocation considering interference where it is shown that BER improves significantly [SRL14a].

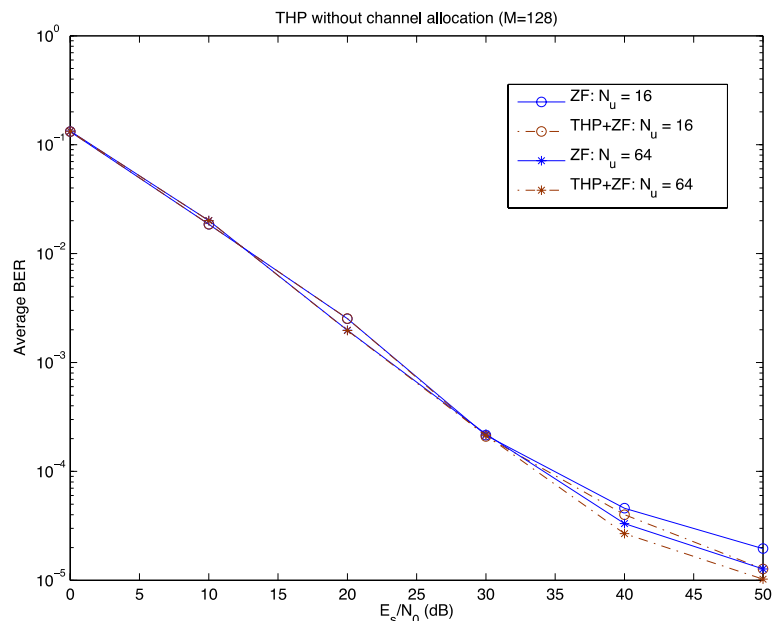


Figure 6-75 BER vs. SNR for different receivers, No. of taps in channel (Δ)=2, No. of subcarriers 2M=256, No. of users - N_u , L=1023, No. of antennas=2

6.11.2.2 MA using cognitive radio

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRT2 : Centre Frequency Range (CFR) Achieved : 470MHz – 790 MHz
Baseline for performance comparison	The performance comparison is based on the CR enabled secondary user. When used for LTE devices baseline standard is LTE Rel-11
Update of the results w.r.t D2.3	Implemented a CR enabled WLAN transceiver. CR functionality across standard power level inputs and CRF : 470MHz – 790MHz
Revolution or evolution	CR brings significant changes in system design and can be considered as Revolution.
Targeted test cases (TC)	TC10: Emergency Communication TC7: Blind spots
Impacted HTs	TeC has not been considered for the MN system concept. (Can be considered for Flexible Spectrum use)
Cross-connection to other TeC clusters	TeC#3 is the technology driver for the high-IF concept.
Contribution to METIS overall goals	
Energy efficiency and cost	With the addition of high-IF converter, commercial application can be CR enabled ensuring flexibility and faster implementation with reduced cost.

6.12 Medium Access Control

6.12.1 Contention based massive access

6.12.1.1 Combined TeC#12.1.1 and TeC#12.1.3 approach

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM5#PER (TeC#12.1.3): random access with nearly the PHY performance of “scheduled access” GRM2#RASE (TeC#12.1.1 & #12.1.3): approximate gain of 3-10x compared to LTE Rel11 baseline
Baseline for performance comparison	PHY performance in terms of GRM5#PER is compared to perfect activity knowledge modelling scheduled access. GRM2#RASE is compared to LTE Rel. 11 PRACH performance determined by the probability of collisions in the RRC connect request (see 3GPP TS 36.211 and 36.213) under the assumption that no collisions can be detected in the first step 3GPP TS 36.211 and 36.213

Update of the results w.r.t D2.3	Novel result for TeC#12.1.3 is the scalability shown by GRM5#PER results in terms of MA design and access load. Novel results are the combination of TeCC#12.1 TeCs “Coded Random Access” and “Advanced physical layer processing for enhanced MAC: Joint detection of node activity and data” with random PN UE ids/CDMA spreading sequences and with a limited codebook of PN spreading sequences thereby introducing additional PHY collisions.
Revolution or evolution	If PRACH resources in LTE are provided to facilitate this new access scheme, it could be combined with an evolution of LTE. However, this would probably require major changes in connection handling. Therefore, a revolutionary approach is very likely required and needed to support MMC.
Targeted test cases (TC)	TeC has been selected for TC evaluation of the following TCs: TC11 TeC has not been selected, but could be useful for the following TC(s): TC2, TC3, TC5, TC9 (Access Reservation with same system as MMC)
Impacted HTs	TeC has been selected for the HT system concept: MMC, URC
System aspects addressed	TeC#12.1 mainly addresses “massive MTC” but is also relevant for “ultra-reliable MTC”
Cross-connection to other TeC clusters	TeC#11.1.2 (SCMA) is a candidate technology to replace CDMA TeC#8.1(FBMC) /#8.2 (UFMC) are enablers to provide MMC access bands, where TeCC#12.1 could be employed.
Contribution to METIS overall goals	
10-100x number of devices	Potential 3-10x increase over LTE PRACH under different idealizing assumptions about MAC and PHY.
10x longer battery life	Indirectly targeted through minimization of signalling overhead and active transmit times (e.g. less slots required, see above).
5x E-E reduced latency	Indirectly targeted through minimization of signalling overhead and optimization of PRACH procedure (e.g. less slots required, see above)
Energy efficiency and cost	Energy indirectly targeted through minimization of overall PRACH and data transmission signalling overhead for short messages. Cost indirectly affected through minimum complexity solution at UE.

Further evaluation results

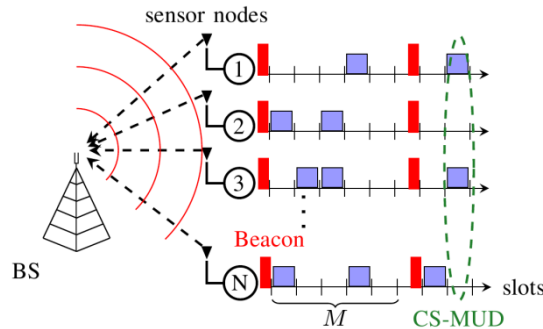


Figure 6-76 General setup of Coded Random Access with CS-MUD. BS sends a beacon to start communication, UEs send in slots until contention period ends at the next beacon. In each slot data processing according to Figure 6-77 takes place.

The combination of Coded Random Access and CS-MUD shortly characterised in section 3.12.1 uses a unique data processing flow to successfully decode users over all received slots. Overall the communication scheme is set up like indicated as in Figure 6-76, where the BS starts a MAC contention period with some form of beacon, then UEs send data randomly in a number of slots leading to collisions. The degree of activity of all UEs is predetermined by the BS in the beacon but not the activity pattern per UE over the slots. The contention period ends once the BS sends a new beacon. In each slot data processing takes place as shown in Figure 6-77. First, PHY processing is facilitated by CS-MUD and standard FEC decoding to recover the UE data. A CRC check ensures correct packets, incorrect packages are assumed to be discarded in this example. If a UE's packet is correctly decoded, it will immediately be subtracted from the received signal at the current slot and PHY processing may be repeated until no new UEs are successfully recovered. All successfully decoded UE packages are then stored for inter-slot interference cancellation, which takes place once processing in the next slot starts. This way PHY and MAC processing work hand in hand to recover as many UE packets as possible until the throughput is maximised.

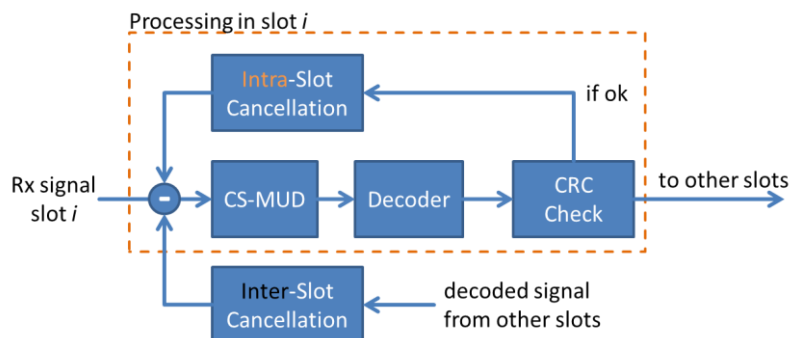


Figure 6-77 Data processing flow of Coded Random Access with CS-MUD

The overall performance of the described scheme is strongly dependent on the performance of the PHY processing which is depicted in Figure 6-78 in terms of the symbol error rate (SER) before decoding. The SER is defined as the total number of errors, including activity decisions, over the total number of symbols, including “zero symbols” for inactive users. Each active UE, i.e. each collision, in this CDMA example leads to more interference that has to be suppressed. Consequently, performance degrades with increasing number of active users in a slot.

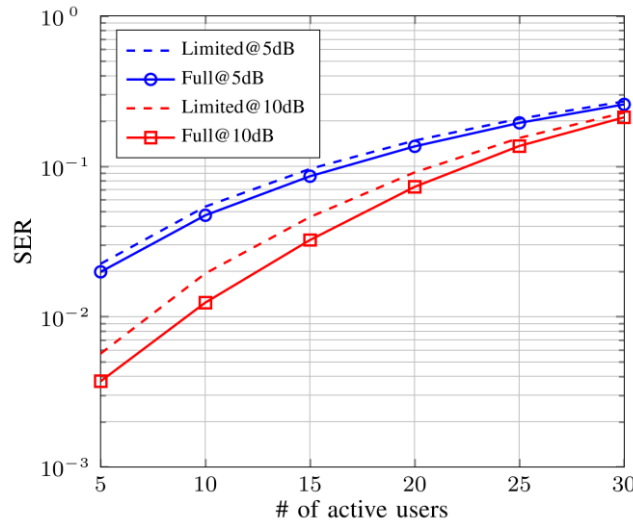


Figure 6-78 Symbol Error Rate (SER) vs. # of active users

To quantify the impact of this SER loss in terms of the MAC performance, however, one needs to consider the capture probability, which describes the probability of successfully decoding a specific (but arbitrary) user out of all users currently active in a slot. Figure 6-79 illustrates the capture probability for this exemplary setup in terms of a full PN sequence codebook as well as limited codebook build by 128 random PN spreading sequences of length 32 at an SNR of 5dB and 10dB. This result already includes the intra- and inter-slot interference of the MAC scheme and shows the very robust performance at 10 dB independent of full or limited codebook. At 5dB, however, the capture probability is much lower and also strongly affected by the introduction of PN collision if sequences are drawn from a limited codebook. The results in Figure 6-79 are clearly the reason why the overall GRM2#RASE results shown in Section 3.12.1 are nearly unaffected at 10dB but strongly affected at 5dB.

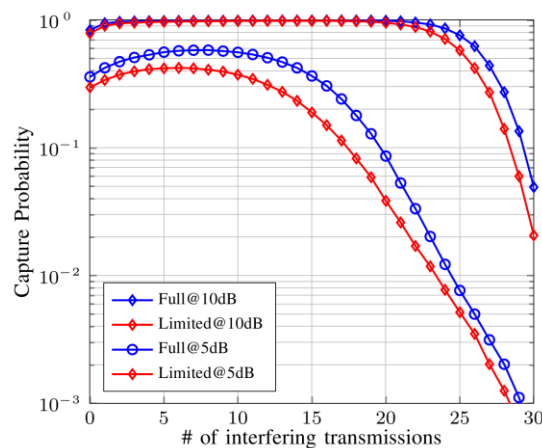


Figure 6-79 Capture Probability vs. # of interfering users resulting from Figure 6-78.

LTE Comparison

As a comparison the LTE PRACH performance can be calculated by the number of average users served without collision at the RRC message stage (cf. 3GPP TS 36.211 and 36.213) assuming that collision are undetectable for the first PRACH message and neglecting any preamble detection errors through channel or noise influences. Then, the average number of collision free users is given by $N * \left(1 - \frac{1}{B}\right)^N$, where B is the number of orthogonal resources. Given 3 slots (see TeC#12.1 results in section 3.12.1) and 64 Zadoff-Chu Sequences per

slot (cf. 3GPP TS 36.211 and 36.213) clearly $B = 192$ results. Therefore, LTE can serve roughly 65 users out of $N = 128$ overall users given 192 orthogonal resources. However, the access is organized in two phases, thus requiring overall 3 PRACH slots and 3 PRBs for the whole PRACH procedure, thereby lowering the average number of users per slot to approximately $GRM2\#RASE=10.9$ without data transmission. Furthermore, LTE allocates 800 μ s and 1.08 MHz to PRACH slots with Zadoff-Chu sequences of 839 symbols. In this example TeC#12.1 requires 3328 symbols, roughly 4 PRACH slots. Finally, TeC#12.1 transmits data during random access which requires at least 2 PRB (minimum size in scheduling) per user in LTE after access reservation. Additionally, considering the wasted resources in a PRB for very small messages as denoted above, TeCC#12.1 at 10dB offers a net gain of 3x-10x higher number of served users per time-frequency resource including data transmission. Note, that the LTE PRACH performance is still highly idealized due to the lack of modelling for preamble detection, channel influences and simplifications regarding the occurrence of collisions. Furthermore, only a single exemplary operating point has been highlighted, the joint performance of Coded Random Access with CS-MUD for multiple scenarios is still under investigation. However, the gains over LTE will likely increase with an increasing number of users as LTE PRACH gets increasingly inefficient.

6.12.1.2 Coded access reservation

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM2#RASE (Normalized System Efficiency/Throughput) approaches one for very high loads, while the baseline tends to zero.
Baseline for performance comparison	LTE Rel. 11 PRACH performance is determined by the probability of collisions in the RRC connect request (see 3GPP TS 36.211 and 36.213) under the assumption that no collisions can be detected in the first step 3GPP TS 36.211 and 36.213
Update of the results w.r.t D2.3	No updated results
Revolution or evolution	Can be introduced in a future release of LTE (evolution), since it reutilizes the LTE physical layer (RACH preambles) although requiring new feedback message structure.
Targeted test cases (TC)	TeC has been selected for TC evaluation of the following TCs: 11
Impacted HTs	TeC has been selected for the HT system concept: MMC and URC
System aspects addressed	This TeC addresses massive MTC and to some degree ultra-reliable MTC.
Cross-connection to other TeC clusters	



Contribution to METIS overall goals	
10-100x number of devices	Can support as many devices as required with trade off in latency
10x longer battery life	Indirectly targeted through minimization of signalling overhead and active transmit times
5x E-E reduced latency	Indirectly targeted through minimization of signalling overhead and optimization of PRACH procedure
Energy efficiency and cost	Energy indirectly targeted through minimization of overall PRACH and data transmission signalling overhead for short messages. Cost indirectly affected through minimum complexity solution at UE.

6.12.2 Distributed network synchronization

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM2: MAX#-CONN, RASSR, RASE 1) with time agreement, more effective channel access mechanisms and synchronous transmissions can be used to support higher number of devices; 2) by setting a threshold on synchronization accuracy and a threshold convergence time, TeC#12.2 can support a higher number of devices compared to the baseline
Baseline for performance comparison	TSF in 802.11
Update of the results w.r.t D2.3	Yes – by considering both out-of-coverage and partial-coverage cases
Revolution or evolution	Evolution
Targeted test cases (TC)	TC has not been selected, but could be useful for TC12, since it requires time agreement when using time related channel access mechanism.
Impacted HTs	Moving networks, device-to-device. MN: time related channel access requires time agreement. When some vehicles are out of coverage and have not good enough GPS, network synchronization is needed. D2D: synchronous transmission and device discovery require time agreement. When some devices are out of coverage, network synchronization is needed
System aspects addressed	Massive MTC Indirectly ultra-reliable MTC

Cross-connection to other TeC clusters	TeC#6.5 (Ad-hoc MAC for V2V), which requires time agreement
Contribution to METIS overall goals	
10-100x number of devices	Yes. With time agreement, more effective channel access mechanisms and synchronous transmissions can be used to support higher number of devices
5x E-E reduced latency	Indirectly targeted by enabling direct V2V or D2D communications that can reduce latency
Energy efficiency and cost	Indirectly targeted by enabling direct V2V or D2D communications that improve energy efficiency compared to communications over traditional cellular links

Further evaluation results:

The main principles of the proposed ARES are as follows.

1. Besides leader and follower, we define a new identity of nodes ---- pseudoleader. In particular, a follower will become a pseudoleader after it synchronizes to the leader(s). Furthermore, leaders and pseudoleaders are endowed with different hierarchical numbers. Specifically, all the leaders have hierarchical number 1; the hierarchical number of a pseudoleader equals to
 $[1 + \text{mean of (used) transmitters' hierarchical numbers}]$.
2. The transmission mechanism at each node is a modified version of the contention based broadcast protocol specified in 802.11 [IEEE802.11]. The modification is: when a leader or a pseudoleader appears, it decreases its contention window by a factor of W_D for the first T_D synchronization rounds. In this way, leaders' clock value can be spread in a faster manner, which handles **challenge 2**.
3. For partial-coverage synchronization scenario, each follower or pseudoleader firstly collects N_E timestamps from other (pseudo)leaders with smaller hierarchical numbers, then estimates the leaders' frequency and offset and updates its own hierarchical number. Compared to the traditional tree-based synchronization scheme where a node synchronizes to its only parent node, the information from all possible (pseudo)leaders is exploited in ARES. As a result, leaders' clock frequency and offset are spread quickly, which tackles **challenge 2**. Moreover, with the proposed simple and effective estimation algorithm, **challenge 5** can be handled.
4. For out-of-coverage scenario, the consensus-based synchronization algorithm proposed in [SSBG15] with improved weight design is used. Specifically, we improve the consensus algorithm in [SSBG15] by giving different weights to different nodes, where the weights depend on local counters denoting the number of local clock updates. This feature can solve **challenges 3 and 4**.
5. Each follower implements the consensus-based synchronization algorithm specified in principle 4 before it receives a timestamp from any (pseudo)leader. On the other hand, the follower ignores the timestamp from any other follower after it receives a message from any (pseudo)leader. By doing so, **challenge 1** can be solved.

In simulations, we consider a randomly time-varying network with N nodes, where the connectivity degree of each node is assumed to be 15. Moreover, initially, the clock

frequencies are uniformly and randomly selected from the range [0.9999, 1.0001] Hz by following [IEEE802.11]. Besides, in the simulations, we only assume 1 leader for the proposed ARES scheme. In fact, the performance of ARES can be further improved when multiple leaders exist.

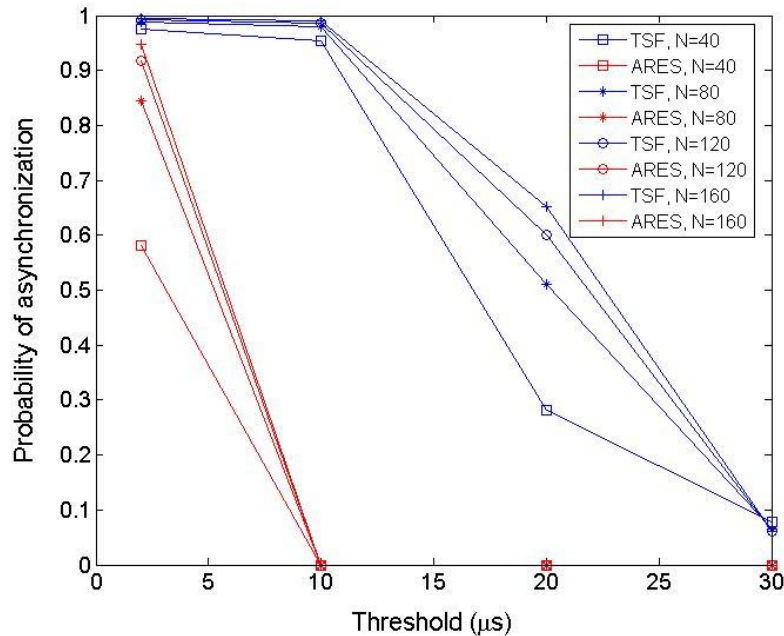


Figure 6-80 Probability of asynchronization versus accuracy threshold

In Figure 6-80, when assuming the existence of transmission delays, the probabilities of asynchronization are plotted with respect to different accuracy thresholds and different number of nodes for TSF and ARES. Again, ARES shows superiority over TSF regarding both accuracy and robustness to the number of nodes. In particular, when setting the accuracy threshold as 10 microseconds, the ARES can achieve the synchronization through the entire network where the number of nodes scales from 40 to 160. However, the TSF has very poor performance with more than 90% asynchronization probability.

6.12.3 MAC for UDN and mmW

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM1: (MU/C-TP & MAC-SE) Increase system throughput with increased network density, with use of high gain beamforming and wireless self-backhauling.
Baseline for performance comparison	LTE Rel11 Scheduled recourse
Update of the results w.r.t D2.3	Brief summary of novel results are detailed in the text and annex.
Revolution or evolution	The contention based MAC would require a redesign and hence should be seen as a revolution.
Targeted test cases (TC)	The TeC has been selected for evaluation of the TC3.

Impacted HTs	TeC has been selected for the HT system concept: UDN
System aspects addressed	The TeC is applicable to the main system aspect xMBB (eXtreme Mobile BroadBand)
Cross-connection to other TeC clusters	TeCC#1 provides air interface support.
Contribution to METIS overall goals	
1000x data volume 10-100 user data rate	10-100x: via providing TeCC#1 with a MAC protocol
5x E-E reduced latency	The contention based MAC realizes significant latency reductions.
Energy efficiency and cost	Via wireless self-backhauling, and high gain beamforming

Further evaluation results:

A MAC approach with contention control signalling to allocate data resources provided by the resource coordination functionality is carried out and evaluated with the METIS test case 3.

Control signals, *transmission reservation request* (TxRR) and *resource confirmation* (RC), are transmitted in time-frequency resources separated from the data resources, using directive beamforming allowing only the set of other nodes that will be interfered by the transmission to overhear the resource reservation.

Every node keeps a “local view” (a local look-up register) of the spectrum environment. This “local view” represents the present and near future use of the communication resources. It is kept in order to plan future transmissions. Once a node overhears a reservation request, a TxRR, it will determine if the reservation request will be incorporated into the “local view” or not. This is handled via a threshold that may be set dynamically. Depending on the propagation environment and system throughput it may be beneficial to not include all overheard TxRR messages into the “local view”.

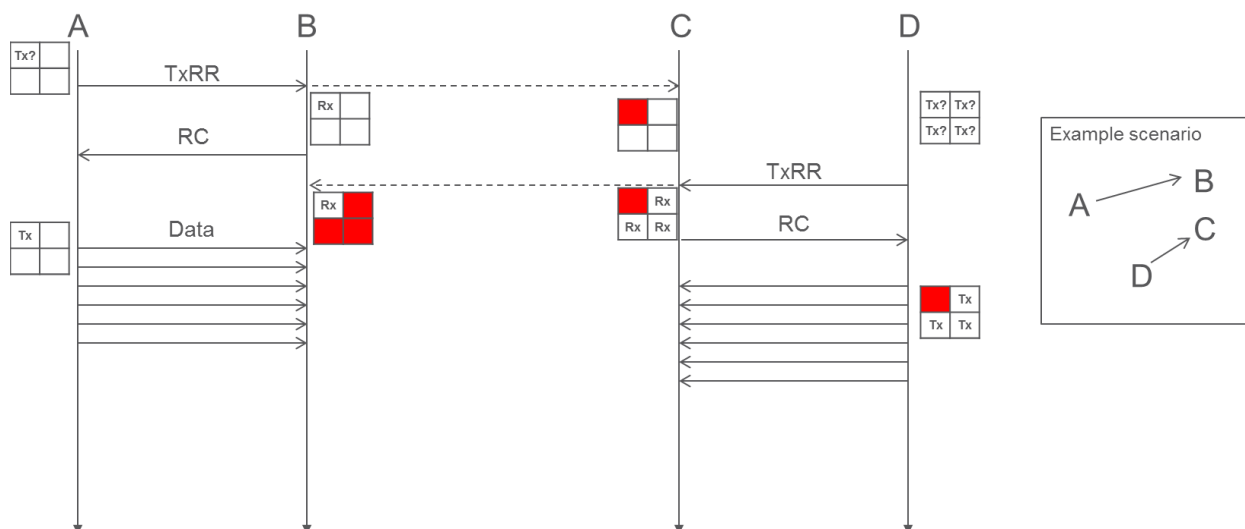


Figure 6-81 An overview example of the contention signalling in the contention based MAC protocol. Time runs downwards in the illustration.

Figure 6-81 is an example illustration of TxRR-RC message exchange and the impact on the “local views” kept by the nodes. The tables represent the “local view” of the respective node, and the upper left resource is denoted (1,1).

The example setting is that node A wants to transmit data to node B, node D wants to transmit data to node C.

The procedure in Figure 6-81 is as follows:

1. Node A want to transmit to B using resource (1,1) at a given time interval
2. It sends a TxRR specifying the resource and time
3. Node B receives the transmission request and updates its “local view” of the resource usage.
4. Node C overhears the reservation request and updates its “local view”
5. Node B checks its local view if the transmission from A is ok to receive, and as it in this example is so, it transmits a RC to Node A.
6. Node D sends a TxRR to Node C indicating a reservation of all resources for a time interval overlapping with the planned transmission by node A.
7. Node C and node B receives this information and updates their “local views” accordingly.
8. Since node C has knowledge of the conflicting transmission of node A it transmits a RC to node D indicating the free resources.
9. Node D updates its “local view” according to the info from node C.
10. Node A starts to transmit the data on the data channel and using resource (1,1).
11. Node D adjusts its planned transmission to avoid interfering with the transmission from node A and starts transmitting data to node C using the remaining resources.

The following simulations were performed to evaluate the METIS TC3 scenario, and designed to meet its KPI requirements [METIS D1.1].

The simulated environment in TC 3 is an open area “food court” with 6 m up to the roof, as shown on Figure 6-82. Two topologies were considered, one with only 6 AgNs, and another one with the same 6 AgNs plus 4 extra wireless ANs. In the figure, red crosses corresponds to the possible user locations, orange dots are AgNs and green dots are wireless ANs. Blue stars are unused possible AN locations.

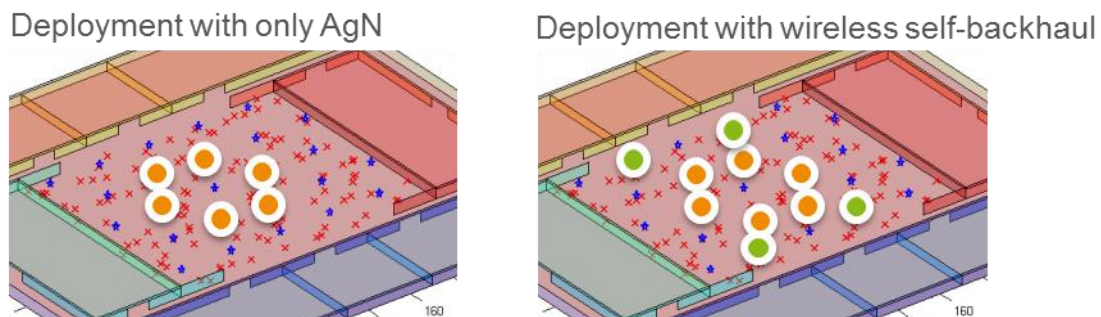


Figure 6-82 Example deployments

Simulation set up:

- System level simulation
- Wireless self-backhauling
 - Access nodes with in-band wireless self-backhauling to aggregation nodes
- The contention based MAC protocol controls all nodes (UEs, ANs, wireless ANs)
- Traffic Model:
 - Total throughput target: 5.6 TB/h
 - 82 files (20MB) per second (Poisson arrival)
 - One file per user
 - 70% Downlink, 30% UL
 - "UDP" file transfer
 - The MAC protocol controls the transmissions
- Users are deployed when they have a file
 - Static users only (the user is static during the download/upload – less than a second)
 - User position randomly selected from 128 available locations
- Ideal beam forming

The CDF of user's file transfer rate is shown in Figure 6-83. The delay measure is taken from the creation of the file demand until its final completion (including attachment procedure, retransmissions and forward of the wireless Aps). The simulation results are shown in Figure 6-84.

It is shown that the METIS TC3 throughput and file delay transfer KPIs are met by the MAC (cf. Table in section 3.12.3).

It is worth to note that overall, the difference of user throughput between the scenarios with only AgNs and the one with added wireless ANs is not very important. However, it can be seen that the uplink performance is increased when the wireless ANs are added, while it is reduced when there are only AgNs. The improvement on the uplink is particularly visible on the 5th percentile user, since the addition of wireless AN improved its performance by 50%. This gain is mainly due to the ease of access to the resource, since adding access nodes reduced the load per cell, which is more critical for uplink (heavier procedure).

Note that in this set of simulations we did not assume the latest version of the frame structure in the sense that we require a separate frequency field for the contention control signalling instead of a separate time field which will be evaluated.

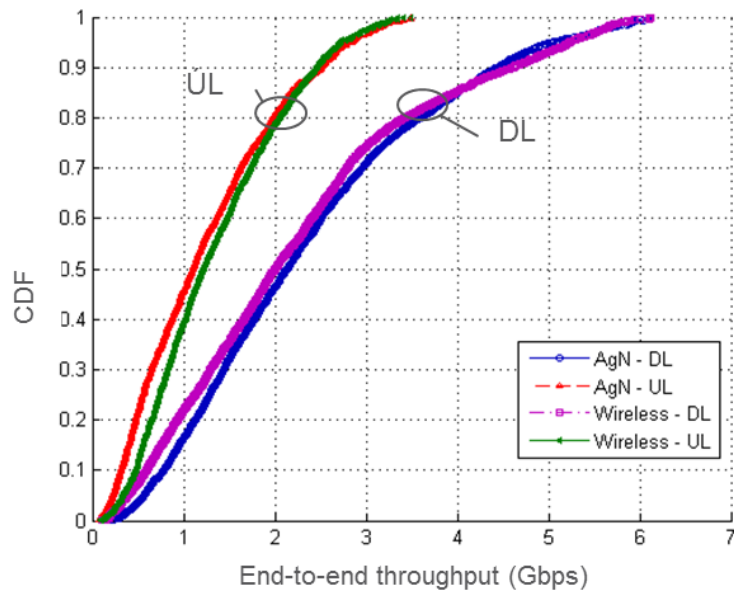


Figure 6-83 CDF of file throughput

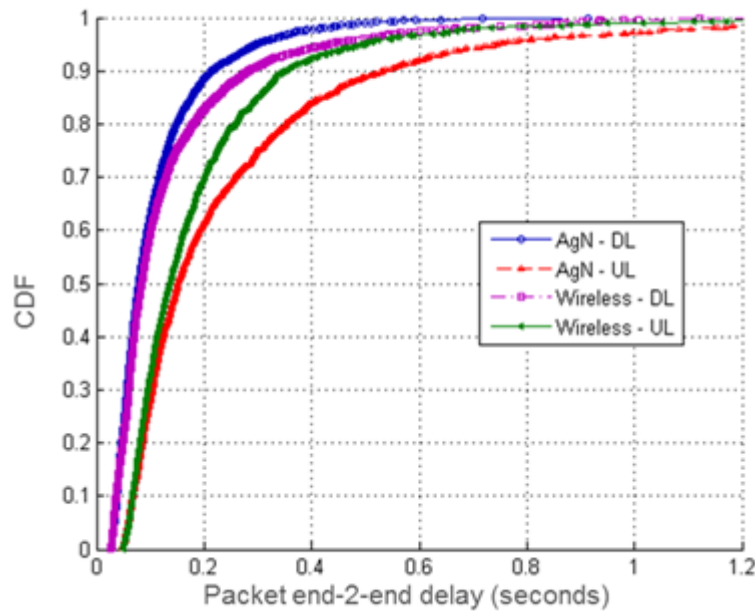


Figure 6-84 CDF of User file transfer delay

6.13 Hybrid Automatic Repeat Request (HARQ)

6.13.1 Backtrack Retransmission with Ternary Feedback

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	GRM1 (Throughput) and GRM4 (Latency) Approximately 2x (throughput and latency).
Baseline for performance comparison	Without HARQ
Update of the results w.r.t D2.3	It was not present in D2.3
Revolution or evolution	Revolution, since feedback mechanism and backtrack decoding needs to be added to the system.
Targeted test cases (TC)	TeC has not been selected, but could be useful for the following TC(s): TC 2, but others also applicable since it is an HARQ technique.
Impacted HTs	TeC has been selected for the HT system concept: URC
System aspects addressed	This TeC is applicable to xMBB (extreme mobile broadband) and ultra-reliable MTC.
Cross-connection to other TeC clusters	Not applicable
Contribution to METIS overall goals	
10-100x number of devices	It allows to reduce the amount of signalling (CSI-T) required to perform link adaptation.
10x longer battery life	Indirectly, since MTD can connect to the cellular network via MT-D2D, therefore the MTD will potentially require less power.
Energy efficiency and cost	Indirectly, since MTD can connect to the cellular network via MT-D2D, therefore the MTD will potentially require less power.

6.13.2 Reliability-based HARQ

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	For data transmissions: ~30% gain achievable in cell Tput, user Tput and energy efficiency. Average transmit power for ACK/NACK: ~1.6-1.9dB gain for 2bit ACK/NACK transmission with A-QPSK over QPSK.
Baseline for performance comparison	LTE



Update of the results w.r.t D2.3	Added assessment of Tput and delay performance of reliability-based HARQ. A-QPSK performance over LTE PUCCH, whereas D2.3 considered AWGN channel only.
Revolution or evolution	Evolution: can be implemented in LTE e.g. by replacing the 1bit ACK/NACK of HARQ by a 2bit ACK/NACK.
Targeted test cases (TC)	TC2: Dense urban information society (potential for cellular throughput enhancement) TC8: High mobility environment (assessed performance for user velocities up to 250km/h)
Impacted HTs	MN: assessment for the higher UE velocities
System aspects addressed	xMBB (extreme mobile broadband) ultra-reliable MTC
Cross-connection to other TeC clusters	Reliability-based HARQ w/ multi-level ACK/NACK can be applied with waveforms other than OFDM (TeCC#8) while similar performance advantages are expected. Reliability-based HARQ w/ multi-level ACK/NACK can also be applied with novel frame formats such as those proposed for UDN (TeCC#1), where performance would be for further study.
Contribution to METIS overall goals	
1000x data volume	~30% cell Tput gain by reliability-based HARQ.
10-100 user data rate	~30% user Tput gain by reliability-based HARQ.
10x longer battery life	~30% energy efficiency gain for same data Tput by reliability-based HARQ.
5x E-E reduced latency	For traffic streams involving a larger number of packets (e.g. large file downloads or video streaming), ~30% reduction in over-the-air latency due to ~30% Tput increase. For single packets (e.g. small payloads for voice or M2M traffic), small increase in over-the-air latency from first transmission of a packet until successful decoding of the packet, say 0.8-1.6ms on average in LTE case.
Energy efficiency and cost	~30% energy efficiency gain for same data Tput by reliability-based HARQ.

Further information concerning the research work performed in the area of reliability-based HARQ can be found in [BDZ+14] and [WWD+14].

6.14 Radio link enablers for Radio Resource Management (RRM)

Technology component profile	
KPIs, GRMs and GRTs considered and achieved gain	KPI: 1000x data volume GRMs: GRM1 MU/C-TP Achieved gain: 30% for Macro cell and 10% for Micro cells
Baseline for performance comparison	LTE Rel.11
Update of the results w.r.t D2.3	System simulation results are provided within TC2 scenario where heterogeneous deployment is considered.
Revolution or evolution	Evolution
Targeted test cases (TC)	TeC has been selected for TC evaluation of the following TCs: TC2, TC6
Impacted HTs	TeC has been selected for the HT system concept: D2D
System aspects addressed	xMBB (extreme mobile broadband)
Cross-connection to other TeC clusters	This TeC is connected to T4.2-TeC#14 Context-aware smart devices and RATs/layers mapping. The smart signalling algorithm developed here cooperates with RRM algorithms developed in T4.2-TeC#14.
Contribution to METIS overall goals	
1000x data volume	This TeC could contribute to higher data volume by exploiting D2D communications to offload cellular traffic in a smart manner.