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Radio Channel Measurement for Realistic Simulation of Adaptive Antenna Arrays

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***Abstract* — The application of adaptive antenna arrays offers several advantages for the design of new mobile communications systems. For the investigation of array processing algorithms a realistic simulation of the radio channel is required. The basic principle of a real-time channel sounder with an array frontend is presented. The inclusion of measurement data into the simulation of a communications system is discussed and examples are provided.**

I. INTRODUCTION

The demand for wireless data transmission is growing rapidly. New applications such as mobile computing and multimedia with their heterogeneity in data contents and data rates place requirements on air interfaces which by far exceed those of the current digital mobile telephone systems [1]. This concerns network protocols as well as the physical radio transmission. Since the Asynchronous Transfer Mode (ATM) is a well established standard in wireline networks, suitable for many modern communication applications, a seamless wireless extension of ATM networks is desired. This results in new challenges for the design of an air interface, because lower bit error rates (BER) are required, an application conformed coding (as with speech or video data) is impossible and support for different qualities of service (average data rate, maximum delay etc.) is necessary.

Other difficulties are due to the generally higher data rates resulting in either higher radio bandwidth or more complex modulation techniques. Therefore the transmission is more sensitive to multipath propagation. As an alternative to the utilization of sophisticated equalizer algorithms, the use of adaptive antenna arrays in mobile broadband systems is frequently proposed [1] because it seems to offer several advantages:

- Multipath echoes can be suppressed if they impinge from different directions, resulting in a reduction of the delay spread,
- The Signal to Interference and Noise Ratio (SINR) of the radio link can be increased,
- A space component for multiple access (SDMA) can be introduced, resulting in a higher system capacity.

The wireless modem design requires detailed knowledge of the dispersive, time varying channel transfer characteristics. Link level simulations based on realistic channel modeling allow to compare different receiver concepts without the need for expensive hardware and real-time software implementation [2]. The direct use of a sequence of measured channel impulse responses gives a completely realistic description of a particular situation. Another approach uses ray tracing tools based on real map data to synthesize impulse responses. By far most popular is the application of various statistical channel models.

A well accepted statistical model for the mobile radio channel is the Wide Sense Stationary Uncorrelated Scattering (WSSUS) model. Validity (and limitations) of this model have been proven in a large number of measurements for one-channel transmission [4], [6]. If we use an array of N receive antennas we can measure N different impulse responses. Equivalently, the impulse response becomes vector-valued with N dimensions. Although more and more multi-channel receiver algorithms are being proposed (see for instance ICASSP '97), publications on the verification of vector channel models are sparse. For the simulation of multiple antenna reception, Martin [7] proposed the Directional Gaussian Scattering (DGS) model as an extension of WSSUS which includes directions of arrival (DOA's) for the different multipath components. The basic assumptions of WSSUS, stationarity (with time and/or distance), and uncorrelated scattering (with delay) limit its applicability for the simulation of dynamic algorithms such as adaptive equalization, interference rejection filtering, antenna beam steering, and synchronization. This is especially true for microcells, because of the complicated and rapidly changing field pattern. Realistic simulation would require continuously changing directions, delays, and amplitudes of the different paths due to terminal mobility or scatterer movement [2], [4].

Propagation measurements are a prerequisite for the development of statistical channel models as well as for measurement based simulation. Section II discusses constraints for measurement procedures and gives a short evaluation of known methods. The concept of a new channel sounder to be developed within the project line ATM*mobil* is introduced. This new channel sounder (RUSK ATM) is able to measure up to 8 receive antenna signals quasi simultaneously. Section III motivates and explores the consideration of non-stationarities for array processing simulations. It presents an overview of different methods for adjusting parametric statistical channel models including directions of arrival. Section IV treats the processing steps required for including measurement data into the simulation. Using the system simulation tool PTOLEMY, some details of a radio transmission simulation using an adaptive antenna array are shown.

II. MEASUREMENT METHODS FOR MOBILE COMMUNICATION CHANNELS

This section considers requirements for time-variant channel sounding experiments. It is shown that for typical scenarios time multiplex measurement of several receiver channels is acceptable.

A. Statistical Analysis of Time Variant Multipath Propagation

The input/output relation of the randomly time varying mobile radio channel for any receiver array output i is given by the following linear transformation,

$$y_i(t) = \int_{-\infty}^{\infty} g_i(t,\tau)x(t-\tau)\,d\tau, \tag{1}$$

where $g_i(t,\tau)$ is the complex valued baseband impulse response. Since the impulse response is random, a statistical description is required for system design and simulation and for a proper definition of measurement procedures. A general second order characteristic is given by the six-dimensional correlation function:

$$r_{g_{ij}}(t_1,t_2;\tau_1,\tau_2) = \mathrm{E}\left\{g_i(t_1,\tau_1)g_j^*(t_2,\tau_2)\right\} \tag{2}$$

with the indices i and j denoting the individual impulse responses at the antenna array outputs. Essential simplifications are introduced by the WSSUS model approach. Then $g_i(t,\tau)$ is considered a wide sense stationary random process in t that is uncorrelated with respect to scattering delay time τ. Equivalently in the frequency domain, the time dependent frequency response $G_i(t,f)$ is stationary in both time and frequency [3]. Uncorrelated scattering also implies uncorrelated directions of arrival, corresponding to stationarity in the array aperture area domain. That allows (2) to be reduced to a three-dimensional correlation function $r_{g_{\Delta i}}(\Delta t,\tau)$ according to $r_{g_{ij}}(t_1,t_2;\tau_1,\tau_2) = r_{g_{i-j}}(t_1-t_2,\tau_1)\delta(\tau_1-\tau_2) = r_{g_{\Delta i}}(\Delta t,\tau_1)\delta(\tau_1-\tau_2)$. Here we suppose for simplicity that the geometric Euclidean distance between the antennas is proportional to Δi.

To describe the Doppler dispersion of the individual impulse responses, the expected delay-Doppler spectrum (or scattering function) is introduced by Fourier transform with respect to Δt:

$$\begin{aligned} C_{g_i}(\alpha,\tau) &= \int_{-\infty}^{\infty} r_{g_{ii}}(\Delta t,\tau)e^{-j2\pi\alpha\Delta t}\,d\Delta t \\ &= \lim_{T\to\infty}\frac{1}{T}\mathrm{E}\left\{\left|\int_T g_i(t,\tau)e^{-j2\pi\alpha t}\,dt\right|^2\right\} \end{aligned} \tag{3}$$

The expectation is required since the magnitude squared Fourier transform does not yield a consistent spectral estimate out of a random impulse response record [12]. Fig. 1 and Fig. 2 give examples of $|g_i(t,\tau)|$ and $C_{g_i}(\alpha,\tau)$. A description of this particular measurement follows below.

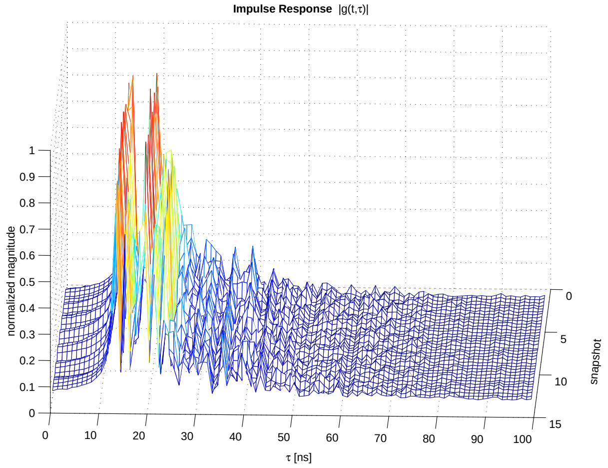

Fig. 1. Magnitude of impulse response

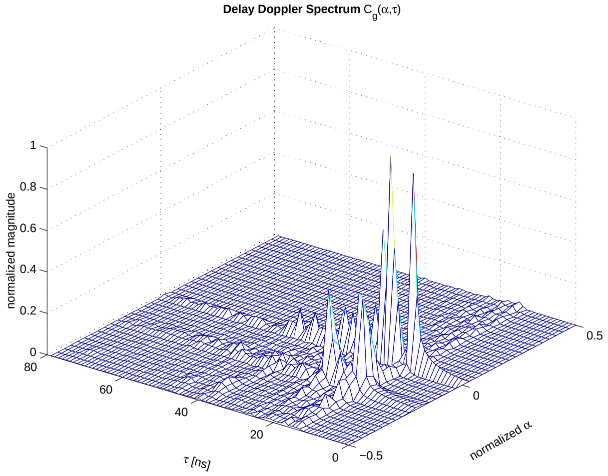

Fig. 2. Delay-Doppler spectrum

Obviously, a physical radio channel is practically limited in delay and Doppler spread. The product $S = 2\tau_{\max}\alpha_{\max}$ of maximum delay and Doppler spread, called spreading factor, covers the nonzero support area of the scattering function. If the complete channel statistics including, e.g., Doppler spread and Doppler spectra is to be reproduced by a sequence of estimated $g_i(nt_0,\tau)$, the value of S imposes constraints for choosing the parameters of the channel identification procedure in some two-dimensional Nyquist sampling sense. $t_0 < 1/(2\alpha_{\max})$ is required for real-time acquisition with respect to the time variability. On the other hand, the minimum measurement time for one channel impulse response is given by the length of the impulse response, $t_0 > \tau_{\max}$. This is necessary in order to sample all details of the frequency response $G_i(t,f)$. The channel has to be assumed time-invariant at least within the time interval $\tau_{\max}$. It follows that $S < 1$ is required for the channel to be identifiable. Fortunately, for a typical mobile radio channel (e.g., GSM), S is on the order of 1%. Thus, some time is left that can be used in order to reduce the measurement expense.

The simple WSSUS model imposes constraints that will obviously be too stringent if micro cell or indoor scenarios are considered. Then non-stationarities due to changes of mean path delays, amplitudes and directions as well as correlated scattering have to be taken into account. It seems feasible, however, to assume at least local stationarity over some limited time interval $T \gg 1/(2\alpha_{\max})$. This is the only chance to get reasonable estimates of the scattering function by some standard spectral estimation procedure such as WOSA (weighted overlapped spectral averaging), that is based on sliding (with t) windows.

B. Existing Methods

Having taken a look at physically motivated requirements for a complete measurement of the mobile radio channel, we are now ready to analyze the suitability and limitations of various known measurement procedures. There exist two classes. The first class ignores the channel's time variation. This allows a considerable reduction of measurement expense by using some slow sequential procedure in the time, frequency, or spatial domain. Such procedures are: (i) stepped frequency response methods that use commercial vector network analyzers, (ii) sequential correlator methods that are based on a slow shifting of the reference impulse train, (iii) synthetic aperture principles that are based on sequential spatial sampling of an antenna array grid [8], and (iv) angular sampling methods based on slowly rotating high resolution antennas [17]. Sequential methods in the frequency/delay and array/angle domains could of course be combined, but that would require even more measurement time and, hence, would cause additional problems with reference phase stability. Moreover, the usage of off-the-shelf equipment for method (i) calls for a wired connection between analyzer, transmit and receive antennas, which practically limits application of this method to short distances.

The second class uses real-time processing so that the time variation of the channel is recordable. But parallel processing in all the three domains (frequency, delay, and spatial) would require excessive hardware expense since the typical value of the spreading factor S, as discussed above, offers indeed the possibility to use some sequential operation without violating the Nyquist sampling criterion w.r.t. the channel's time variation. There are again some examples of channel sounders that use fast sequential operations, preferably in the delay domain. But most of the existing (commercial as well as experimental) real-time channel sounders are limited to one or two receiving antennas.

First measurement results using the new channel sounder will be produced in August '97. Meanwhile, despite its limitations a commercial vector network analyzer has been used in order to obtain some experimental data in the interesting frequency range for model and algorithm verification. The measurement results shown in Figs. 1 and 2 were obtained using omnidirectional disk-cone antennas at a center frequency of 5.5 GHz and a bandwidth of 1 GHz in a typical laboratory environment. With that equipment such a large bandwidth can easily be achieved but only the time average of the impulse response can be measured. However, a reflector movement has been simulated by a sequence of measurements whereby a strongly reflecting object was equidistantly moved step by step along a straight line. We found at least three strong multipath components, two of which arrived via the moved object. All of these paths are identified as reflections from items with a metal surface. It is interesting to note that the dominant paths are better distinguished from the delay-Doppler spectrum (Fig. 2). This is due to the fact that two paths with small time delay difference are subjected to different Doppler shifts.

C. Broadband Channel Sounder for Vector Radio Channel Identification

Within the project ATM*mobil*, a new channel sounder is being developed which is able to measure up to 8 receive antenna signals quasi simultaneously by fast array signal multiplexing. We call this a *vector channel sounder*, because it can be thought of as collecting one time snapshot of all array output signals into one signal vector. For this channel sounder we use a uniform linear antenna array that consists of 8 patch antennas for a frequency band of 5.1...5.3 GHz. The element spacing is chosen as 0.4 times the wavelength, so that spatial aliasing is avoided. The basic principle of broadband channel sounding that we apply is described in [5], [6]. The channel excitation signal is a periodic pseudorandom multisine signal with minimized crest factor and a bandwidth of 120 MHz. The individual antenna array channels are sampled block by block as indicated in Fig. 3. This multiplexing solution takes advantage of the influence of the spreading factor discussed above. Since we use FFT based real-time correlation, no time is wasted for other sequential operations so that the real-time channel sampling conditions are met. That offers the possibility to analyze WSSUS time variation (Doppler dispersion) and correlation between different array outputs as well as non-WSSUS parameters such as multipath correlation with respect to time delay and direction angle and the statistics of the non-stationary behavior introduced by the multipath scenario's variability.

RUSK ATM features several modes of operation because the sizes and access times of the employed memory media require a compromise between the length of the measured impulse responses, the measurement bandwidth, the number of vector channels, the block frame rate, and the total number of consecutive frames. While Fig. 3 shows standard Doppler mode with a measurement rate of 1 kHz, a fast Doppler mode with up to 20 kHz measurement rate can be realized for 8 channels, 3.2 μs impulse response length, and 256 consecutive blocks per channel.

In order to align the staggered blocks of $g_i(t, \tau)$ on the time axis, a lowpass interpolation is required. This is permissible since Nyquist sampling with respect to the Doppler

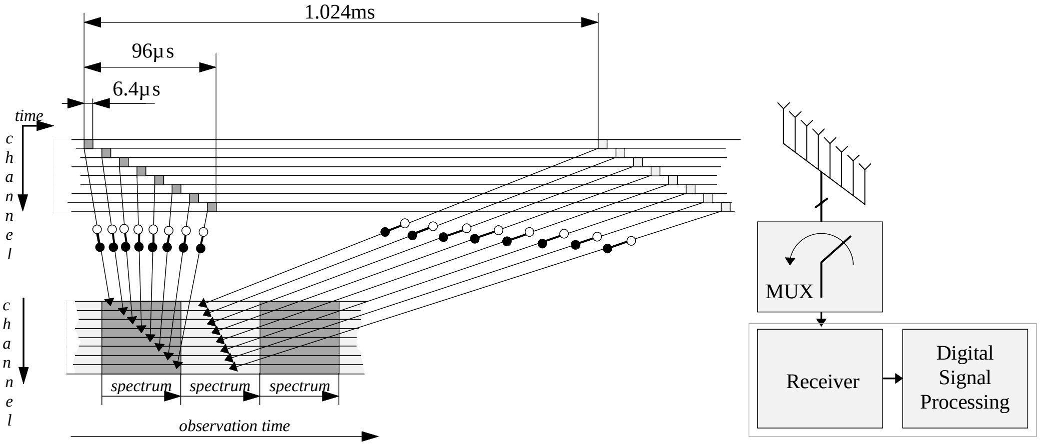

Fig. 3. Principle of sequential sampling of the vector output of an antenna array

bandwidth is assumed. Time averaging can additionally be used in order to enhance the channel parameter estimates if the time for multiplexing is smaller than $1/(2\alpha_{\mathtt{max}})$.

Compared to other methods, the proposed principle of sequential real-time vector sampling has some advantages:

- Hardware expense is only moderately higher when compared to synthetic aperture or rotating antenna principles since only a single receiver channel is required.
- Fast antenna multiplexing allows real-time estimation of the stochastic time variant channel impulse response. Stationary time variation as well as non-WSSUS characteristics can be measured.
- Fast antenna multiplexing is less sensitive to receiver phase instability and inaccuracies of sampling grid positions as compared to slow synthetic aperture principles. Calibration effort is reduced to the antenna array and to the multiplexer.
- Small array dimensions can be used since superresolution techniques can be applied in both the time and angular domain in order to resolve a sufficient number of multipaths. This is of interest especially for indoor applications.

III. FROM CHANNEL MEASUREMENTS TO CHANNEL SIMULATIONS

This section describes aspects for the derivation of a statistical simulation model for the directional radio channel. It turns out that inclusion of the non-stationary channel behavior is desirable but difficult. Model parameter extraction from real-time vector channel sounding data is discussed.

A. Empirical and Statistical Input/Output Vector Channel Modeling

The aim of link level simulations is to analyze the performance of transceiver algorithms. As a result, we expect to obtain some knowledge about the bit error statistics (instantaneous and average BER, error pattern) which in turn is necessary to evaluate the system capacity, the performance of coding schemes, etc. Clearly, for these tasks we need a radio channel model which is able to continuously transform an input signal into an output signal, as opposed to channel models that are intended to provide knowledge about some global channel parameters. On the one hand, this model must not be oversimplified (e.g., AWGN), and on the other hand, it has to be tractable in terms of computational complexity.

The frequently employed WSSUS model complies with these requirements. In simulations, it is usually realized as a tapped delay line with randomly time varying tap weights. If we use a receiver with multiple antenna reception, we have to account for the directional anisotropy of the radio channel due to spatially distributed transmitters and scatterers. An extension of WSSUS that includes the directions of arrival (DOA's) of the different multipath components was proposed by Martin [7] and is called Directional Gaussian Scattering (DGS). It assumes narrowband conditions in the frequency domain and in the angular domain. The first allows to convert time delays within the array into phase differences, the second allows to assign a discrete DOA to each macroscopic scattering area in the radio channel.

Both WSSUS and DGS models are by definition stationary in a statistical sense. For the DGS model this means, for instance, that there exist a fixed number of paths, that the DOA's and mean powers of these paths are nearly constant, and that the delays vary only within a part of the reciprocal of the transmission bandwidth. It is easy to recognize that there exist a multitude of real world situations which are not well described by these assumptions. For illustration only, assume that a relatively slim object is passed at a small distance. This results in a brief shadowing of all paths from one direction. Or imagine moving around a corner of a building. Some paths will disappear, some new paths will appear, and most of the remaining paths will change their DOA. These are among the most critical situations for receivers which employ adaptive signal processing; hence, they should be taken into consideration when assessing the performance of different algorithms. Especially in case of a directional channel, it is an unsolved problem how to include the numerous effects of a non-stationary environment into a statistical simulation model. With the term statistical we associate the idea that the model is representative of a number of situations. But this would require a much finer distinction of scenarios than is used, e.g., to classify GSM channels (Typical Urban, Hilly Terrain, ...). Furthermore, we would have to investigate the statistics of non-stationarities such as the appearance and disappearance of scatterers as well as change of DOA's and path delays.

As an easily applicable alternative, we can include in the simulation measured impulse responses recorded in typical scenarios ("stored channel"). This is in the spirit of modern simulation approaches which extend the simulation to include, step by step, more of the developed hardware components. In this way, if the measurements are carried out with the antenna or even the RF part of the system under development, the properties of these system components can be incorporated into the simulation. It should be noted that it is possible to implement a continuous signal transmission in the simulation even though the measurement works block by block. This can be accomplished by appropriate interpolation procedures provided that the measurement satisfies the sampling condition w.r.t. the channel's time variation. Table I summarizes some of the pros and cons of DGS model based vs. measurement based simulation of directional radio channels.

For statistical investigations as well as the use of the stored channel model, it is of essential importance to record the time variation of the directional radio channel completely. To our knowledge a measurement system for this task is not available yet. Up to now, time-variant channel modeling is based either on a model for the physical wave propagation phenomena [16] or on a combination of the results from scalar-valued time-variant and vector-valued time-invariant sounding experiments [8].

B. Estimation of Model Parameters

For the configuration of a directional channel model, it is necessary to estimate the appropriate parameters from measurement data. If the time-variant delays, angles, and tap weights can be treated as approximately constant within the observation interval, it is possible to use eigenstructure based superresolution algorithms for parameter extraction. Such algorithms are ESPRIT (Estimation of Signal Parameters via Rotational Invariance Techniques) or MUSIC (Multiple SIgnal Classification) [12] and their derivatives like Unitary ESPRIT [13] or root-MUSIC. For example, Martin used an ESPRIT technique for path delay estimation from mobile radio channel measurement data [6].

If, besides delays, DOA's of paths are to be estimated from vector channel sounding data, it is reasonable to use the same superresolution algorithms also for direction finding. For accurate results the array covariance matrix must be estimated from measurement data as well as possible. The three alternatives for averaging are shown in Table II.

The number of vector channels is usually limited by expense considerations. We decided for an 8 channel sounder. Consequently, the averaging in the space domain is hardly possible. As we are using a sequential block sampling method over time, averaging in the time domain is limited, or rather limits the measurable time variation. Therefore, only smoothing in the frequency domain is really possible.

It should be pointed out that the number of antenna elements limits only partially the absolute number of resolvable paths, because two paths with a delay difference greater than the reciprocal of the signal bandwidth B_s are already resolved in the delay domain. To determine the number of spatially separable paths, we must consider the following. As we are not estimating DOA's of different signal sources but of complex weighted versions of one signal (specified by a deterministic snapshot of the actual channel state) we have to deal with a coherent case. DOA estimation of coherent signal sources requires a special preprocessing called spatial smoothing. The so-called forward-backward (FB) spatial smoothing technique uses the interference pattern over the antenna array to solve a rank defect problem w.r.t. the source covariance matrix [14], [15]. In practical situations, using FB smoothing the number of resolvable directions is 2/3 times the number of antenna elements. Therefore, the number of resolvable paths with delay differences less then $1/B_s$ (=8.4ns) is roughly 5.

As stated above, path delays and DOA's can be estimated separately with eigenstructure based 1-D high resolution techniques such as ESPRIT. Here, the disadvantage is a loss of resolution and accuracy since almost all paths have distinct directions *and* delays. A better approach is joint detection of delays and DOA's. For example, Martin proposed a joint delay-DOA ESPRIT [8] based on Zoltovski's 2-D Unitary ESPRIT [9]. With this algorithm, it is possible to estimate the directions of arrival and delays simultaneously and there is no need to search for pairs of DOA and delay.

IV. SIMULATION OF DIGITAL RADIO TRANSMISSION USING ADAPTIVE ANTENNA ARRAYS

This section discusses implementation issues for receiver simulations using RUSK ATM measurement data. Interpolation procedures, bandwidth accommodation, and appropriate sampling are of importance. Finally, examples of the behavior of adaptive beamforming algorithms are shown.

A. Preprocessing

For the investigation of particular receiver components, e.g., an adaptive beamformer, it is desirable to abstract from possibly imperfect other components. This is most easily achieved when we restrict the simulation to an equivalent discrete-time model working at the symbol rate. If measurement based channel data are to be included, this might be an

TABLE I
ASPECTS OF RADIO CHANNEL SIMULATION

	Model based	Measurement based
Requirements for the scenario to be modeled	Stationarity in time and direction of arrival	One individual measurement for each scenario necessary
Exchange of antennas	Antenna characteristics can be included	One measurement is only valid for a particular antenna
Numerical complexity	Moderate, increases with model complexity	Slightly lower than model based, fixed
Storage capacity	Low	Very high

TABLE II
ALTERNATIVE METHODS FOR ESTIMATING COVARIANCE MATRICES

	Time domain	Frequency domain	Space domain
Averaging over	multiple channel impulse responses	distinct frequency bins	multiple subarrays
Averaging limited by	channel spreading factor	measurement bandwidth	number of elements

oversimplification because sampling at the symbol rate introduces aliasing into the measured impulse responses. This is due to the fact that, typically, the transmission bandwidth will exceed the symbol rate by a certain amount (excess bandwidth). The use of a Nyquist sampled channel model has two main consequences for receiver simulation. (i) It is necessary to perform some kind of timing recovery in order to adjust the symbol rate sampling phase. (ii) It is possible to perform some of the receiver processing with oversampling w.r.t. the symbol rate (e.g., fractionally spaced equalization), thus avoiding some of the drawbacks or limitations of symbol rate equalizers.

The following tasks are necessary for a simulation with RUSK ATM measured impulse response data that are stored as complex frequency responses:

The impulse responses to the different antenna elements are not measured at the same time. This time offset has to be compensated in a first step. In section II, we have shown that the time variation of the radio channel is fully captured if the sampling rate is at least twice the maximum Doppler frequency. Since the time offset to be compensated is relatively small compared to the sampling interval, a low complexity interpolation (e.g., linear) is sufficient.

The measurement bandwidth of RUSK ATM is 120 MHz, with a frequency bin spacing of 156.25 kHz. The complexity of a simulation for a transmission system with lower bandwidth can be reduced by bandlimiting and subsequent downsampling of the impulse responses. The bandlimiting lowpass filter can be (i) the pulse shaping filter of the transmitter, (ii) the transmit filter frequency-shifted to the baseband, or (iii) any lowpass filter with flat frequency response in the transmit band and just enough attenuation to prevent excessive aliasing due to the downsampling process. Downsampling is easily accomplished by decimation if the simulation sampling rate is chosen as a 2^n fraction of the measurement sampling rate.

A further reduction of complexity can be achieved if the analysis of the data shows that the effective duration of the impulse responses is much less than the measurement time (=6.4 μs). In this case the frequency sampling interval can be increased, most easily again by decimation (now in the frequency domain) if the impulse response length is shortened to a 2^n fraction.

All these operations can be completed before the simulation. The starting point is given by raw measurement data on a tape and the result consists in preprocessed impulse responses for a particular transmission system. These are stored in a special file format readable by the simulation program.

During the simulation, a further processing of the impulse responses takes place. The simulation of a continuous transmission system requires a much finer resolution of the impulse responses w.r.t. observation time than it is given by the measurement rate. For instance, for a measurement interval of 1.024 ms and an impulse response length of 6.4 μs an interpolation factor of at least 160 is required. To limit complexity, we propose a two stage procedure consisting of an interpolation using an FIR lowpass filter and a subsequent linear interpolation. The linear interpolation is known to produce negligible interpolation errors when operating on oversampled waveforms.

As long as there are no RUSK ATM measurement data available, we employ a statistical channel model for simulation. This model could be used for synthesizing channel impulse response data; here, the same processing steps as described above could be applied. An equivalent approach, however, integrates this model into the simulation system. This is what we have done for the following examples.

B. Examples

In order to evaluate the performance of array receivers in a non-stationary environment, we are developing simulation models with the freely available software environ-

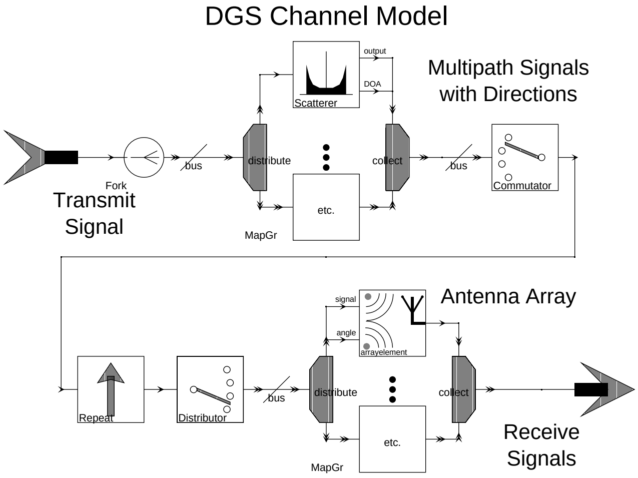

Fig. 4. PTOLEMY implementation of DGS channel model

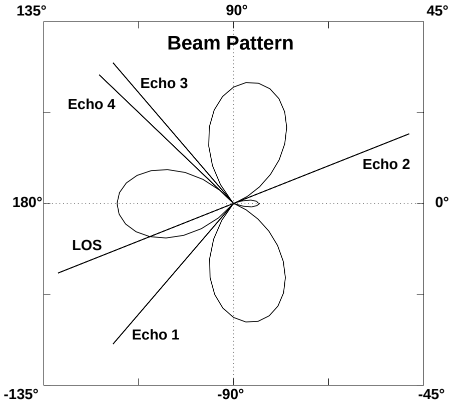

Fig. 5. On-line representation of array beam pattern

ment PTOLEMY [10] from UC Berkeley. This environment offers different models of computation along with a signal processing and communications systems library that includes elements for interactive simulation and animated output graphics. We designed a number of new building blocks for radio channel simulation and array processing.

In the following, some of the details of a symbol rate, single link simulation are presented. This simulation is suitable for comparing adaptation algorithms combined with different modulation schemes. A slightly extended version of DGS is used as channel model for this example. The extensions allow changes of DOA's and the appearance and disappearance of individual paths. Thus, the dynamic or transient behavior of signal processing algorithms in a not only time varying, but truly non-stationary environment can be observed.

Fig. 4 shows a PTOLEMY implementation of the extended DGS channel model that is part of a transmission system. The application of higher order functions (with the MapGr elements) offers a compact system representation. We use these higher order functions for a flexible mapping of signal paths to array output signals. The transmit signal is split up into the multipath components that are individually delayed, randomly weighted, and assigned to DOA's in the scatterer elements. The scatterers can be parameterized separately (mean power, Doppler PSD, discrete reflector, ...). The chain of commutator/repeater/distributor distributes each path to all array elements. The array elements calculate a weighted sum of all paths depending on their position within the array. Fig. 5 shows one of the periodically updated output blocks of the adaptive array simulation. The actual angular distribution of multipaths and the array beam pattern are depicted. A four element uniform circular array (UCA) has been used for this example.

We compare the performance of the well-known recursive least squares (RLS) algorithm and the least mean square (LMS) algorithm [11] in the case of one disappearing and one appearing echo path. Figs. 6, 7, and 8 show the progression of the mean squared adaptation error of the array output signal w.r.t. the transmit signal, obtained by averaging over 100 independent trials. There are 3 echo paths within the time interval marked by "Disappearing Scatterer" and "Appearing Scatterer" and 4 echo paths otherwise. We assume time varying Rayleigh or Rice distributed echo weights with classical Jakes type Doppler spectra. The delays of these paths are larger than the reciprocal of the transmission bandwidth. If the transmit sequence is white, the multipath components are similar to an uncorrelated interference. The array tries to suppress these signals, but in Fig. 6 the 4 echoes cannot be fully suppressed with 4 antennas. At the time instant marked by "Disappearing Scatterer," the array is able to suppress the remaining 3 paths considerably better. Approximately 100 time steps are required for this adaptation. When the new scatterer appears ("Appearing Scatterer"), the array has to redistribute its ability to suppress the interferers and finds the optimum within 20 time steps. In case of 5 antennas (Fig. 7), the array can suppress all interferers at any time but it needs a short time to adapt to the new situation when a scatterer appears. In Figs. 6 and 7 the RLS algorithm has been used for adapting the complex weights. This algorithm offers faster convergence and lower residual error as compared to the LMS algorithm which is used in Fig. 8 for the case of 5 antennas. The LMS algorithm is not able to handle the situation with 4 antennas. The advantage of the LMS algorithm is its substantially lower numerical complexity.

Some of the conclusions from our simulations can be summarized as follows. Directional nulls are more important for good performance of adaptive phased arrays than directed high gain beams. The suppression of multipath echoes with larger delays causes a reduction of the effective channel's delay spread. Accordingly, the necessity for an equalizer can be sidestepped. But a new tracking problem arises (tracking here means the decision directed adaptation phase) when we encounter the disappearance of the dominant propagation path. An equalizer can use the energy of the other paths to form its output signal and adapt the coefficients to the new situation whereas the array processor has suppressed all other paths and cannot find a new path with-

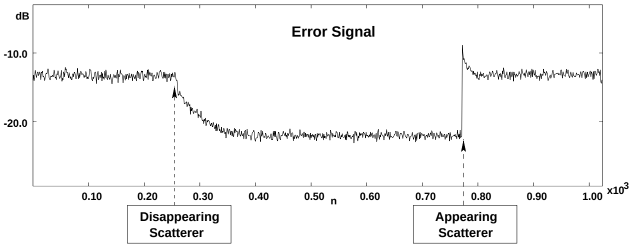

Fig. 6. Error signal of RLS array processor with 4 antennas

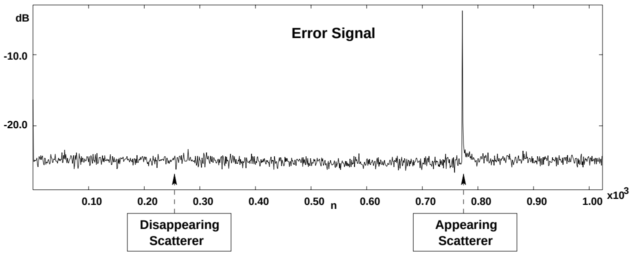

Fig. 7. Error signal of RLS array processor with 5 antennas

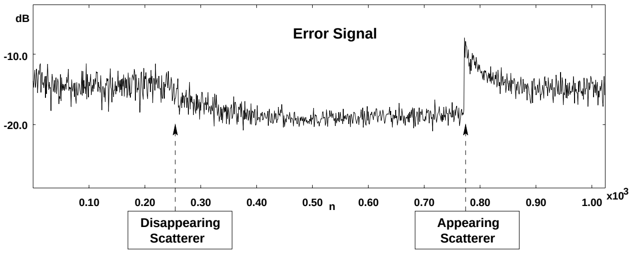

Fig. 8. Error signal of LMS array processor with 5 antennas

out a new training sequence. A solution would be the use of a combination of equalizer and array processor as proposed in [11].

V. SUMMARY

The comparison of receiver concepts with array front-end requires an appropriate consideration of the directional anisotropy of the radio channel. We have discussed possible approaches to implementing realistic simulation models. The main problem with statistical models is a suitable inclusion of the non-stationary nature of real world situations which in turn is important to evaluate the dynamic behavior of signal processing algorithms. An ad hoc solution is the use of an empirical model based on measured channel impulse responses. This requires real-time vector channel measurements which will be done with the RUSK ATM system. These measurement data will also be used to develop non-stationary statistical channel models. The application of high resolution techniques for estimating delays and DOA's is required. An example simulation of a transmission system illustrates the way an adaptive phased array works.

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REFERENCES

[1] E. Ayanoglu, K. Y. Eng, M. J. Karol, "Wireless ATM: Limits, Challenges, and Proposals," *IEEE Personal Communications*, Vol.3, No.4, pp.18-35, Aug. 1996.

[2] B. D. Woerner, J. H. Reed, T. S. Rappaport, "Simulation Issues for Future Wireless Modems," *IEEE Communications Magazine*, vol. 32, no.7, pp. 42-53, Jul. 1994.

[3] J. D. Parsons, *The Mobile Radio Propagation Channel*. Pentech Press, London, 1992.

[4] B. H. Fleury, P. E. Leuthold, "Radiowave Propagation in Mobile Communications: An Overview of European Research," *IEEE Communications Magazine*, vol. 34, no. 2, pp. 70-81, Feb. 1996.

[5] K. Schwarz, U. Martin, H. W. Schüssler, "Devices for Propagation Measurement in Mobile Radio Channels," *The Fourth International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC93)*, Yokohama, Japan, pp. 387-391, September 8-11, 1993.

[6] U. Martin, *Ausbreitung in Mobilfunkkanälen: Beiträge zum Entwurf von Meßgeräten und zur Echoschätzung*. Ausgewählte Arbeiten über Nachrichtensysteme, Erlangen 1994.

[7] U. Martin, "Statistical Mobile Radio Channel Simulator for Multiple-Antenna Reception," *IEICE 1996 International Symposium on Antennas and Propagation*, Chiba, pp. 217-220, Sep. 1996.

[8] U. Martin, "Charakterisierung und Simulation des richtungsabhängigen Funkkanals," *ITG-Fokus-Projekt "Mobile Kommunikation"*, Zürich, 11. 10. 1996.

[9] M. D. Zoltowski, M. Haardt, C. P. Matthews, "Closed Form 2D Angle Estimation with Rectangular Arrays in Element Space or Beamspace via Unitary ESPRIT," *IEEE Trans. Signal Processing*, vol.44, pp. 316-327, Feb. 1996.

[10] The Ptolemy Project Home Page, http://ptolemy.eecs.berkeley.edu

[11] J. Fuhl, F. Molisch, "Space Domain Equalisation for Second and Third Generation Mobile Radio Systems," *ITG Conf. Proc. "Mobile Kommunikation"*, Neu-Ulm, pp. 85-92, Sep. 1995

[12] C. W. Therrien, *Discrete Random Signals and Statistal Signal Processing*. Prentice Hall, Englewood Cliffs, 1992.

[13] M. Haard, J. A. Nossek, "Unitary ESPRIT: How to Obtain Increased Estimation Accuracy with a Reduced Computional Burden," *IEEE Trans. Signal Processing*, vol. 43, pp. 1232-1242, May 1995.

[14] J. E. Evans, J. R. Johnson, D. F. Sun, "High Resolution Angular Spectrum Estimation Techniques for Terrain Scattering Analysis and Angle of Arrival Estimation," *Proc. First ASSP Workshop Spectral Estimation*, Communication Research Lab., McMaster University, Aug. 1981.

[15] S. Pillai, B. Kwon, "Forward-backward Spatial Smoothing Techniques for the Coherent Signal Identification," *IEEE Trans. Acoust., Speech, Signal Processing*, vol. 37, pp. 8-15, Jan. 1989.

[16] G.Raleigh, S. N. Diggavi, A. F. Naguib, A. Paulraj, "Characterization of Fast Fading Vector Channels for Multi-Antenna Communication Systems", *IEEE Asilomar Conf. on Signals, Computers and Systems*, pp. 853-857, 1994.

[17] A. Klein, W. Mohr, "A Statistical Wideband Mobile Radio Channel Model Including the Directions-Of-Arrival", *IEEE ISSSTA '96*, Mainz, pp. 102-106, 1996.