

**System Model of VAMOS for GSM Receiver**Anitha¹, Dr. Nagamani K², Sahul Hameed Badhusha³¹Student, RV college of Engineering, Assistant Professor, RV college of Engineering Bangalore²Senior Project manager, Aricent Technologies

Abstract— GSM network is seeing its greatest expansion because of the growing demand for mobile voice services in emerging markets recently while maintaining backward compatibility with the legacy system. A newly proposed technology, Voice services over Adaptive Multiuser channels on One Slot (VAMOS) feature has the potential to double voice capacity of the existing GSM mobile communication system while maintaining backward compatibility with the legacy system. This would help operators in densely populated cities to alleviate the strain on their networks. Concept of MUROS is based on multiplexing two or more users onto one time slot without degrading the speech quality. It is based on Orthogonal Sub-Channel (OSC) approach. Lot of research work is dedicated to signal generation and especially to new receiver concepts as well as power control in order to implement VAMOS transmissions. The aim of our work is to implement a model system able to evaluate performance parameters related to downlink transmission in VAMOS. The model implements generation of the VAMOS modulated signal with power control and receiving solutions based on a modified MLSD (Maximum Likelihood sequence Detector) structure that allows comparison of performances in VAMOS downlink transmission.

Keywords—VAMOS, GSM, MLSD, subchannel power imbalance ratio, OSC.

I. INTRODUCTION

VAMOS is specified as an extension to the Global System for Mobile communications (GSM), one of the most widely used cellular communications system in the world. It is claimed by the increased demand for voice services, especially in emerging markets and the introduction of new wireless services such as machine to machine (M2M) communication. As specified by 3GPP Technical Report Release 8, the solution introduces an adaptive modulation scheme, new orthogonal training sequences and a VAMOS sub-channel power control feature which is fully backward compatible, i.e. it can be introduced without any impact on existing end user devices [1].

The technology enhancements introduced in 3GPP specifications is to increase the voice capacity handled by each Base Station (BS) by multiplexing two users simultaneously on the same physical resource in the circuit switched mode both in downlink and in uplink, using the same timeslot, the same frequency (Absolute Radio Frequency Carrier Number - ARFCN) and the same TDMA frame number while maintaining backward compatibility with the legacy system.

A new set of training sequences that allow distinguishing between the two VAMOS users representing a VAMOS pair is defined. The new set of training sequences (TSCs) has been found based on cross correlation properties between existing and new training sequences. The orthogonality property of the new TSCs is also required.

On each side (base station and mobile station) the receiver uses its assigned training sequence for channel estimation and, de-correlation processes, thus eliminating the data sequences from the paired VAMOS user allocated to the same resource. As cross-correlation properties of the training sequences are not ideal, this leads to additional interference experienced by the user device.

In uplink there is no need for a special solution as the GMSK modulation scheme is maintained. Different MSs transmit their own signals simultaneously and the BS receiver has the capability to distinguish and demodulate two simultaneous GMSK signals by applying a multi-user detection algorithm taken advantage by the fact that it has two or more antennas [2].

In downlink, each mobile receives the signal based on its single antenna. The concept proposed by 3GPP is to transmit each signal in different OSCs. The two signals are multiplexed together and transmitted by BS. The MS receives the signals and remove the interfering signals using advanced equalizing techniques.

VAMOS can also take advantage of the widely available Downlink Advanced Receiver Performance (DARP) Phase 1 handsets which can perform Single Antenna Cancellation (SAIC) [3]. Eight new Training Sequence Codes TSCs, which have low cross-correlation with the existing TSCs have been added to the system.

In addition the VAMOS concept implement the power control mechanism which means that the power given to each user is based on their path-loss to provide the lowest required signal level for reliable reception by each user. Starting from the path-loss of each paired mobile the base station set up the required power level P1 for user 1 and P2 for user 2. The ratio

P1/P2 is called Sub-Channel Imbalance Ratio (SCPIR). The SCPIR is also equal to the ratio of the path-gain values of the two users. The VAMOS sub-channel power control feature is symmetric, i.e. if one of the paired users has a SCPIR of 6dB, the other will have a SCPIR of -6dB. Thereby VAMOS mode of operation distributes suitable level of RF powers. This mechanism is important since it allows legacy mobiles (both DARP and non DARP) to operate in VAMOS mode. The feature also enables to increase the number of viable VAMOS pairs (number of variants of pairing) and to enlarge VAMOS service coverage reducing the number of handovers.

Substantial research is dedicated to MUROS candidates techniques regarding the performance objectives: increasing voice capacity of GERAN by means of multiplexing at least two users simultaneously on the same radio resource both in downlink and in uplink, investigation of different optimized pulse shapes for MUROS and balancing between low frequency reuse factor and high time reuse [4], [5], [6].

Finally the MUROS study leads to standardization of VAMOS mode of operation which characteristics have been presented earlier [7], [8]. The aim of our work is to implement a model system able to study technical aspects and to evaluate performance parameters related to downlink transmission in VAMOS.

The model system implements:

- generation of the VAMOS modulated signal;
- a receiver solution based on a modified DFE structure as proposed in [9] that allows comparison of performances in case of 2 reception filter solutions. The model system is implemented using C programming. Based on this model we study the performance parameters associated to VAMOS downlink transmission: BER, raw BER, and BLER functions of SNR and SCPIR. We compare the performance of the implemented receiver for synchronous and asynchronous interferers relative to the useful signal.

The outline of the paper is as follows: in Section II generation of the downlink VAMOS signal is presented. In Section III the receiver structure for VAMOS downlink transmission is discussed. Section IV presents simulation results regarding the performance evaluation of the implemented system model.

II. DOWNLINK SIGNAL GENERATION SCHEME

3GPP standards mention different types of MUROS/VAMOS modulation schemes for DL signals [1], [10]. The most applied are: - Linear sum of 2 GMSK signals degrees phase separation; - QPSK modulation with 900 linear Gaussian filter; and QPSK with Root Raised (RRC) filter.

In our work we linearly combine 2 GMSK baseband modulated signals and sent in OSCs. This is done by rotating one of the two GMSK signals by 90 degree to the other signal. Power control for the paired users is also included in the generation scheme indicated in Fig.1. With P1 and P2 expressed in linear scale, the amplitude ratio of the two users, $R = P1/P2$, is applied to control the gains G1, G2 for each user according to: $G = \cos \alpha$, $G = 2\sin \alpha$, $\alpha = \arctan R$; where α shall be chosen such that SCPIR ≤ 10 dB.

The scheme also allows a single GMSK signal transmission making SCPIR value very low. The resulting signal after summing is applied to a RF modulator and power amplifier for radio transmission.

In Fig. 1 $x_1[n]$ and $x_2[n]$ are generated based on {0,1} bit sequences derived by digital processing the voice signal. degrees and by adding it

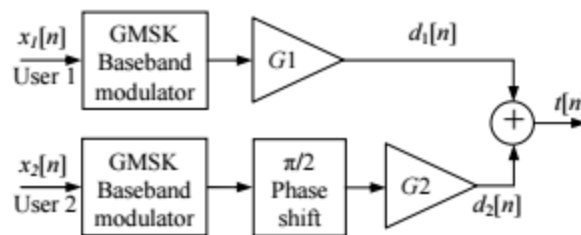


Fig. 1. VAMOS downlink signal generation

Transmission on the GSM channel is based on normal bursts, consisting of 148 bits. Of these: 114 bits are voice (useful) coded bits using a convolution code followed by an inter leaving operation; 26 bits represent the training sequence embedded in the middle of the burst; 3 bits are tail bits at each end of the burst and 2 bits are used for control. Furthermore these sequences are differentially encoded and level shifted to produce $x_1[n]$ and $x_2[n]$. GMSK is a Continuous Phase Modulation (CPM) technique, characterized by constant envelope and narrow bandwidth [10]. The

information is carried out by the transmitted signal phase which is a linear function. Controlled ISI is introduced to improve spectral efficiency.

III. RECEIVER SYSTEM MODEL FOR VAMOS DOWNLINK

The receiver scheme has many tasks: eliminates the propagation channel effects, removes the signal intended for the paired mobile station as a CCI, and removes the deliberately ISI introduced by GMSK modulation.

For VAMOS downlink transmission, different types of receivers intended to eliminate the interferer based on SAIC principle can be applied [13], [14], [15]. These can be categorized based on detection criterion used to extract the desired signal and to mitigate the interferers as follows:

- 1) Joint MLSE, which is considered the optimal receiver in noise limited scenarios. The JMLSE simultaneously extract $1 \times n$ and $2 \times n$ data sequences using the Viterbi algorithm, based on a specific branch metric of state transition that suppose the knowledge of the transmit channel. JMLSE receiver implementation is difficult because the computational complexity of the detection criterion is extremely high. The algorithm also requires the knowledge of the impulse response of the interferer which is difficult to estimate, as well as the training sequence of the paired user.
- 2) Mono Interference Cancellation (MIC). The received signal is first filtered with a complex-valued filter $(z)P$. The resulted signal is projected onto an arbitrary complex vector c and next fed into a trellis based equalizer that removes ISI. The undesired interference can be totally reduced with an appropriate choice of the filter $(z)P$.
- 3) MIC with Successive Interference Cancellation. As the performance of MIC degrades severely with the increase of interference level, the MIC algorithm performs better if the VAMOS interferer is not stronger than the desired signal. MIC with Successive Interference Cancellation (SIC) first detect the higher interfere signal using MIC and trellis based algorithm. This signal is subtracted from the received signal and the result is fed into another MIC receiver to reconstruct the desired data sequence.

We consider the discrete time model of VAMOS receiver which considers alpha-QPSK detection in the presence of dominant GMSK interferer. Although the phase orthogonality of the two paired users is guaranteed in transmitter side, the two sub channels will be leaked and interfered each other in front end of the receiver because of the inter symbol interference (ISI) resulting from multipath characteristic of radio channel and non-linearity of transmit and receive filters. Thus results in system performance degradation. The channel qualities of the two paired users cannot be guaranteed at an acceptable level because of sub channel interference and cannot be improved immediately by the means of the transmit power because of slow power control mechanism. Consequently, it may make the voice traffic drop and lower the quality of service. So an efficient sub channel interference cancellation is important to improve the channel condition of VAMOS system [4].

A.Channel estimation

Due to the multipath fading while transmission of particular signal there is some inter symbol interference and inter-sub channel interference in the received signal of VAMOS. Therefore, a signal detector needs to know the characteristics of channel impulse response to ensure successful equalization (removal of ISI) and to extract the original user information. In GSM system, usually channel impulse response is extracted based on the known training sequence, which is transmitted in each normal burst as shown in fig 2. Normal burst (NB) is used to carry information on traffic and control channels, except for RACH (Random Access Channel). It contains 116 bits and includes a guard time of 8.25 bits duration (30.46 μ s). The receiver can use the known training sequence bits and the corresponding received samples to estimate the channel impulse response typically for each normal burst separately

tail	Data bits	training	Data bits	tail	GP
3	58	26	58	3	8.25

Fig 2: Burst structure

In GERAN/VAMOS system, each user also suffers from the paired sub channel interference, co-channel interference from neighboring cell, adjacent channel interference and additive Gaussian white noise. Because the frequency reuse in GSM is loose, the co-channel interference is from one cell to its homologue cell (i.e. using the same set of RF channels, or cell allocation). Generally speaking, the VAMOS sub channel interference is the dominant interference source if there is no power control of the sub channel [7].

A VAMOS scenario is considered. After AQPSK modulation and pulse shaping, the user information is transmitted with its paired interfering signal over a multipath fading channel. In the receiver, thermal noise is generated and modeled by additive white Gaussian noise, which is sampled at the symbol rate. The demodulation problem here is to detect the transmitted bits from the received signal. Besides the received signal the detector also needs the estimated channel impulse response, which is provided by a specific channel estimator device using the known training sequence bits [7].

B. Channel estimator for single channel

Digital signal or required signal for the user is transmitted over a fading multipath channel, after which the signal has memory of L symbols [2]. At the receiver thermal noise is generated and which is modelled by additive white Gaussian noise n, it is sampled at the symbol rate. The demodulation problem here is to detect the transmitted bits a from the received signal y. Besides the received signal the detector needs also the channel estimates $\hat{h}$, which are provided by a specific channel estimator device. Noise corrupted system is shown in fig3.

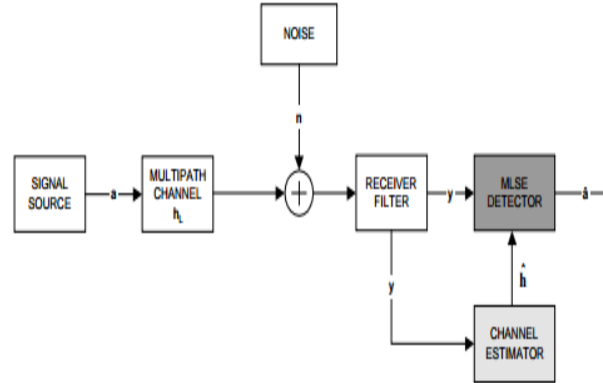


Fig3:Noise corrupted system

The received signal y can be expressed as follows

$$y = Mh + n \quad (1)$$

Where the complex channel impulse response h of the wanted signal is expressed as

$$h = [h_0, h_1, \dots, h_{L-1}] \quad (2)$$

and n denotes the noise samples. Within each transmission burst the transmitter sends a unique training sequence, which is divided into a reference length of P and guard period of L bits, and denoted by

$$m = [m_0, m_1, \dots, m_{P+L-1}] \quad (3)$$

Having bipolar elements $\in \{-1, +1\}$ Finally circulate training sequence matrix M is formed as in equation (4)

$$M = \begin{bmatrix} m_L & m_1 & m_0 \\ m_{L+1} & m_2 & m_1 \\ m_{L+P-1} & m_P & m_{P1} \end{bmatrix} \quad (4)$$

The Least Square channel estimates are found by minimising the following squared error quantity

$$\tilde{h} = \arg \min \|y - Mh\|^2 \quad (5)$$

C. JOINT CHANNEL ESTIMATOR FOR 2 SIGNALS

Presence of co-channel interference that is shown in fig 4. Two synchronised co-channel signals have independent complex channel impulse responses

$$[h_n, n = h_{0,1}, h_{0,2}, \dots, h_{0,L}] \quad n=1,2 \quad (6)$$

Where L is the length of the channel memory. The sum of the noise n and co-channel signals is sampled in the receiver. The joint demodulation process is to detect the transmitted bit streams a1 and a2 of the two users from the received signal y. To assist that joint detection operation the joint channel estimator provides channel estimates and .

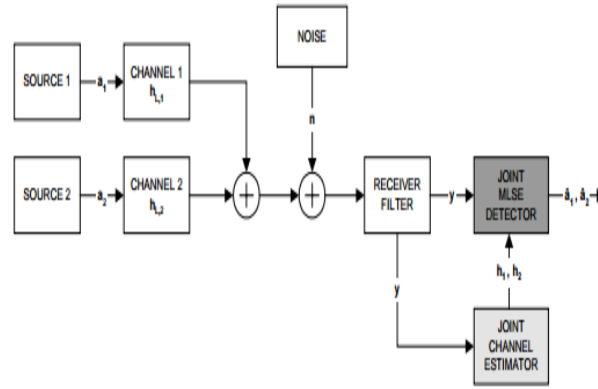


Figure 4:Block diagram of co channel interference

The complex channel impulse responses of the two synchronous co-channel signals are expressed with a vector containing the channel taps of the individual signals denoted in (7)

$$\tilde{h} = \begin{pmatrix} h_{L-1} \\ h_{L-2} \end{pmatrix} \quad (7)$$

Hence, $\tilde{h}$ has totally $2 \times (L + 1)$ elements. Both the transmitters send their unique training sequences with a reference length of P and guard period of L bits.

The circular training sequence matrices are denoted by

$$\tilde{h}(n) = \begin{bmatrix} h_{0,1} \\ h_{1,1} \\ h_{L,1} \end{bmatrix} \text{ where } n=1,2.. \quad (8)$$

$$m_n = \begin{bmatrix} m_{0,1} \\ m_{1,n} \\ m_{p+L-1,n} \end{bmatrix}, n=1,2.. \quad (9)$$

$$M_n = \begin{bmatrix} m_{L,n} & m_{1,n} & m_{0,1} \\ m_{L+1,n} & m_{2,n} & m_{1,n} \\ m_{L+p-1,n} & m_{p,n} & m_{p-1,n} \end{bmatrix}, n=1,2.. \quad (10)$$

and they are gathered into one large matrix

$$M = [M1 \ M2]$$

With these notations the received signal y is again given by

$$y = Mh + n \quad (11)$$

The Least Square channel estimates can be found simultaneously for the both users by minimising the squared error quantity, which produces in the presence of AWGN the following solution

$$\hat{h} = \text{argmin} ||y - \tilde{M}\tilde{h}||^2 = (\tilde{M}^H \tilde{M})^{-1} \tilde{M}^H y \quad (12)$$

D. Joint Maximum Likelihood Sequence Detector

After the estimation of channels is completed next step is to decode the sequence received, Method used is MLSD.

It selects the data sequences that minimize the squared Euclidean distance between the noisy observations and all possible hypotheses of noiseless received sequence.

$$a_0, a_1, a_2, \dots, a_J = ||y - (ha(-1))||^2 \quad (13)$$

IV. PERFORMANCE EVALUATION AND RESULTS

Simulations for performance evaluation are achieved based on the VAMOS downlink generation scheme, Fig. 1, and on the receiver structure described in Fig. 2 implemented in C programming. In the transmission part, the normal GSM burst is generated based on the input data bits stream according to GSM standards. The input data is a random sequence. The

sequence length corresponds to a voice frame according to GSM Full Rate speech encoder. Channel encoding corresponding to GSM normal burst of TCH FR (Traffic Channel Full Rate Speech) is implemented by splitting input data sequence in classes and applying convolution encoding to these classes according to GSM standard. In the interleaver stage the data blocks resulting from channel encoder are mixed over a number of bursts to spread a block error over more received data blocks. Ciphering of data provided in GSM standard is not implemented. Input data for GMSK modulator are obtained by multiplexing the data from the interleaver according to GSM normal burst structure. Both old and new training sequences defined by [1] are implemented and can be selected for generation of the GMSK signal for the desired and paired users. The normal burst differentially encoded and mapped into a NRZ representation is GMSK modulated according to the equations presented above. The GMSK modulator generates the oversampled in-phase I and quadrature phase component Q. The oversampling ratio, denoting the number of signal sample per data bit, can be selected. At the receiver the corresponding down sampling operation is completed after the receiver filter in Fig. 2. According to Fig. 1, two GMSK signals, generated as described above are weighted with an appropriate factor corresponding to the desired SCPIR parameter and summed in order to obtain the transmitted signal. Generating only one GMSK signal corresponding to a single GSM user is possible by setting SCPIR factor to an appropriate value. The paired user signal in Fig. 1 holds a training sequence that belongs to the new TSCs set and suffers a $\pi/2$ rotation by multiplying each symbol by j . It provides satisfying the orthogonality. For a correct interleaver operation the GSM signal generated according to Fig. 1 is organized in blocks. One block contains four GSM normal bursts of 148 bits each. In all our simulation an oversampling ratio of 4 is used and the transmitted number of blocks is 2,000, containing overall 8,000 GSM normal bursts. In the simulations the used radio channel for small-scale fading is supposed to be a Typical Urban Channel with 6 taps and the user speed is assumed to be 3 Km/h (TU3 channel) at central carrier frequency in GSM 900MHz band. Independent fading for each transmitted block is considered. Other types of channels can be implemented as well. The desired user signal and the paired user signal are passed through the same channel, but different channels for the two users can also be considered. At the radio channel output an Additive White Gaussian Noise (AWGN) was added to the radio signal. In the simulations we set appropriate SNR limits in order to highlight the VAMOS systems performances both in noise limited and interference limited scenarios. In our simulations we consider only a single interferer as a dominant interfering signal produced by the paired user. The interferer is considered in different scenarios being perfect synchronous as well as asynchronous with the desired signal. At the reception a synchronization operation with the desired signal is performed based on the autocorrelation property of the training sequence. At this step, an estimate of the radio channel is evaluated.

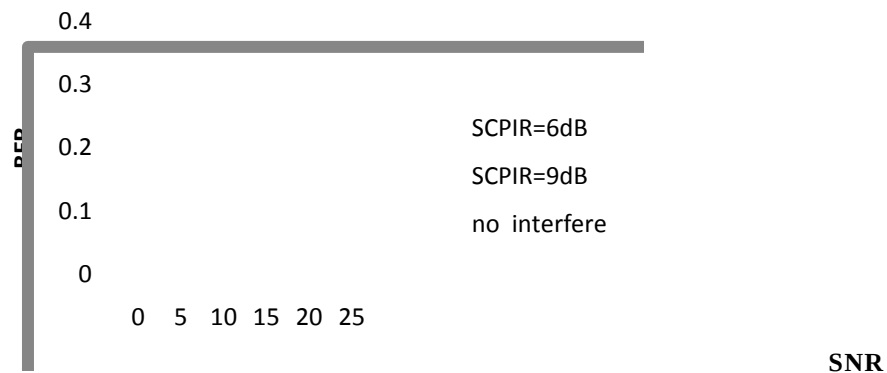


Fig 5: BER rate versus SNR for various SCPIR and no interferer

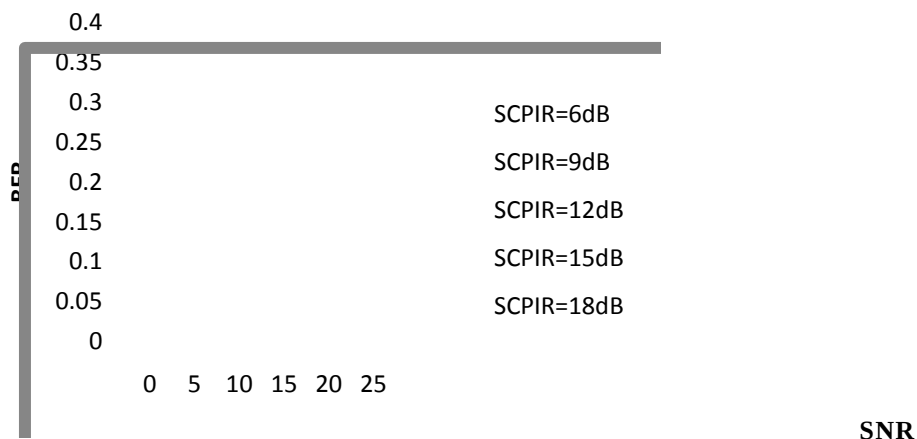


Fig 6:BER versus SNR for different SCPIR

Fig 5 shows that Bit Error Rate (BER) is low when interferer is not present. Whereas BER rate increases as sub channel power is reduced. For SCPIR of 9dB BER is less compared to SCPIR of 6dB.

Fig 6 provides graph of BER versus SNR for different SCPIR ranging from 6dB to 18dB. Which shows that if there is more power imbalance between the two sub channels then BER is less.

V. CONCLUSIONS

The aim of our work is to create a system model that represents a useful tool for studying VAMOS downlink transmission. We include in our model the significant aspects related to VAMOS concept: signal generation, reception, and interference cancellation. We achieve performance evaluation based on the VAMOS downlink generation scheme and on the receiver structure implemented in C programming. We present some performance parameters obtained using the proposed model for different scenarios.

The model can be extended by including other receiver structures and techniques to improve interference cancellation. The proposed model can be used to design a power control algorithm and to set the initial estimate of the optimal power levels for two VAMOS users. Furthermore based on the model a user pairing strategy can be developed by including also practical measurements gathered from a realistic network.

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