L*IP Satellite Gateway

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Executive Summary

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Contents

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2 Introduction

The L*IP Satellite System developed by Joanneum Research and Graz University of Technology is an communication product hand-tailored for the Internet Protocol and its real time and non-real time applications. Video, voice and data applications can be supported via this network.

Meshed systems like the L*IP avoid the drawbacks of star topology networks, i. e., it satisfies the requirements of real-time services with only a *single hop*. The available satellite bandwidth is used in an optimum fashion via QoS-aware DAMA control which balances the bandwidth for all the users in the uplink as well as in the downlink.

The L*IP system encapsulates IP packets based on a *time granularity* and not the common cell (ATM,MPEG) granularity which results in less overhead on average compared to an ATM style encapsulation. We see our hardware architecture as another big advantage over the approach of some competitors. By using a dual Pentium platform, we have a very powerful and future-proof platform, which is not the case for a lot of embedded processors. Thus, we have a *flexible hardware* architecture concerning access schemes and protocol requirements. Our software radio approach is based on PC and FPGA hardware, which is flexible, relatively cheap and also available in the coming years.

We have developed with our platform every module from the Ethernet Interface to the 70MHz Modem interface, which includes: IP handling, IP header compression, QoS support, DAMA, MF-TDMA access and adaptive transmission.

3 Architecture of the L*IP Satellite Gateway

In order to be prepared for future extensions and modifications, flexibility and reconfigurability have been essential aspects for the architectural design. Therefore, an *open platform* based on FPGA devices is realized, which allows the implementation and test of a great variety of modulation schemes, base-band shapes and synchronization algorithms [1]. All subsequent processing, even error control coding, is carried out in pure software. The hard real time tasks are executed on an FPGA –PCI card hosted in a dual Pentium IV (3.0 GHz) where also the tracker and codec is located. The higher layers are processed in a second PC with the same performance. With this architecture we are able to perform the processing of up to 4 Msymb/s with variable code rate (R = 1/3, 1/2, 2/3, 3/4, 4/5, 8/9), MF-TDMA access system, IP header compression, IP-optimised encapsulation, QoS-aware DAMA and a Java-based GUI for control and monitoring.

Figure 1 shows the integrated gateway with Burst Modem PC, which hosts also the FPGA card. Further the gateway PC, the AD and DA converter box, an off the shelf L-band converter and keyboard / Monitor for both modem and gateway PC, which can be accessed with a keyboard/monitor/ mouse switch.

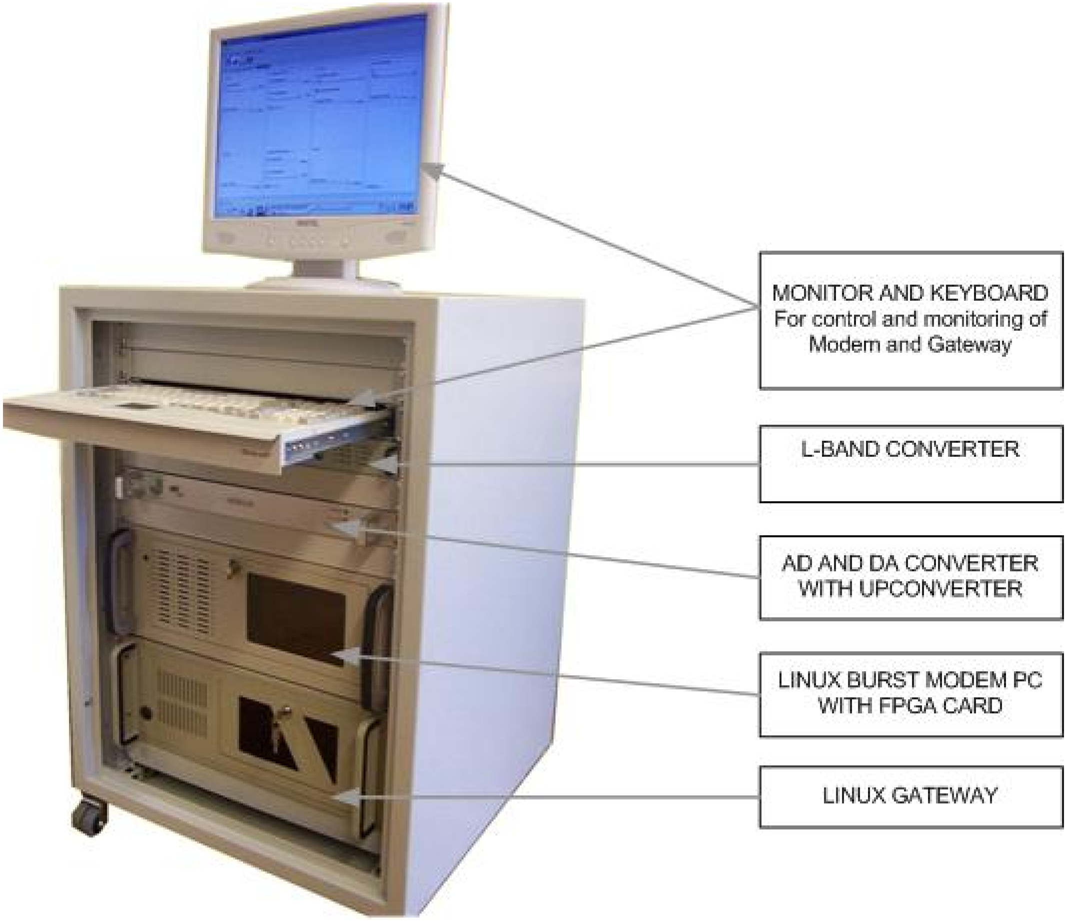


Figure 1: Integrated L*IP Satellite Gateway

4 The MF-TDMA Modem in Software-Radio Technology

4.1 Principle Architecture

An external box contains all EMC sensitive devices like amplifiers, filters, AD and DA converters. Complex tasks from the algorithmic point of view, like media access and channel coding, are performed in a PC, whereas all high-speed processing, like modulation, down-conversion and acquisition, is handled by a FPGA card. This result in a scalable architecture, where parameters like symbol rate, carrier frequency, coding rate and/or coding scheme can be changed easily. The main advantage of this approach is the tight interconnection between modem hardware and access system, but also the adaptability of interfaces during the design process. The result is a perfectly matched system, which has great potential for future extension. This has been exploited to include fade mitigation techniques as well as an algorithm for joint recovery of carrier and data [2].

Figure 2 illustrates the overall architecture of the MF-TDMA modem. All analogue parts, i. e., anti-alias filters, amplifiers, AD and DA converters, are placed in an external box to protect signal processing against EMC in transmitter and receiver. This box is connected via a LVDS bus to the FPGA card. The high-speed samples from AD conversion are passed to mixer and down-conversion block to obtain the over sampled base band signal. It has been decided to implement these functions in FPGA. In the sequel, the initial acquisition process provides a first estimate on the main transmission parameters, i. e., carrier frequency/phase and symbol timing [3], which delivers a stream of soft-decision samples with a maximum rate of 64 Mbit/s transmitted over the PCI bus without problems.

Much more flexibility can be achieved by using higher programming languages running on the Pentium machine under Linux, as done with the phase tracker and the FEC unit such that a maximum of flexibility and scalability is guaranteed.

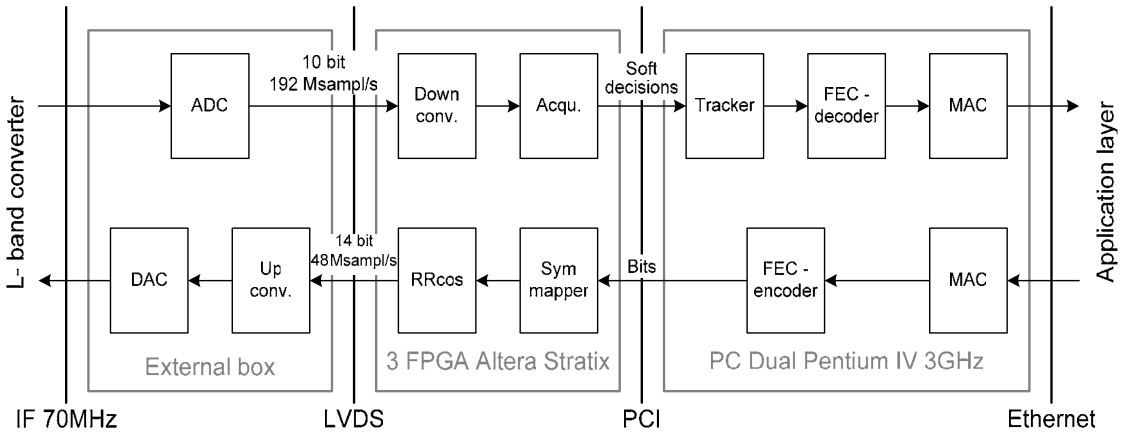


Figure 2: Block diagram of the MF-TDMA modem

The core of the demodulator is the powerful acquisition algorithm such that the demodulator can be operated down to a SNR = 5 dB. The main advantage is the close interaction between the media access control (MAC) and the modem. For instance, all information gathered in the MAC module can be employed in the modem and vice versa, e. g., frequency offsets of all stations measured at the master station in order to use them for monitoring purposes. Due to the efficient interfacing between physical and MAC layer, the guard-time between bursts could be reduced to 2.5 μs at 4 Msym/s although the roundtrip time varied in a range of about 300 μs. Another big step to improve availability of the system is fade adaptation, where the modem collects the SNR estimates of each burst such that the access system can adapt the transmission power of each station individually or assign the burst to channels with less symbol rate. Alternatively, a downlink fade can be compensated by changing the code rate.

State-of-the-art burst modems require a preamble up to 80 symbols for timing recovery as well as a unique (sync) word of 32 symbols and more for frequency and phase estimation. The developed MF-TDMA modem needs only a unique word of 40 symbols applied to a correlation algorithm for *joint* recovery of carrier and timing. Reference bursts, broadcasted regularly on a specified channel, are used for estimation and correction of the carrier frequency offset by a Luise-Reggiannini algorithm [4], [5]. This sort of bursts is also employed when a station comes into the system.

Parameter	Developed L*IP modem
Burst concept	Variable
Operational SNR	≥ 5 dB @ PER=10^{-6}, R=1/2
Preamble + sync word	0 + 40 symbols

Table 1: Main specification of the MF-TDMA modem

4.2 Performance

The key parameter of the modem performance is the bit error rate (BER) as a function of the SNR. For QPSK as modulation scheme, Figure 3 provides the ideal BER (no sync loss) as black diamonds and the measured one as black rectangles joined by solid lines. The two grey curves in between represent MATLAB simulations related to symbol timing recovery [3] with Lagrange interpolation (rectangles) and a scenario with the sample taken next to the optimum (triangles) which has been used in the FPGA implementation. The difference between the best-sample approach and the measured curve identifies the implementation loss, whereas the difference between ideal curve and best-sample strategy is due to the sampling error which amounts for L*IP to ±1/6 of the symbol period in the worst case. Hence, the overall implementation loss is approximately 1 dB if the modem is operated at an SNR = 8 dB. It can be observed that the loss increases for larger SNR’s since sampling errors imply less signal power and more noise through intersymbol interferences. Of course, the performance could be increased if interpolation were used, but it turned out that the effort is higher than the benefit, not at least, because the modem is designed to work at an SNR of 5 dB, where the improvement becomes negligible.

Another important parameter for MF-TDMA modems is the burst-loss rate. Mainly three effects contribute to this effect: missed burst, cycle-slips, and decoding failures. For instance, if the acquisition fails due to heavy noise, the related block is denoted as missed. In contrast, as soon as the phase tracker loses lock, the QPSK constellation will be rotated by multiples of $\pi/2$, rendering the rest of the burst virtually useless since not correctable by even very powerful FEC algorithms. Finally, a burst is also lost if the FEC decoder fails which is discussed in more detail in the next section.

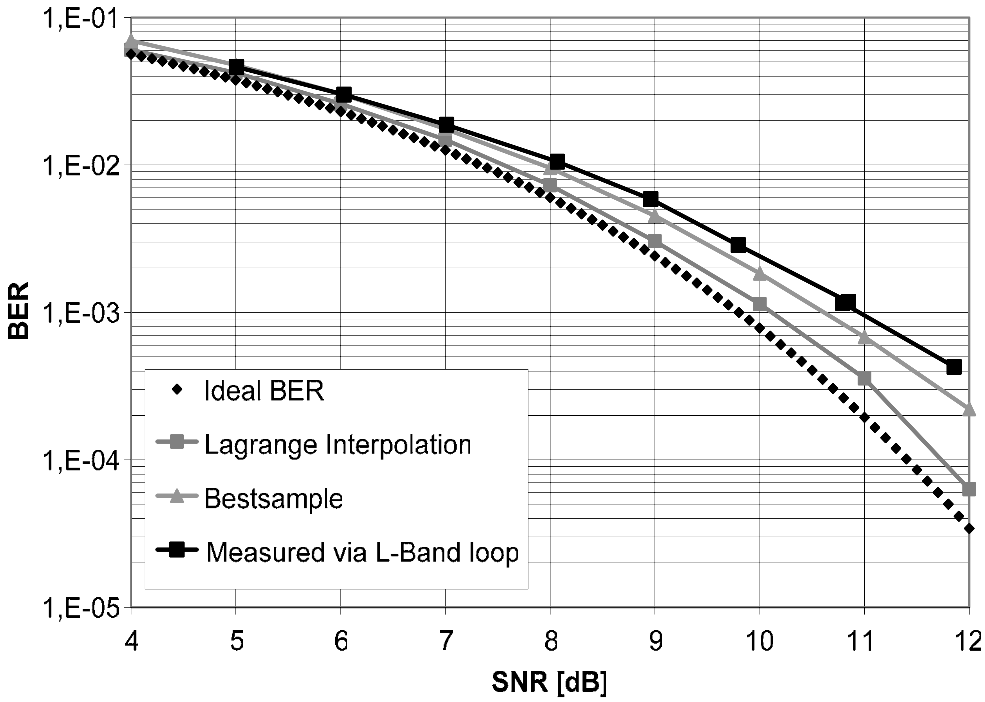


Figure 3: Overall BER performance of the QPSK MF-TDMA modem

5 Channel Coding with Turbo-Viterbi Codecs

5.1 Description of the Principle

Current PC's have sufficient memory and processing power such that forward error correction (FEC) algorithms may be implemented and run in real-time as high-level software. However, such prominent schemes like Turbo codes [6] require rather complex operations because the maximum-a-posteriori (MAP) principle, which the decoder is based on, includes exponential and logarithmic functions. In a pure software solution, the latter would consume definitely too much computing power to be useful in practice. But an appropriate simplification, the so-called max-log-MAP decoder, involves only additive and comparative operations such that the computational complexity is drastically reduced.

Unfortunately, the max-log-MAP approach exhibits a significant degradation of the error performance, i. e., mainly the *error floor*, related to the asymptotic behaviour, is raised considerably. The latter can be lowered most effectively by increasing encoder memory and/or block length. But with a low-complexity approach, the memory length is limited by the available processing power and the block length cannot be altered arbitrarily if the application or service behind is packet-oriented as it is more and more the case with current and future systems.

According to theory, the error floor may be decreased most easily by an increase of the minimum Hamming distance of low-weighted code words via appropriately designed interleavers. A second approach, frequently mentioned in the open literature, is the concatenation with outer block codes, which suffers from additional computational complexity and extra bandwidth. With L*IP, however, a solution has been developed (see Figure 4), where one of the max-log-MAP trellises is used for Viterbi decoding *after* the iterative procedure has been stopped by a simple criterion [7], [8].

Compared to a max-log-MAP decoder, a significant decrease of the error floor is observed with the implemented Turbo-Viterbi (TV) hybrid. A detailed analysis confirms that the benefit is due to the fact that the Viterbi algorithm follows a maximum-likelihood (ML) principle minimizing the sequence error probability, not the bit error probability as do MAP devices. The TV hybrid must not be confused with any form of concatenation and, as such, no extra bandwidth is required. Furthermore, no additional resources are needed insofar as both max-log-MAP and Viterbi algorithm use the same trellis.

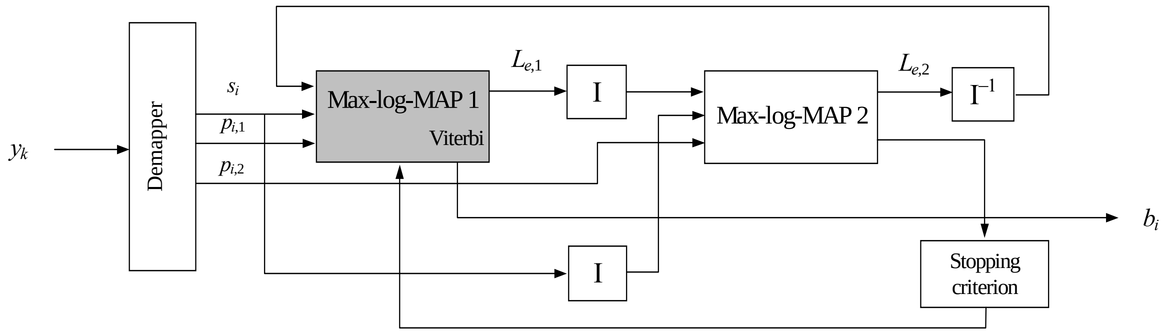


Figure 4: Hybrid Turbo-Viterbi decoder

5.2 Joint Carrier Synchronization and Turbo Decoding

For L*IP, *carrier synchronization* and *Turbo decoding* have been implemented as separate modules]. What has been implemented for L*IP-PLUS is the *joint* recovery of carrier and data. This is mainly motivated by the fact that the tracking loop, which is currently used for carrier recovery, suffers from cycle-slips and hang-ups with a detrimental effect on the error performance if the decoder has to be operated at lower SNR values as it is typical for iterative coding schemes.

With respect to L*IP, it is clear that the error performance of the Turbo decoder is limited by the performance of the preceding unit for carrier synchronization. The additional loss is mainly due to cycle-slips and hang-ups introduced by the recovery loop. Therefore, an advanced solution has been implemented for L*IP-PLUS: *joint* carrier synchronization and Turbo decoding achieved through *per-survivor processing* (PSP).

For joint recovery of carrier phase and user data, PSP has already been suggested in the open literature [12], i. e., the carrier frequency/phase is recovered via an array of trackers attached to all nodes in a trellis section. Each tracker is operated in a decision-directed (DD) manner, where decisions on QPSK symbols are obtained through the branch of the corresponding survivor path. In a simplified version, only the DD tracker related to the best path metric is implemented, causing solely a degradation of the error performance on the order of a few tenths of a dB. The DD device is shown in the next figure with the VV detector replaced by a DD module. If $\hat{c}_k$ denotes the decision value of the QPSK symbol, the DD algorithm provides simply the imaginary part of $\hat{c}_k^* r_k$ as a measure for the carrier phase offset, i. e., $u_k = \mathrm{Im}[\hat{c}_k^* r_k]$.

For QPSK as modulation scheme and rate 1/2 codes, the PSP principle has been successfully applied to Turbo codes [13]. In this context, it is to be noticed that the DD tracker must be fed with QPSK symbols, whereas the Turbo decoder as such remains bit-oriented. Therefore, not every QPSK symbol can be employed for carrier recovery because, according to [7], the parity bits $p_{i,1}$ and $p_{i,2}$ are related to MAP1 and MAP2, while the systematic bits s_i are shared by MAP1 as well as MAP2 separated by an interleaver π. For clarity, this is exemplified subsequently in more detail.

First, we assume that the QPSK sample r_k is attached to a trellis section with index i in MAP1, i. e., $r_k \sim (s_i, p_{i,1})$ is used for DD tracking in each section node (in the simplified version, only the node with best metric is taken). In the following trellis section, however, only the systematic part of the QPSK symbol corresponds to MAP1 since $r_{k+1} \sim (s_{i+1}, p_{i+1,2})$ when the code rate $R = 1/2$. Thus, the tracker state related to index i must be applied to recover the phase of r_{k+1}. Note that the latter includes the parity information for MAP2. The procedure continues periodically up to the end of the data block. The extracted soft decisions, i. e., s_i and $p_{i,2}$, are processed together with the extrinsic information $L_{e,1}$ in MAP2 without PSP as suggested in [13]. Finally, as soon as $L_{e,2}$ is available at the output of MAP2, the next iteration starts again with MAP1.

Of course, the PSP approach described previously can be extended to coding rates other than 1/2. The periodic arrangement of systematic and parity bits composing a QPSK symbol is of particular interest in this case. With $R = 2/3$, for example, the sequence of r_k's at the input of MAP1 may be periodically aligned as $(s_i, p_{i,1})$, (s_{i+1}, s_{i+2}), $(p_{i+2,2}, s_{i+3})$. Note that $r_{k+2} \sim (p_{i+2,2}, s_{i+3})$ cannot be used in MAP1, whereas $r_{k+1} \sim (s_{i+1}, s_{i+2})$ can, although the trellis must be traced back for two branches. Similar relationships are easily found for coding rates other than 1/2 or 2/3 (preferably 3/4, 4/5, and 8/9).

Since the joint module is to be run in pure software, the relationship between error performance and computational complexity is of vital interest to cope with 1/2/4 Msymb/s. Therefore, the following points have to be taken into account for the detailed software design:

Adaptation for *integer arithmetic's*

Evaluation of tracker functions (phase shift) via *look-up tables*

Application of the *vector instruction set* (available on Pentium processors) to compute the max-log-MAP recursions in parallel

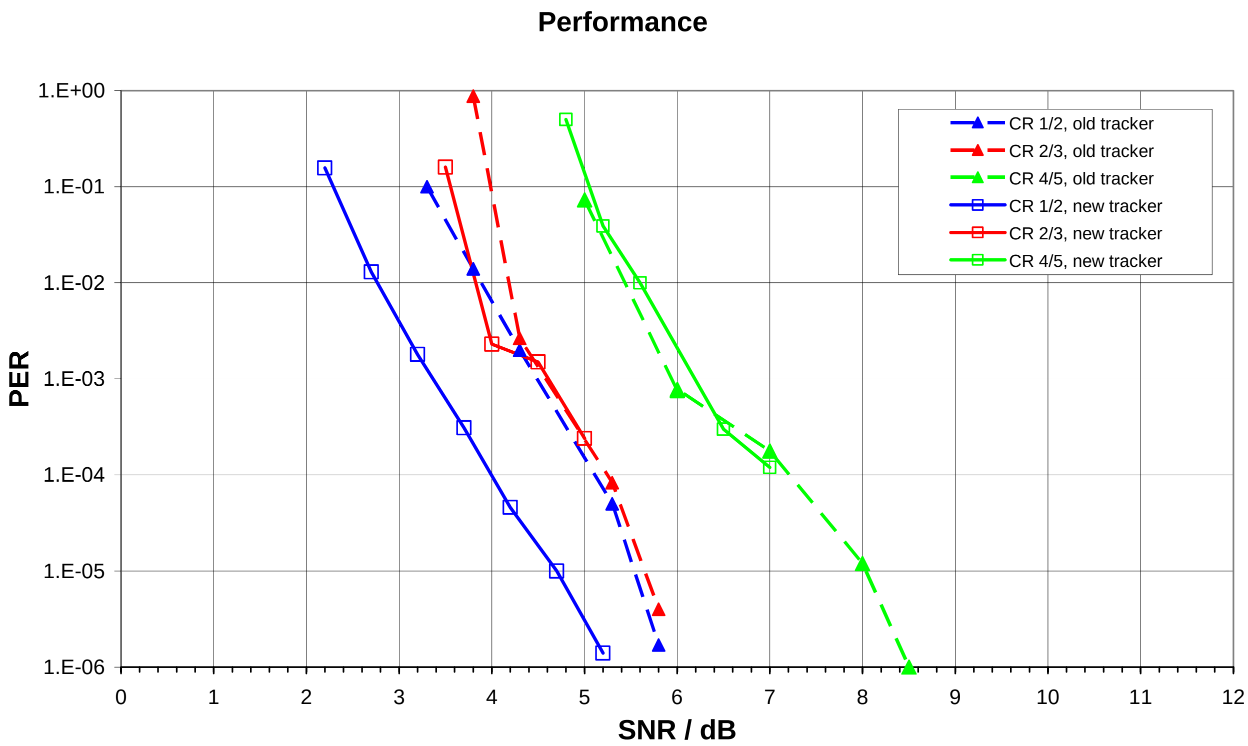
Appropriate *simplifications* of the joint algorithm (only one DD tracker per trellis section, phase recovery only for MAP1)

5.3 Overall L*IP Error Performance

The requirements to achieve a gain of 0,5dB compared to the original implementation are verified in this test.

- Test of the Phase Tracking & Turbo software. The codec operates at code rates 1/2, 2/3 and 4/5.
- The packet error rate PER as a function of SNR can be measured. To reduce the measurement time an internal packet generator tool is used to fill up the entire allocation plan.
- The measurement is done for the old and the new tracker with the joint synchronisation approach.

Curves having the same colour indicate identical code rates. Solid lines represents the new tracker, the dotted lines the old tracker.



It can be shown that at code rate 1/2, the overall performance of the modem is enhancing at about 1 dB. At 2/3 improvement are small, and just at PER rates of 1e-3 and higher significant. At a code rate of 4/5 there is no improvement noticeable. This is the expected result, since there is not much implementation loss of the modem at 2/3 and 4/5.

It was possible to decrease the operational point of the modem from 6dB to about 5 dB. In fact **stable operation of the modem is performed at 4.5 dB**!

6 Fade Mitigation

The master station has in the L*IP access system the central control over the system resources. Bandwidth allocation in form of burst assignment is completely performed by this central control instance. Information when (time slots), where (RF-channels) and how (channel-modes) bursts may be sent and must be received is provided in detail in the allocation plan AP, which is sent periodically to the slaves.

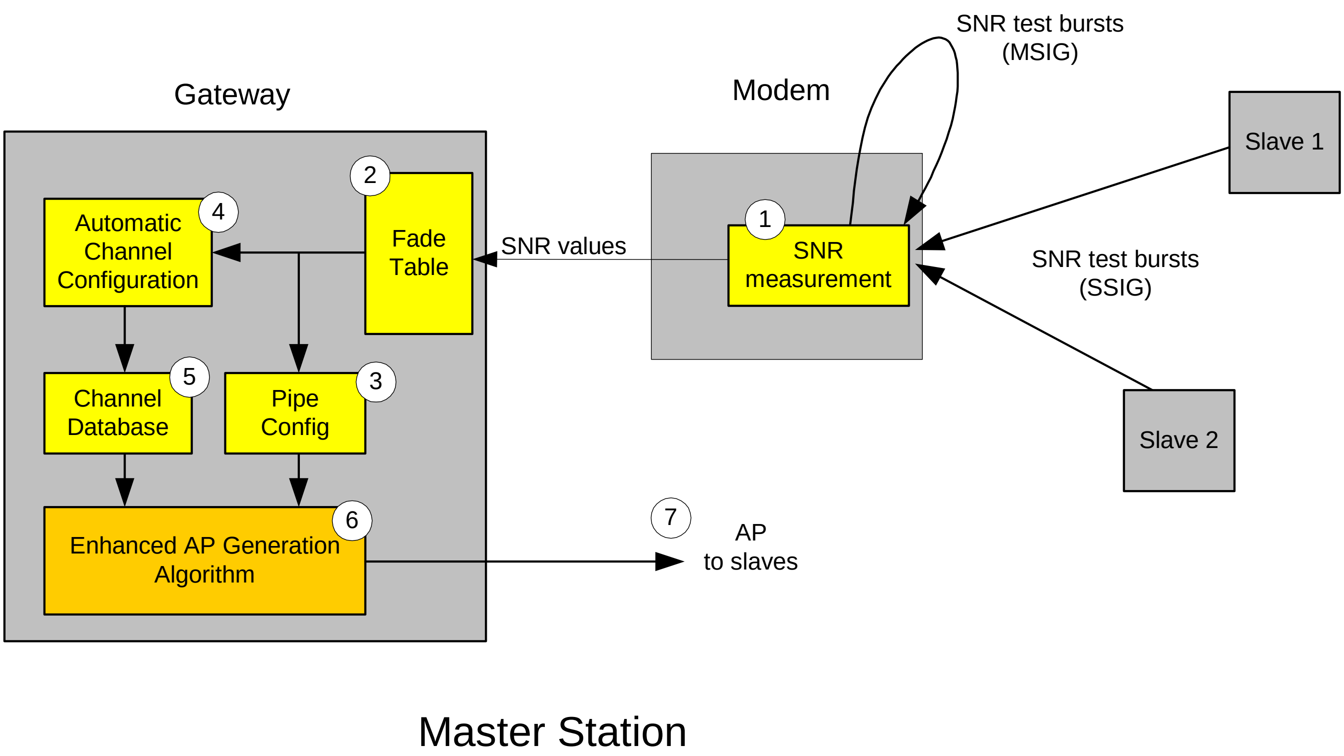


Figure 5: Architectural overview of the fade mitigation algorithm

Introduction of fade mitigation requires a number of changes and enhancement throughout the access system. Besides new modules, which are to be implemented, there are several enhancements of the existing concepts necessary. In detail following issues must be taken into account:

- SNR measurements of each slave station
- Extension of the configuration database (pipes, channels) to store and interpret the measured values
- Algorithm for automatic reconfiguration of the channel set-up
- Enhanced Burst Scheduler, which takes fade events into account
- Synchronisation between signalling messages and TDMA frame numbers to provide set-up changes without service interruption.

The architectural design overview of the fade mitigation system:

(1) SNR measurement is done in the modem, based on the received symbols (soft decisions). Measurement is performed on SSIG and MSIG bursts.

(2) SNR values are stored in the fade table.

(3) Based on the information about fade events in the complete network, the pipe configuration is updated. The up- and downlink attenuation between each pair of stations is taken into account. A look-up table is generated to grant fast access to this information.

(4) Based on the fading status and the estimated traffic load of the stations, a channel reconfiguration module decides when RF-channels have to be split up and rejoined again in order to support an optimised utilisation of the network resources.

(5) In the channel database the different transmission variants (channel-modes) of the RF-channels are stored.

(6) The AP generation algorithm employs the pipe configuration for an update of the burst plan, which guarantees that all stations are able to receive their bursts correctly and independent from the attenuation on the corresponding link.

(7) The AP, which is broadcasted to all slaves, contains now all necessary information to transmit and receive bursts with respect to the measured fading events.

With the existence of fade events, part of the user traffic is limited to RF channels with lower bandwidth, which leads to additional restrictions in the burst distribution process. To avoid a negative impact on the overall bandwidth utilisation, reconfiguration of the RF channel set-up is necessary, e.g., larger channels are divided into smaller ones. On the other hand, it should be possible to supply stations not affected by fade events with larger portions of the bandwidth. In fact, an optimisation algorithm is performed, which maintains constantly the channel set-up (split and rejoin of RF channels) with respect to traffic volumes affected by fade events or not. All in all, it is possible to perform fade mitigation in the L*IP access system without waste of bandwidth and without impact on channel utilisation.

For fade adaptation two different mechanisms are applied in the enhanced L*IP access system:

- Increasing of the transmission power, if attenuation is in the uplink path. This can be done by reduction of the symbol rate.
- Increasing of the code rate, if attenuation is in the downlink path.

If fade events occur up- and downlink both mechanism are applied. Furthermore the transmit power in the slave station P_{ES} relative to the maximum possible transmit power $P_{ES}^{(max)}$ and the power flux density at the satellite Φ_{SAT} relative to the maximum allowed power flux density $\Phi_{Sat}^{(max)}$.

7 DAMA

In many wireless communication systems like LMDS or VSAT systems the available bandwidth is shared among several user stations. In such systems one strategy to use bandwidth efficiently is to assign bandwidth to these stations that presently require bandwidth- Demand Assignment Multiple Access (DAMA).

The access method used in our system is based on a centralized control by the master station that controls the whole satellite network. The access to the satellite data slots is reservation based. This means that bandwidth for a connection has to be explicitly assigned by the master before the slave can transmit the data.

In principle two levels of assignment are currently implemented in the system. Bandwidth for a connection can be **pre-assigned**, or it can be **assigned on request**. Pre-assigned bandwidth is suitable for types of traffic with constant or nearly constant rates, like telephone calls, but most IP-traffic is highly unpredictable. Pre-assignment of bandwidth means simply waste of bandwidth in this case. So assignment on request is the method for IP-traffic.

The Slave Station generates periodically bandwidth requests based on the amount of incoming traffic (volume based bandwidth request) and sends them to the Master Station via the slave signalling (SSIG) connection. The Global Access Agent collects the bandwidth requests from the stations and builds an Allocation Plan depending on the QoS parameters and the bandwidth requests. The Allocation Plan is broadcast periodically to each Slave Station via the master signalling (MSIG) connection. The Cell Multiplexer in the Slave Station assigns at least the reserved bandwidth to the virtual connections.

The method of bandwidth assignment affects in general throughput and delay of satellite communication systems. The basic problem is the fact that bandwidth has to be requested at the master station and so the earliest reaction on a request arrives 2 satellite round trip times later.

Three different approaches are used in the system:

- **Fixed Bandwidth Allocation FBA**: In the FBA scheme a fixed amount of slots is given to a station for the duration of its connections. With FBA delay guarantees can be delivered, if the bandwidth is assigned for the peak rate of the traffic. This method is used for the guaranteed assigned QoS class.
- **Pure-DAMA**: In this scheme bandwidth is assigned based on bandwidth requests from the stations. The amount of assigned bandwidth is limited to the present traffic load of the station. The packet delay in Pure-DAMA is at least 3 round trip times (one for the request, one for the transport of the allocation plan and one for the packet itself). This method is used for the bulk transfer QoS class.
- **Controlled Over-Assignment DAMA CO-DAMA**. In the CO-DAMA scheme more bandwidth is granted to the station than requested, but on a predictive manner. This method is used for the guaranteed available and high and low priority best effort QoS class.

8 IP optimized Encapsulation

A new encapsulation scheme was developed to transport IP packets with low overhead. We called it Low Overhead Encapsulation (LOE) [10]. The segmentation at the transmitter is carried out in the following way (see figure 6):

1. Select a non-empty queue with respect to QoS aspects and queuing strategies. For instance weighted round robin could be used, but in principle every multiplexing method is applicable.

2. Format a Sub-network data unit (SNDU) or, if this has already been done, use the remaining part of the previous assembled SNDU.

3. Build a fragment such that it fits into the burst allocated from the master station. The selected size of the fragment should be as big as possible to maximize efficiency. This means that either the burst is filled up completely or the SNDU is transmitted totally.

4. Repeat this process until the burst is completely filled up or all queues are empty.

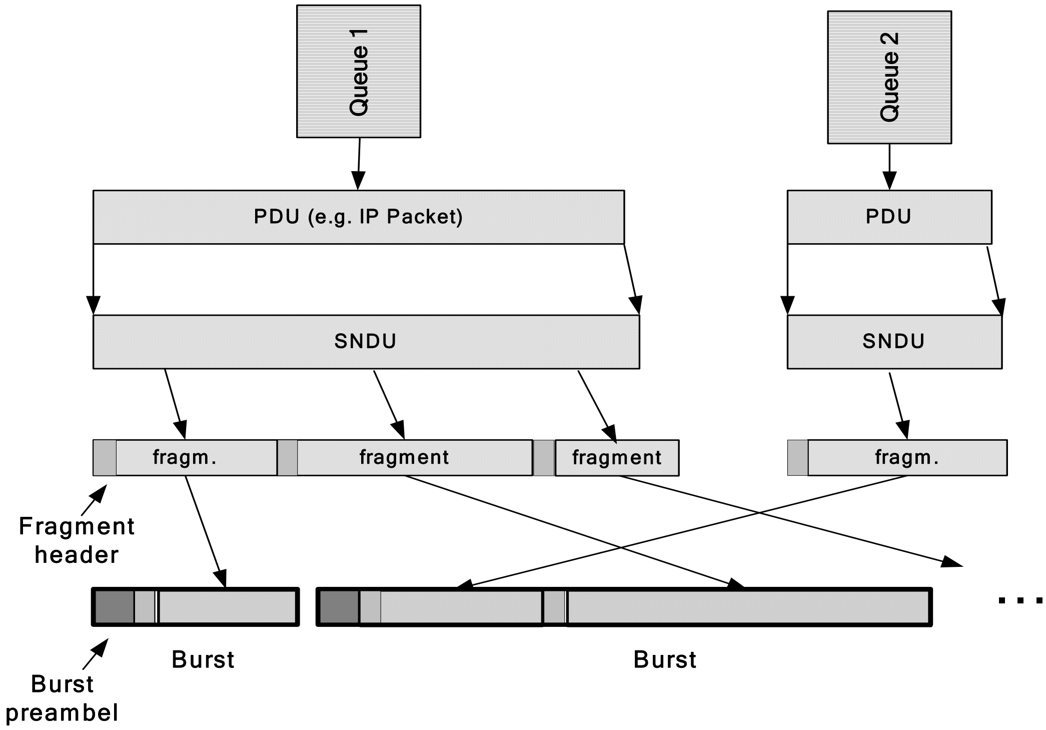


Figure 6: Example for LOE Encapsulation

With the implemented encapsulation process can the number of fragments and accompanying number of headers be significantly lower than with an ordinary cell-based approach like the DVB-RCS with ATM or MPEG cells.

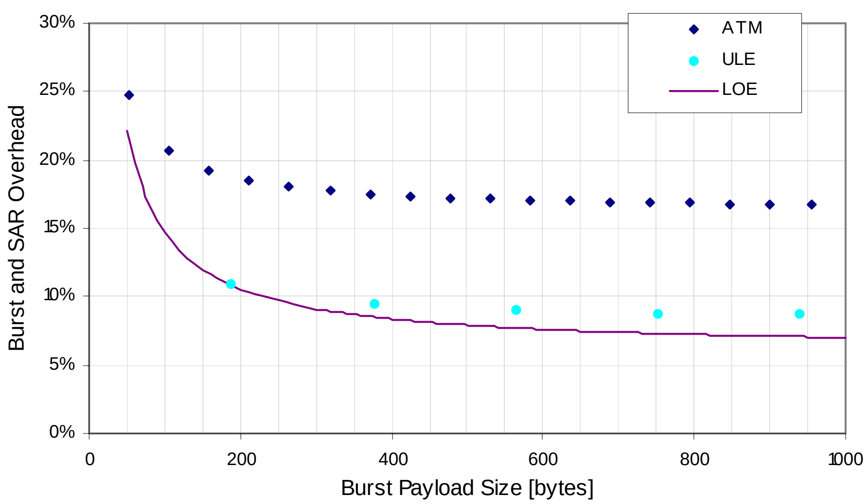


Figure 7: Overhead of LOE compared to ATM and ULE (IP over MPEG)

We were able to verify that about 10-15% of bandwidth could be saved with LOE in comparison to ATM and still up to 2% in comparison with Unidirectional Light Encapsulation ULE for MPEG streams [11] , assuming that each MPEG-2 TS unit could be filled up completely (Figure 7).

9 Graphical user interface for control and monitoring

The Monitoring, Control and Configuration software of the L*IP network is based on a client-server architecture implemented in Java language. This is necessary in order to provide a remote access to the system. The software exists of two main parts:

- the GUI (client) at the Central Monitoring and Control station (central host)
- agent programs (servers) at the Central Station (master)

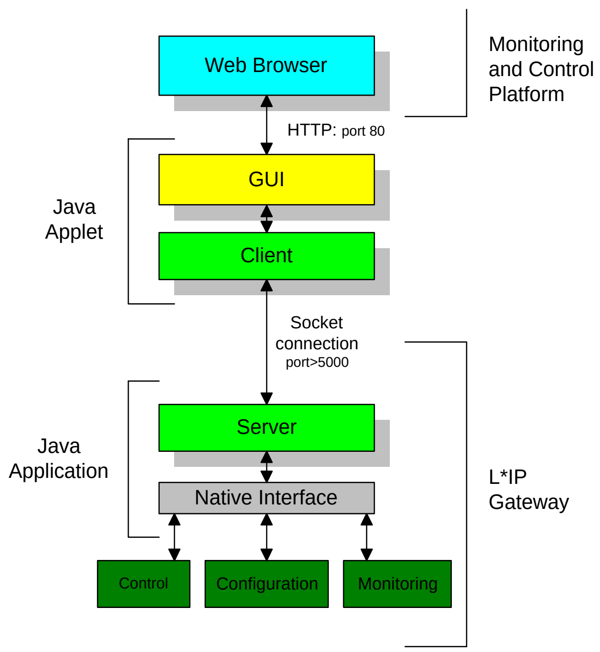


Figure 8: Overview of Monitoring and Control Software Architecture

9.1 Selected Screenshots from the User Interface

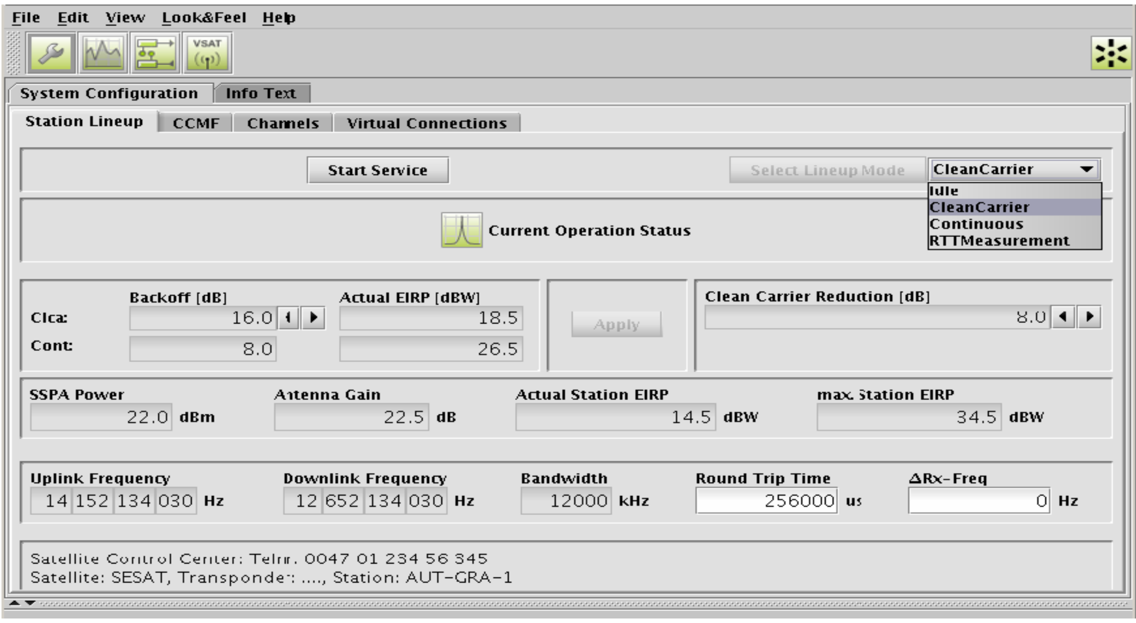


Figure 9: Station Line-up GUI

The Uplink and Downlink centre frequency values are derived from the Space Segment Resource (SSR) settings displayed in the 'Channels' panel. The Uplink Frequency value is the sum of the transponder uplink frequency in MHz and the frequency offset in Hz requested from the used SSR. The Downlink Frequency value is the uplink frequency minus a frequency offset defined through the outdoor unit equipment.

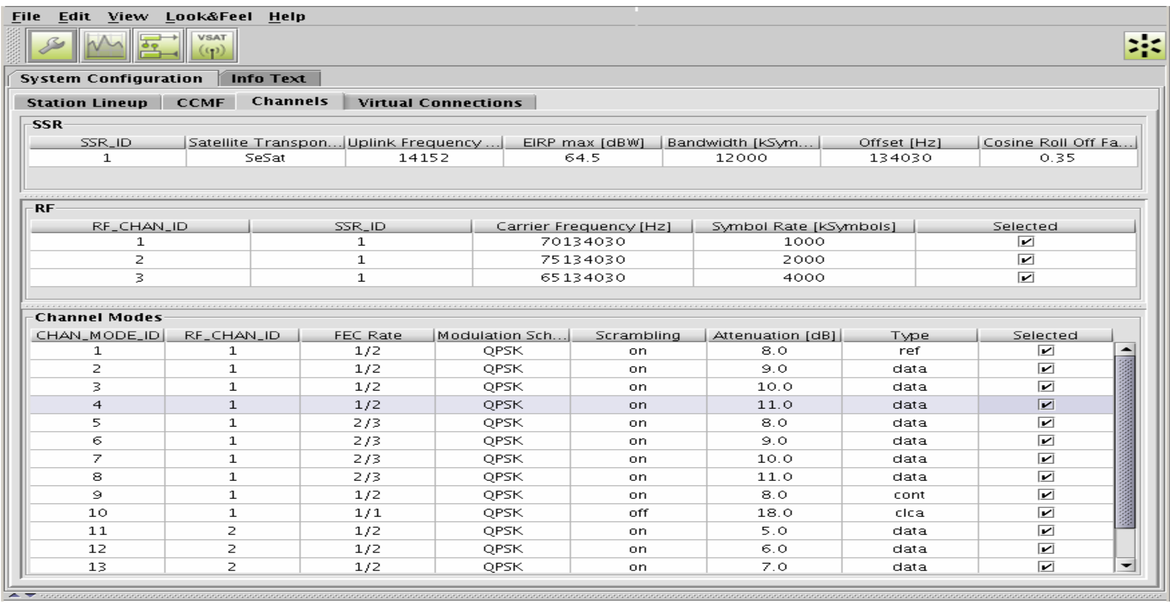


Figure 10: Channels Panel

The user interface mainly consists of three tables in the following order: SSR, RF and Channel Modes. Each row of the subsequent table contains a reference (SSR_ID and RF_CHAN_ID) to a row in the preceding table and thus is dependent on the configuration of this table. The independent cells in the table are editable either by editing the text field directly (e.g. all parameters in the SSR table and the Carrier Frequency) or by selecting a value from a pull down list (e.g. SSR_ID, Symbol Rate, FEC Rate, Modulation Scheme and Mode). The last column in the RF and Channel Modes table with the check boxes is provided to activate the settings in the system. A consistency check of the configured values is done before activation. Unselecting a table row means that these settings are no longer used by the system. If for example the operator unselects a row in the RF table all dependent rows in the Channel Modes table is unselected automatically. A row can be added or removed using the Edit menu item ‘Add channel’ or ‘Delete channel’. Only unselected rows can be removed from the tables. If the operator removes a row from the RF table all rows in the Channel Modes table, which are referenced to this row, is removed as well. If the operator presses the ‘Refresh’ View menu item, the current settings of the system are displayed in the tables.

The performance monitoring module provides a tool to produce a) time series, b) bar graphs, c) list loggers and d) x-y plots of selected parameters.

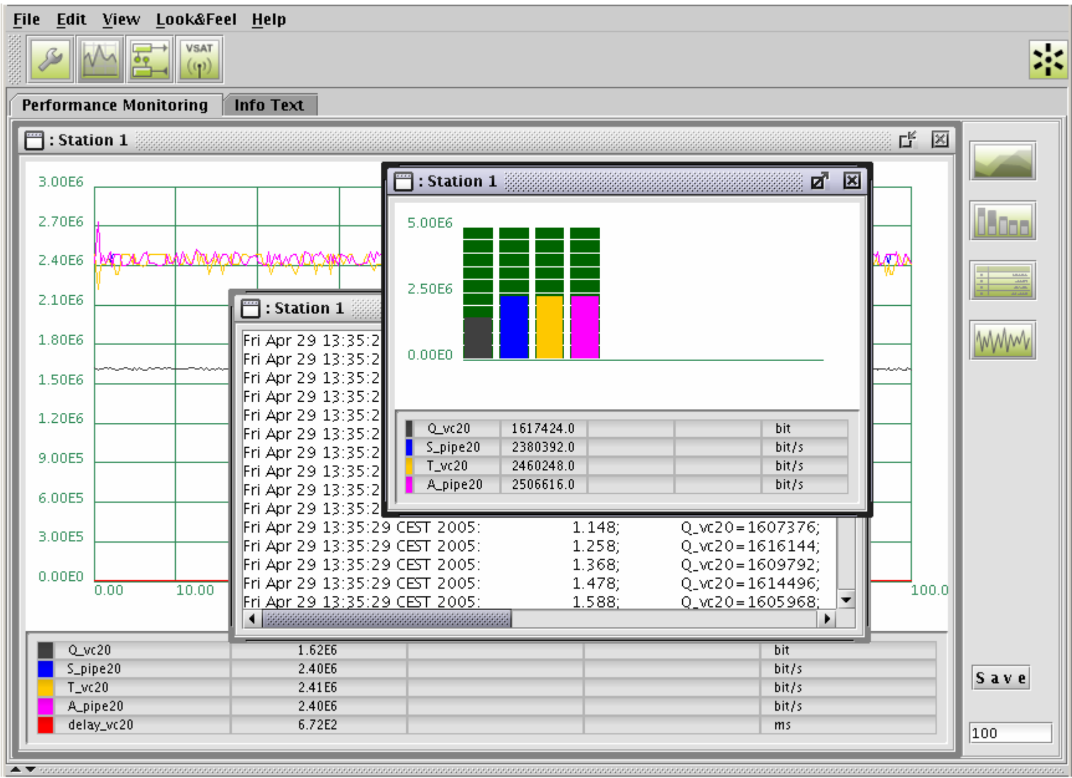


Figure 11: Line-graph, bar-graph and list-graph of performance parameters

More than one display type with up to ten parameters can exist simultaneously. Once a selection is committed it remains unchangeable. For a change of selection the current window has to be closed and a new window with the new selection has to be opened. The data to be displayed is retrieved from the system by polling. The polling rate is configurable from 1 ms upwards.

10 Summary

The L*IP satellite Gateway is a good example of a software radio system where all modules from the IP layer to the Modem interface were developed from one team. The competitive performance and flexibility for future extensions promises a good positioning on the market.

The innovative aspects can be summarized as follows:

- L*IP is an <u>open platform</u> offering a fully programmable gateway and a fully software-defined modem and codec (different modulation/synchronization and FEC coding schemes can be implemented in VHDL/C++).
- L*IP is a meshed system addressing the corporate user which has more symmetric traffic requirements, whereas DVB-RCS is a star network targeted at fast Internet access with asymmetric traffic. L*IP therefore can support highly interactive applications <u>between</u> user terminals in single hop (this comparison implies transparent transponders).
- L*IP uses enhanced IP encapsulation to reduce overhead. The variable burst concept is well suited to IP traffic.
- L*IP <u>guarantees QoS on IP level</u>. It handles different IP-based applications and services, each with different QoS requirements, in different queues through its dynamic DAMA system.
- A patent was gained with a unique Turbo codec approach
- The L*IP supports fade countermeasures by adaptive power, symbol rate and coding
- A low operational point of < 5,5dB

11 Acknowledgement

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12 References

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