Electronic News Gathering via a wireless broadband infrastructure

E. Pallis♣, A. Kourtis♦, G. Gardikis •, G. Xilouris♦
♣ University of East London, Electrical & Electronic Engineering Department, Longbridge Road, Dagenham, Essex, RM8 2AS, UK.
♦ Institute of Informatics and Telecommunications NCSR «DEMOKRITOS», Agia Paraskevi Attikis,15310 Athens Greece.
• National Technical University of Athens, Mobile Radiocommunications Lab., 9 Iroon Polytechneiou, 15773 Zografou, Athens, Greece.

e-mails: pallis@iit.demokritos.gr, kourtis@iit.demokritos.gr, gardikis@iit.demokritos.gr, xilouris@freemail.gr

Abstract: **-** This paper describes a prototype system that enables for Electronic News Gathering applications through a wireless broadband networking infrastructure. The network makes use of the TCP/IP protocol stack and operates in the frequency band of 2.4GHz by utilising Spread Spectrum (SS) modulation technique. The use of the TCP/IP enables for various additional applications (e.g. videoconferencing), while the use of SS technique in the wireless links provides for portability and low mobility. Furthermore, the utilisation of broadband communication channels in the forward and the reverse path enables for high quality full-motion video transmission. Performance evaluation results are also reported.

Key-Words:- ENG, WLL, Broadcasting, Multicasting, MPEG

1 Introduction

Advances in telecommunication and computer sciences, have enhanced the role of information-gathering by converging media and telecommunication industries towards a more dynamic stage: the "*Electronics News Gathering*" (ENG) industry. Companies of this industry and the news-teams that staff them are equipped with an array of high-tech apparatus, including mobile and portable stations that allow them to gather information and broadcast events as they unfold. The speed of transmission means that journalists can direct the world's attention to events as they transpire. The media no longer simply records events to be broadcasted later, but by editing and packaging a story at its point-of-origin they remove several layers of interpretation from the news gathering process.

Currently, an ENG system consists of a *portable* station that gathers raw video/audio information (at its point-of-origin) and a *central node* that processes and distributes it to the end viewers. Specifically, the portable station is usually a van, which is equipped with all the appropriate gathering devices (i.e. cameras, microphones, , transmitters, etc.), while the central node is the TV shack that broadcasts to the viewers.

The communication between the portable station and the central node is either satellite or terrestrial. Satellite communication enables for large distance (global) ENG applications, while terrestrial provides a cheaper solution for small distances (local).

In case of terrestrial ENG applications the communication between the portable station and the central node is via a one-way communication RF link using either analogue or digital technology. For other communication applications traditional voice communications technologies are utilised (e.g. GSM).

This paper proposes a networking configuration that enables for terrestrial ENG applications, which utilises interactive wireless broadband communication channels in the Industrial Scientific & Medical (ISM) frequency band, while enabling for portability and low mobility. The system uses a hierarchical architecture with two core subsystems: an *Agent* by the service origin (portable station) and the *Manager* that processes and broadcasts the gathered news (central node). The communication between the Manager and the Agent is based on the TCP/IP protocol stack using the unicasting/multicasting delivery methods. This enables for the provision of additional to ENG services (e.g. videoconference, text/voice messages, Internet access) between the agent and the manager, which is very useful for such applications. The broadband communication RF link between the agent and the manager is based on Spread Spectrum modulation technique conforming to the IEEE 802.11 standard.

Moreover, the use of the MPEG compression technique for the delivery of ENG services guarantees their high quality. Finally, the overall architecture provides for scalability and flexibility, while maintaining the installation and maintenance costs at a minimum, provided that the system is built around components of standard technology, commercially available.
The organisation of the rest of the paper is as follows: Section 2 presents the overall system architecture. The two core subsystems, i.e., the Manager and the Agent, are described in Sections 3 and 4 respectively. Section 5 reports on performance measurements and on the present operational status of the described system. Finally, Section 6 concludes the paper.

2 Overall System Architecture

The overall system architecture is depicted in Figure 1.
This figure describes two core sub-systems namely Agent and Manager. The Agent is responsible for the collection of events (as they unfold at their point-of-origin), while the Manager distributes the gathered information to the users via terrestrial TV channels.
In such a configuration an ENG Agent may utilise portable video and audio devices that capture the information, a compression/encoding chain for compressing/encoding the captured audiovisual information and a transmission/reception suite for a bi-directional communication with the manager. The compression/encoding chain consists of a typical MPEG-1 or MPEG-2 encoder. The communication between the manager and the agent is via a point-to-point 2.4GHz RF link that utilises the IEEE 802.11 communication protocol. In this context, this RF link makes use of the Direct Sequence Spread Spectrum (DSSS) technology, while featuring broadband capabilities (2Mbps for the specific equipment used).
On the other hand, the manager consists of the appropriate decompression/decoding chain using MPEG-1 or MPEG-2 decoders, the suitable transmission/reception devices for the bi-directional communication with the agent in the ISM band and a typical analogue TV broadcasting chain for distributing the gathered news to the users via the UHF band.
Moreover, the system architecture enables for simultaneous transmission of the video signal from a single transmitter (agent) to multiple managers using the multicasting delivery method, in case that only one ENG portable station is to be used on behalf of many broadcasting stations. It is of primary importance that the MPEG-x packets (generated at the agent's premises) are delivered on time, without variable and unpredicted delays (usually caused at the transport layer due to acknowledgements and/or flow control). In this context, the protocols used for the transmission of data should fulfil all the above requirements. Towards this, the described network utilises UDP protocol at the transport layer that provides for connectionless and unacknowledged data flow, which is suitable for real-time applications, while the use of the IP Multicast protocol at the network layer enables for multicasting. Instead of supporting a point-to-point (unicast) link, the IP multicast protocol makes use of the IP addresses specially reserved for this purpose (from 224.0.0.0 to 248.0.0.0) to route the datagrams from a single source to multiple destinations. Provided that all nodes that constitute the network support this extension of the IP protocol, the multicast datagrams, which are transmitted in one of the addresses mentioned before (or "multicast groups") are forwarded to every host that has declared to belong to this group. By agreeing on a common address to be used for the transmitter and the receivers, the video stream can be simultaneously distributed to more than one destinations. In order to demonstrate the multicasting delivery method a commercial software tool was used.

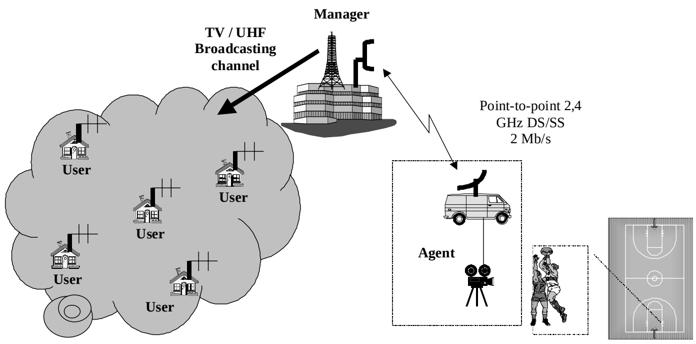

Fig. 1 Overall system architecture

3 Agent

Figure 2 depicts the configuration of the agent. The Agent captures an analog video signal as an input and converts it to a compressed digital stream, whose packets are transmitted across the wireless link. In this context, the analog signal is fed into a digitizer/MPEG 1/2 encoder card, which compresses it to MPEG-x format at constant bit rates. This rate is selectable by the operator and can vary from 500kbps up to 5Mbps. The MPEG encoder card is incorporated within a video server, and the appropriate video delivering software, which is commercially available. This software reads the MPEG stream from the encoder card, organizes it into UDP packets of 1400 bytes size each, and transmits it to the data network (Ethernet) as multicast traffic over the IP Multicast protocol. The transmission of the digital video signal over IP enables the user to multiplex it with other TCP/IP services, like videoconferencing, text/voice messaging or fast Internet access.

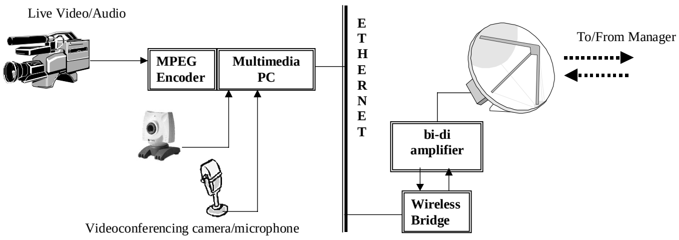

Figure 2: Agent structure

The agent communicates with the manager via the wireless bridge, which operates at the 2.4GHz ISM band by utilising the Direct Sequence scheme to spread the transmitted spectrum, while using the CSMA/CA protocol to avoid collisions. The nominal bit rate is 2Mbps (for the equipment used), but the actual throughput that can be achieved is much lower, as it will be shown later in this paper. The output RF signal is fed through a low-loss coaxial cable and a microwave amplifier into a 17dBi unidirectional antenna.

4 Manager

The configuration of the manager shown in Figure 3 does not differ greatly from that of the agent. The wireless Access Point (AP) is connected to an Ethernet hub, on which a multimedia PC is also attached. This PC is also equipped with a video decoder card and the appropriate multicast video client program, which collects the multicast MPEG packets from the network and reconstructs the original video signal. A video-out jack on the PC's video card enables to connect the client to a monitor or a video-mixing console for further processing and/or broadcasting. In case of broadcasting, the video signal is fed onto a TV modulator/transmission chain that broadcasts it to the users via the UHF band. Similar chain is applied to the sound channel.

Furthermore, the multimedia PC supports videoconferencing applications that enable the manager to communicate with the agent in real time. Figure 4 depicts the configuration of the manager in such a case. In this figure it is clearly shown the TV monitor, which displays the collected by the agent information, while the screen of the multimedia/videoconferencing PC displays the videoconferencing images of the agent.

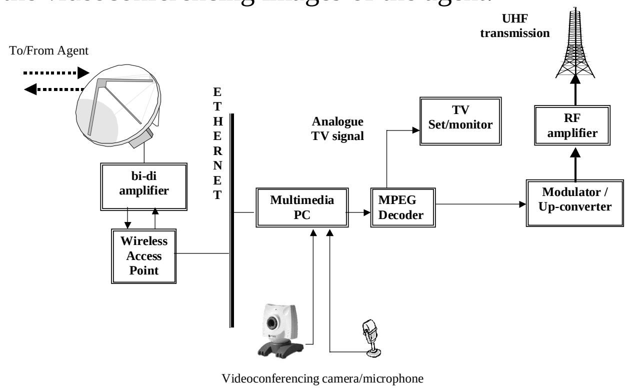

Fig. 3 Manager structure

Fig. 4 System configuration at the manager's side for monitoring the captured information and directing the agent

5 Performance measurements

The main goal for the optimum performance of the system described above is to allocate the proper amount of bandwidth for the multicast video stream and for the data services running simultaneously. In this context, a prototype system was developed for conducting performance evaluation measurements. This test-bed was ranging between a manager and an agent at 4.2km away. The RF link was established using unidirectional antennas properly aligned under Line Of Sight conditions. It is obvious that the bottleneck of the whole network lies in the wireless link, which can support data rates of nominal value up to 2 Mbps. This bandwidth must be shared between the video stream and the TCP

channel(s) used for other services – in our case, for videoconferencing.
The videoconferencing application used in the tests is based on the H.323 standard.
As for the digital video, the bandwidth needed is almost equal to the bit rate used for the encoding of the video signal. A 1Mbps MPEG-1 System stream will requires 1Mbps of the available bandwidth at any time. And since higher bit-rate means higher quality, the highest possible bit-rate must be chosen for encoding, provided of course that such a value will cause negligible transmission problems.
The question that arises is "what kind of transmission problems can occur while transmitting a digital video stream?" It must be stressed out that, unlike what happens in a TCP channel where all packet transmissions are acknowledged and the delivery of a packet is guaranteed, that is not the case with multicast IP traffic, where packets originate from the server and are broadcasted in the data network towards one or more receivers. If a packet is lost, it is never retransmitted. In a wireless link, packet loss can occur due to:

- Signal fading, (i.e. poor propagation conditions)
- CRC error (distorted bits) due to noise or other interference,
- Buffer overflow inside the wireless devices. In this case, the bitrate arriving in the device for transmission is greater than the rate that the device can handle. Excessive packets are discarded.

In order to measure the performance of the network, a special software tool was developed for the needs of this paper. This tool can perform real-time measurements of the packet loss across the wireless link under various load conditions. The developed software simulates the commercial software by splitting the MPEG stream into UDP packets and transmits them as multicast traffic at a rate equal to the bitrate used during MPEG-1 encoding. It is based on the client/server architecture and performs client-initiated video multicasting. The server side performs the multicasting at the client's request using configurable packet size and bit rate, and the client side collects the UDP packets and measures the QoS (bit rate, packet loss) of the wireless link.
To specify the optimum-encoding rate to be used in the platform, a collection of MPEG-1 video clips was used, created from the same analog source, but under different encoding bit rates.
The goal when choosing the MPEG-1 encoding parameters is to achieve the best video quality possible by selecting the highest possible bit rate, while at the same time keeping packet losses at a low value. It is evident that as the bit rate of the multicast stream is increased, the possibility of a packet being discarded is also increased due to transmitter buffer overflow This overflow can occur when the bit rate exceeds a specific threshold.
Measurements for bit rates under 500kbps have no meaning, since video quality in such rates is unacceptable [1-2]. Also, measurements for bit rates above 1500kbps were not taken, since packet loss is continuously increasing making these rates unsuitable for multicasting.
The above mentioned software tool was used to measure the relationship between transmission rate and packet loss. The results are depicted in Fig. 5.

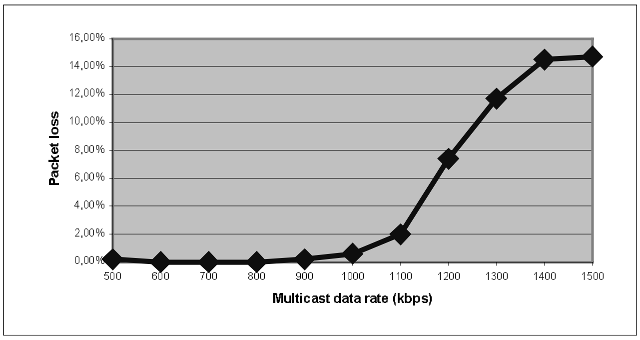

Fig. 5 Data packet loss vs. multicast rate

It must be noted that curves like that of Figure 5 are highly affected by the wireless link conditions [3-5]. A poor link will result in more discarded packets due to fading or bad CRC.
When transmitting digital video, packet loss results in gaps inside the MPEG1 stream. In this case, a typical decoder, based only on software processes, realizes that it cannot reconstruct the frames properly and it freezes the image until a whole frame can be formed correctly [1]. In the case of hardware de-compressor, such errors are less visible, since freezing rarely occurs, and the picture appears with some deformation.
Taking in mind the MPEG philosophy, one realizes that a packet loss in an I-frame makes the following B- and P-frame information useless, since the decoder has to wait for the next I-frame to arrive intact. It is easily understood that a single packet, even if is smaller in size than a video frame, when lost, can cause unpredictable damage in the decoding process.
To define the effect of packet loss on the perceptive video quality, the software measurement tool was configured to save onto the hard disk of the client PC each MPEG clip it received, with the impairments (gaps) which it contained. Every clip with a measured value of

packet loss was played by a software decoder (Windows Media Player), which was able to report the actual frame rate of the clip and the frame rate achieved by rendering the video frames that the decoder was able to form correctly. The video frame loss is easily computed as the ratio of the two above values.
Figure 6 (the horizontal axis is logarithmic) shows clearly that a slight increase in UDP packet loss in a multicast stream has a dramatic effect in video frame loss, which, along with the encoding bit rate are the two main factors that affect video quality.

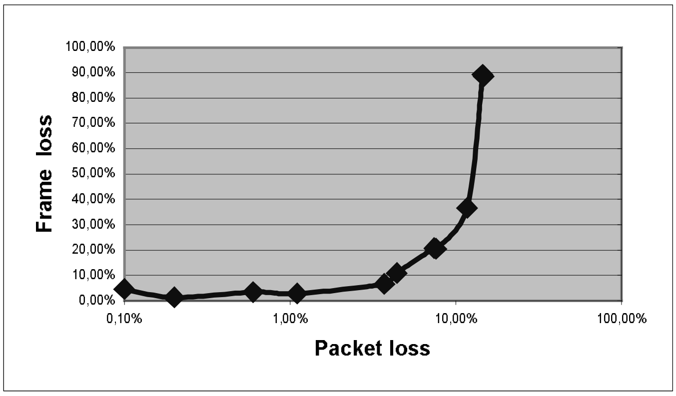

Fig. 6 Video frame loss vs. data packet loss in the MPEG stream

Combining the results of Figures 5 and 6, one can relate the multicast data rate and the video frame loss (Figure 7). From this curve comes the conclusion that one can safely select an MPEG bit rate of about 1100kbps providing an acceptable quality with low packet losses, which are unlikely to cause an annoying result. This was the rate that was chosen for the final testing of the system. Keeping this value constant, various bit rates were tested for the videoconferencing application. The results of the measurements showed that a rate of 128kbps per direction (256 kbps total) provides sufficient picture quality for the videoconferencing, and does not affect the performance of the multicasting, which takes place at the same time. Through the videoconference link, a person from the receiver side could easily give instructions to the transmitter on issues concerning the multicast stream parameters.

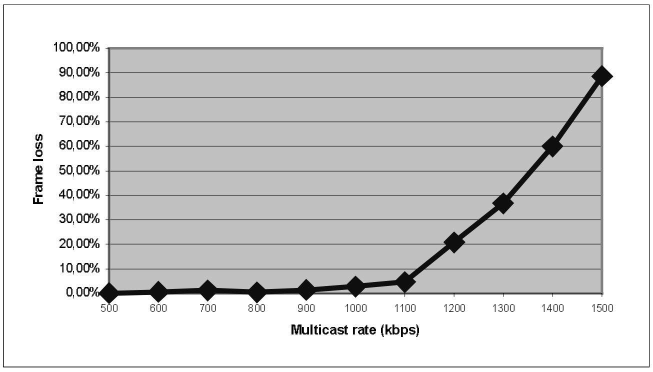

Fig. 7 Video frame loss vs. Multicast rate

6 Conclusions

This paper proposed an alternative way to implement an ENG platform, with the use of a broadband wireless data network infrastructure. The video signal was MPEG-1 encoded and transported over a CSMA/CA DSSS wireless link utilising the UDP/IP Multicast family of protocols. TCP/IP channels were available for data applications, like videoconferencing, which introduce new communication capabilities between the portable station (Agent) and the central node (Manager) of the system. This service was provided using cheap, easy to use and flexible equipment which is commercially available.
The results concerning the performance of the network refer to the specific equipment used. However, in the market faster wireless links can be found, capable of handling 10, 20 or even 100 Mbps. With such devices, one can use high-rate MPEG-2 encoding for professional broadcasting picture quality or even multiplex several video sources over the same link, while maintaining the capability of incorporating high-speed data services over the same link.
Measurement results showed the optimal bit-rates to use for video transport to ensure best quality for the given equipment. Under different conditions, one may conduct the similar tests and use the methods described in this paper to achieve the optimum use for a given wireless infrastructure.

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