Critical Data Streaming on WiFi Networks

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***Abstract* — Sending critical data streaming, usually associated with VoIP/video services, on 802.11 wireless networks is related with specific aspects of radio data links, not only as radio wave propagation but also as used modulation, protocols or codecs. The paper is identifying the main involved factors, limitations and "best effort" solutions to provide a QoS like support for critical data streaming applications as VoIP or related.**

***Keywords* — "best effort" transmissions, QoS, VoIP, WiFi networks, 802.11 standards.**

I. Introduction

A known problem in wireless networks is related to the difficulty of having QoS like parameterized data flow. They are necessary in order to allow optimum bandwidth, delay and latency so that critical data streaming, as multimedia information to be fully and in time recovered as media information at the receiver side. The involved factors are very different, from the technology related ones to working conditions.

Since some problems are as in wired networks they can be solved accordingly. The factors identified by the author as being directly involved are:

- Latency
- Packet loss
- Jitter
- Delay
- Bandwidth / Throughput

However the main problem is related with the particularities of data transmissions when using radio waves. The paper attempts to identify solutions to evaluate the possibility of sending the maximum number of VoIP (Voice over IP) streams as it is available at the application level. Since the problem is already solved in wired networks due to QoS prioritization technique, the article will focus on VoIP over WLAN (Wireless LAN) problems. That is necessary since 802.11b/g standards have no QoS direct mode. Some prioritization techniques are on the way (802.11e) but „the best effort" approach is still the main issue.

II. WiFi Networks Particularities

On 802.11 networks additionally factors have to be considered, the basic ones being the following:

- overlapping
- throughput
- radio bandwidth sharing
- legacy support and RTS/CTS
- range/coverage conditions

If we are considering a wireless bridge operating in 802.11b standard at 11 Mbps it will provide only 4-5 Mbps real speed at the OS interface while in 802.11g transmissions only 22-24 Mbps are available. So usually less than half is useable. The available throughput is also related to many other factors so that many times there is a big problem to assure a quality data flow.

The phenomena are due to different factors as that ones discussed bellow.

A. Radio Channels Overlapping

The difference between two adjacent channels is 5 MHz as specified in IEEE 802.11 standards. This frequency distance is based on the necessary frequency band at 3dB.

Overlap between the channels cause unacceptable degradation of signal quality and throughput. Basically 802.11b/g networks divide the spectrum into 14 overlapping, staggered channels having central frequencies at 5MHz distance specifying the center frequency of the channel and a spectral mask. However the energy of the channel extends further than these limits (±22 MHz from the center frequency). For channels supposed to not overlap, as 1 face to 6 but even for channels at extreme distances, 1 face to 11 for example, there still are interferences strongly related with the radio transmitting power on the involved channels. Laboratory tests shows that, throughput on a file transfer on channel 11 decreased slightly when a similar transfer began on channel 1, indicating that even channels 1 and 11 can interfere with each other to some extent [1].

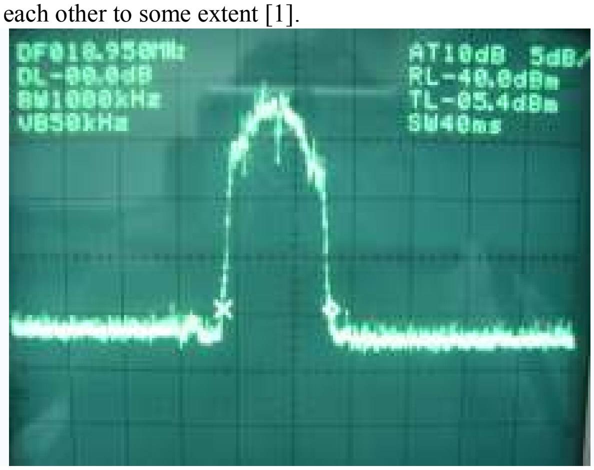

Figure 1. Spectral frequency bandwidth for 802.11g Access Point working on channel 8 (2.447GHz).

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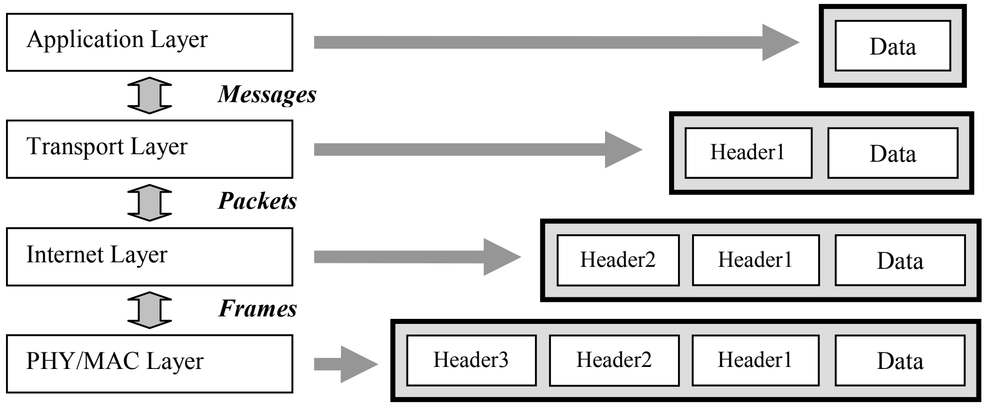

Figure 2. TCP / IP Hierarchy and associated overheads

As shown in figure 1, for an 802.11 Access Point working on the channel 8, the central frequency is 2.447GHz (between 2.436GHz and 2.458GHz if defined at 3dB) but spectral power distribution goes by far to about 20MHz so that 2 extra channels will be overlapped in the left side and another two in the right side. To not overlap we need to have 5 not used channels between any 2 active channels.

The measurements were done using a Hammeg Spectrum Analyzers and a directional Yagi antenna with 17dBi gain, at 3 meters distance from an Access Point working at full power (100mW) and having 2dB antenna.

Throughput / Payload versus Bandwidth

For any data transmission, wired or wireless, the available throughput (or payload) is lower then the channel bandwidth because of specific overhead as shown in figure 2. The body of the data frame carries information, such as TCP/IP and UDP packets. The payload size of the data frame body in 802.11 communications is even more limited, since most information requires multiple data frames to carry the entire load. For example 802.11b has a maximum raw data rate of 11 Mbit/s and uses CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) media access method defined in the standard. Due to the CSMA/CA protocol overhead, in practice the maximum 802.11b throughput that an application can achieve is about 5.9 Mbit/s over TCP and 7.1 Mbit/s over UDP. 802.11g operates at a maximum raw data rate of 54 Mbit/s, or about 24.7 Mbit/s net throughputs, also like 802.11a [1].

Simple calculus shows that the efficiency is 54% over TCP and respectively 65% over UDP. Since usually a mixture of transport protocols are involved in carrying out the information an average value is more realistic. For 802.11g operating at a maximum raw data rate of 54Mbit/s, the corresponding net throughput is about 24Mbit/s. This means that less than 50% is available in real world.

Some laboratory tests performed by the author reveal the results from the table 1.

As particularity, we have to note the described transfer speeds are possible only if no other interferences occur (no overlapping, no significant radio attenuation, infrastructure mode with one client to avoid bandwidth sharing). In these ideal conditions some papers [2] describe that critical application, like VoIP (as the most sensitive ones) can be performed theoretically allowing 11 to 15 VoIP streams for a 11Mbps channel. Commercial practical test shows that only 6 to 7 streams are possible at medium charge, depending on the involved equipment.

B. Radio Bandwidth Sharing

A radio AP usual works with many clients so that its maximum available bandwidth will be shared between these clients working, obviously, on the same radio channel. With infrastructure wireless LANs, which include access points, data frames do not travel directly between

Table 1: Available Throughput for different 802.11 situations

Distance (m)	802.11b (Mbps)	802.11g (Mbps)		
		802.11g only	Mixed env. CTS-to-self	Mixed env. RTS/CTS
3	5.8	24.7	14.7	11.8
15	5.8	24.7	14.7	11.8
30	5.8	19.8	12.7	10.6
45	5.8	12.4	9.1	8.0
60	3.7	4.9	4.2	4.1
75	1.6	1.6	1.6	1.6
90	0.9	0.9	0.9	0.9

Table 2: Data Rate versus Range

Data Rate [Mbps]	54	48	36	24	18	12	11	9	6	5.5	2	1
Range, 30mW with 2.2dBi gain diversity dipole antenna [m]	27	29	30	42	54	64	48*	76	91	67*	82*	124*

*802.11b fall back mode

clients. Instead, a wireless client sends the data frame to an access point, and the access point then sends the contents of the original data frame in a different data frame to the receiving client. So the AP bandwidth (and the radio space) is shared between the AP clients. Accordingly, two clients connected via an AP could use half of the available throughput, three clients will have one third and so on.

C. RTS/CTS and Legacy Support

Using RTS/CTS mechanism is the basic solution to avoid collisions in mixed 802.11b and 802.11g networks. When 802.11g only AP/clients are used then the communication occurs at the highest possible TCP throughput. The AP instructs the network not to engage any protection method against 802.11b traffic and the maximum throughput goes to about 24Mbps for a 54Mbps bandwidth.

In 802.11b/g mixed wireless networks one client is asking the permission for transmission sending a RTS message to the access point. At his turn, the access point is answering with a CTS message. Waiting for permission the client is not transmitting. Other clients receiving not-demanded CTS will also stop the send initiatives. Because like that the 802.11b clients are not transmitting simultaneously with 802.11g clients, this kind of collisions are avoided and the throughput is increased compared with no RTS/CTS solutions [1].

As defined by communication standards, there are two available protection mechanisms, RTS/CTS and CTS-to-self. RTS/CTS procedure is similar with that used in wired transmissions and is based on the demand for transmission (RTS) and than wait for confirmation as clear transmission conditions (CTS). The CTS-to-self is based on the transmission of a CTS message at 802.11b rate (to be processed by all clients in the mixed wireless network) just to clear the channel followed by 802.11g data rate transmission.

The table 1 also shows the expected maximum throughput for IEEE 802.11 environments for possible RTS/CTS communications [1].

In the real world, the throughput is even a little lower, basically due to the fact that is quite difficult to have a completely clear transmission media.

D. Range and Coverage

The 802.11b/g standards define a limited radio output power for the transmitter. Related to that there are some range and coverage limitations. Some behaviors came from radio wave proprieties: a signal transmitted in a lower area of the frequency spectrum will carry further than a signal transmitted in a higher band. Additionally, a longer waveform (from lower band in the spectrum) will tend to propagate better through solids (like walls and trees) than a shorter waveform.

Another fundamental rule is related with the fact that as data rates increase, range decreases. 802.11b uses DSSS to support data rates of 11, 5.5, 2, and 1 Mbps and 802.11g uses OFDM to support data rates of 54, 48, 36, 24, 18, 12, 9, and 6 Mbps. OFDM is a more efficient means of transmission than is DSSS, meaning that at a given range, higher OFDM-based data rates (802.11g) will be supported compared with DSSS-based data rates (802.11b).

Not only the transmit power is involved in range and coverage evaluation, but also the receive sensitivity. The selection of either DSSS or OFDM transmission type has an effect on the maximum power the transmitter can use, as well as the capability of the receiver, particularly at higher data rates. That’s because higher data rates require a high degree of acuity on the part of the receiver. High power coming from the radio's transmitter tends to desensitize the receiver, a phenomenon known as Error Vector Magnitude (EVM). Consequently increasing the transmit power tends to decrease the range of the device. Several environmental factors can also have a dramatic impact on range and resulting coverage area.

If forcing the equipment to only work at a defined speed and not to connect at lower ones or watching at what distance the equipment is automatically switching its data rate, a coverage area for different data rates available in 802.11g communications was measured. The obtained results are shown in the table 2 [1].

III. An Exemple of Critical Data Streaming Calculation for VoIP on WiFi Networks

A typical VoIP packet at the IP layer consists of 40-byte IP/UDP/RTP headers and a payload ranging from 10 to 80

Table 3: Codecs proprieties

Codec Type	GSM 6.10	G.711	G.723.1	G.726-32	G.729
Bit rate BR (Kbps)	13.2	64	5.3/6.3	32	8
Framing interval T (ms)	20	20	30	20	10
Payload PL (Bytes)	33	160	20/24	80	10
Packets speed PS (packets/sec)	50	50	33	50	50

bytes, depending on the codec used. From the beginning the efficiency at the IP layer for VoIP is already less than 70% (20% to 66% depending on the codec). The analog or PCM voice signals are encoded and compressed into a low rate packet stream by codecs. Table 3 lists the attributes of several commonly used codecs. Generally, the codecs generate constant bit-rate audio frames consisting of 40-byte IP/UDP/RTP headers followed by a relatively small payload. For example for the GSM 6.10 codec the payload is 33 bytes. The time between two adjacent frames is 20 ms, corresponding to a rate of 50 packets per second per VoIP stream [2]. Other rates codec proprieties are also shown in the table 3.

On wired networks the VoIP problem consist in the delivery of the data frame stream in such a manner so that the voice could be constantly and continuous rebuild. Since VoIP packages are launched as the voice samples appear the only problem is that the individual packages to be delivered to the destination with an appropriate rate. While in wired networks this problem can be solved by packet prioritization, in 802.11 networks other rules have to be considered.

As we can see in the codec proprieties table (table 3), the way of packing the VoIP date is quite different, depending on the used codec [3]. Different bit rates are involved. The correlation between codec specified parameters are as bellow:

$$BR = PS \cdot PL \cdot 8bits \quad [Kbps] \tag{1}$$

In general:

$$PS = \frac{1}{T}[packets/\sec] \tag{2}$$

The only exception is for G.729 where two frames are combined into one packet so that the value above has to be divided by 2.

For o single VoIP stream, we can evaluate the voice only associated maximum VoIP data rate as

$$B_{VoIP\ MAX} = \frac{PL}{T} \cdot 1000 \cdot 8b \quad [Kbps] \tag{3}$$

For a common used codec as G.711, having 50 packets per second each carrying a payload of 160 bytes, we have

$$50 \text{ packets/sec x } 160 \text{ bytes} = 8 \text{ KBps} = 64 \text{ Kbps} \tag{4}$$

The bytes stream structure for usual audio codecs is presented in the table 4. As we can see, every layer has its own contribution to the overall transferred bytes.

To calculate the number of real possible simultaneously VoIP sessions we have to note that the VoIP services are full duplex, therefore to streams are actively in the same tome on radio environment and on the same frequency channel. Starting from this point we can evaluate the VoIP streaming parameters from a different point of view [3].

The number of VoIP streams (N_{VoiP}) over a known bandwidth channel could be calculated as bellow:

$$N_{VoIP} = \frac{B_P}{B_{VoIP}} \tag{5}$$

with

B_P - Bandwidth available as payload/throughput [Kbps]

B_{VoIP} - Bandwidth necessary for one VoIP stream [Kbps]

Where B_{VoIP} is the necessary bandwidth for one full duplex VoIP communication:

$$B_{VoIP} = \frac{L_{VoIP}}{T} \cdot 2 \cdot 11 \cdot 1000 \quad [Kbps] \tag{6}$$

L_{VoIP} – Length of a VoIP data packet, including all overheads (sum of VoIP package length and overheads for RTP, UDP, IP, MAC, PHY levels), different for each codec [Bytes];

T – Time interval of VoIP full packets or framing interval [ms].

Based on above formulas and evaluations the maximum number of voice streams over a wireless network could be calculated. We can consider as evaluation example G.711 codec but the principle is the same for the other ones. For G.711 we have a 20ms time between packets so that a total number of 50 data packets are sent in one second. For a full duplex voice conversation the number has to be doubled to 100 packets per second. The total number of bytes to be transmitted together with one voice packet is 260 bytes with the contribution of voice packet length and overheads (RTP, UDP, IP, MAC and PHY). Accordingly we could state now that for G.711 an amount of 26 kilobytes per second has to be transmitted. Knowing that a byte has 8 bits and it needs at least 2 to 3 supplementary bits to point the word beginning and end (as start and stop bits), the involved bandwidth goes to 286 Kbps for one VoIP channel.

For G.729 we have 110 bytes and 2 x 100 packets per second, than the calculated bandwidth is 242 Kbps.

Knowing the necessary data flow speed for each codec, the evaluation can go farther considering different networks payload capacity. If the network is 802.11b type and it is working at maximum data rate of 11Mbps, as

Table 4 Voice packet length and overheads for different codecs

TCP/IP layer	Voice codec	G711	G729	G723.1
Application layer	Packet inter-arrival time [ms]	20	10	30
	Voice packet length [bytes]	160	10	24
	RTP layer overhead [bytes]	12	12	12
Transport Layer	UDP layer overhead [bytes]	8	8	8
Internet layer	IP layer overhead [bytes]	20	20	20
Data link sublayer	MAC layer overhead [bytes]	36	36	36
Physical sublayer	PHY layer overhead [bytes]	24	24	24

already shown above (table 4), we can use a throughput of about 6 Mbps in ideal conditions and for short distances. For a distance at the half of the covered range, the throughput goes to about 4Mbps (see table 1). This throughput can carry about 13 VoIP G.711 or 16 VoIP G.729 single streams. While a phone call involve two clients, if both are connected on the same Access Point, the number of pair calls will be divided by 2 [3].

There are different approaches but all of them conclude to some quite similar values as the maximum number of possible VoIP streams on an 802.11 radio channel. As example, for 802.11b communication (2.4GHz, 11Mbps), using G.711 codec, the number of simultaneously VoIP stream goes from 13 [4] to 15 [5] so that a value of 13 obtained above seems to be a realistic one. For G.729 the values are from 15 [4] to 34 [5] while our results shows 16 streams.

The calculations above are not absolute since the theoretical evaluation doe's not take into account lots of other problems which may occur. Some equipment producers [6] recommend considering only 60% of the available bandwidth. Even like that, we have to keep some conditions bellow a certain limit: the coverage distance, the radio interference and the number of other WiFi clients on the same hotspot.

IV. Conclusion

Many up-to-date applications (VoIP, VoD/IPTV) require a real-time data streaming strictly defined at the level of bandwidth and latency (at least constant values are needed). Sending critical (as throughput and latency) data streams on wireless networks have a higher level of risk. However a limited number of VoIP calls over 802.11 networks are always possible, the problem being related with the evaluation of this number in certain environmental conditions.

A wired QoS mechanism is based on IP control at the transport level (TCP/IP) to assure a defined bandwidth for a certain service or user. Unfortunately, TCP/IP cannot guarantee this kind of purpose, it just make a "best effort" to do it. The wireless connections are defined at the physical level or MAC/PHY level (host-to-network level in TCP/IP model) so they are not “viewable” in the next level (Internet level, IP based), except for configuration issues and this is the point were the QoS problems arise. As the physical layer is a CSMA/CA radio environment, based on the principle “verify and transmit only when the channel is not busy” or “listen before talk”, the data packets have to wait a non-deterministic time interval before being launched.

The 802.11 networks are introducing important limitations which are quite difficult to be managed in radio based networks in order to solve the audio streams. Some specific problems need to be taken into account for a successful call delivery. When voice data flow has to travel over wireless media like 802.11, compared with a wired network, supplementary overheads are added basically at the physical level (known as PHY level in 802.11 standards).

In specified environments some QoS like proprieties could be invoked under the strictly defined conditions. The new 802.11e standard is already considering few rules at the radio level packet flow and that is expecting to allow some prioritizations by introducing priority levels at MAC/PHY level (basically thus stations with lower-priority traffic must wait longer than those with high-priority traffic before trying to access the medium). However many 802.11b/g equipments are already in use and pseudo QoS rules here presented became useful when critical data flow are associated.

VoIP services used on wireless LAN looks like a mobile telephony. A PDA or WiFi Phones can be used as telephone. More, VoIP phones seem to be an alternative solution to GSM telephony for limited areas. Many companies are now wireless LAN covered and that is an important challenge to go further on this field.

References

[1] Alin Dan Potorac, Eugen Coca – “QoS Considerations for 802.11 Networks”, *European Conference on the Use of Modern Information and Communication Technologies*, ECUMICT ‘06, Ghent, Belgium, pag.353-362, 30-31 Martie 2006, ISBN 9-08082-552-2.

[2] Wei Wang , Soung Liew, Victor Li – “Solutions to performance problems in VoIP over a 802.11 wireless LAN”, *IEEE Transactions on Vehicular Technology*, Vol 54, Issue 1, Jan.2005, ISSN: 0018-9545, pag.366- 384

[3] Alin Dan Potorac, “An Approach on VoIP over Wireless TCP/IP Networks”, *The XI-th International Conference “Man in the Knowledge Based Organization”* editor Land Forces Academy Publishing House, Sibiu, 23-25 Noiembrie 2006, ISBN 973-7809-51-3 / 978-973-7809-51-3.

[4] Matthew Gust – *“Maximum Number of VoIP Telephone Calls per 802.11b Access Point”*, Internet article, http://www.oreillynet.com/etel/blog/2005/06/, Jun.2005

[5] Dongyan Chen, Sachin Garg , Martin Kappes, Kishor S. Trivedi – “*Solutions to Performance Problems in VoIP over 802.11 Wireless LAN”*, Duke University, Durham, Internet article, http://www.ie.cuhk.edu.hk, 2002

[6] CISCO Systems - *Cisco 7920 Wireless IP Phone Design and Deployment Guide*, www.cisco.com, 2005