White Paper

VoIP Service Over Wi-Fi Networks

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1. Introduction

VoIP (Voice over IP) technology enables packet based IP networks to carry voice. With VoIP, operators and service providers can offer voice telephony service as well as traditional data service over the same data infrastructure, thus increasing revenue stream and improving business models.

Constructing a VoIP telephony service over a wireless network, and specifically VoIP over Wi-Fi (VoWi-Fi), requires basic understanding of the technology in order to achieve high quality telephony and to maximize capacity.

This paper introduces basic elements of VoIP technology and VoWi-Fi and explains how these elements influence voice capacity and quality over wireless networks. The paper also provides guidelines for building a Wi-Fi network based on Wavion products for maximizing VoIP performances and data capacity.

2. VoIP Technology

2.1. VoIP characteristics

VoIP technology uses coder-decoder (CODEC) for compressing/decompressing the sampled voice signal. These samples are packetized and carried over RTP (Real Time Protocol) over the channel. All CODECs sample the 4 KHz voice band 8000 times per second. A single voice sample is converted to 8 bits, which creates 64kbps digitized voice rate.

Different CODECs exist providing different trade-offs:

1. G.711is the basic CODEC. In both its a-law and µ-law versions it samples the 4 KHz voice band at a rate of 8000 samples per second, each with 8 bit resolution, yielding a stream of 64Kbps.
   This CODEC, referred to as *non-compressing CODEC*, requires low computation complexity and provides good voice quality. However, it consumes 64Kbps, which is relatively high compared to other CODECs.

2. G.729 is a more advanced CODEC. It samples the voice band 8000 times per second, with 16 bit resolution, but then performs compressing, resulting in a stream of 8Kbps. It is referred to as a *compressing CODEC*, and it presents a good tradeoff between quality and the bandwidth used per connection.

3. G.723 is another *compressing CODEC.* It has various versions, some generating a 6.3Kbps stream and others a 5Kbps stream.

2.1.1. CODEC Parameters

CODECs have different parameters:

1. Frame length
The frame length of a VoIP RTP packet is defined by the payload the CODEC generates plus the overhead of IP and Ethernet or Wi-Fi (or any other layer 2 protocol) headers.

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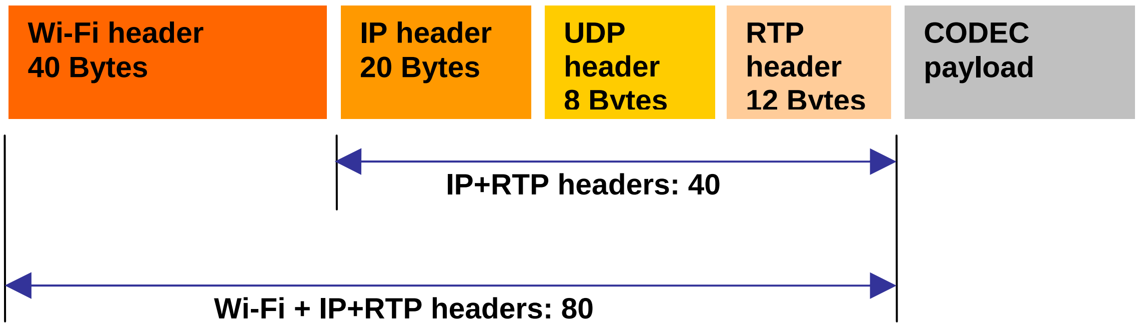
2. Packetization time and Packets per Second

The VoIP works by transmitting a voice packet periodically, one every fixed amount of time. The time that passes between the start of two consecutive packets, generated by a CODEC, is referred to as “packetization time” and is usually measured in milliseconds (ms). This parameter can be easily translated to Packets per Second (PPS) of the CODEC.

3. VoIP Bandwidth

As already mentioned, different CODECs may generate different bit rates. Note that the headers added to the CODEC packets (e.g. Wi-Fi headers and IP headers) should also be considered when measuring the total bandwidth a voice stream consumes.



The following table summarizes traffic parameters of VoIP packets utilizing different CODECs with various parameters:

CODEC name	Packetization time	PPS	CODEC Payload Length	CODEC Bit Rate	Wi-Fi packet length	Wi-Fi bandwidth
G.711	20ms	50	160 Bytes	64Kbps	240 Bytes	96Kbps
G.711	30ms	33.3	240 Bytes	64Kbps	320 Bytes	85Kbps
G.729	20ms	50	20 Bytes	8Kbps	100 Bytes	40Kbps
G.729	40ms	25	40 Bytes	8Kbps	120 Bytes	24Kbps

The table shows that packetization time has a marginal effect when using non-compressing CODECs like G.711, but that it has significant effect when using compressing CODECs like G.729 (about 80% bandwidth increase by utilizing 40ms packetization time vs. 20 ms).

2.2. VoIP Quality

VoIP quality is measured in MOS (Mean Opinion Score) values. The MOS takes into account the three most critical parameters for VoIP quality: packet loss, packet delay, and jitter (variance of packet delay). The MOS values are in the range 1-5, with MOS of 3.8 and higher considered to be high-quality call.

We next present a short description of these parameters.

a. Packet loss
Packet loss occurs when some of the packets sent do not reach the destination. Packet loss degrades voice quality mainly when compressed CODECs are used (e.g. G.729 or G.723). Packet loss occurs because of errors in receptions, which may happen due to:

a. Poor link quality
b. Collisions with other clients' traffic
c. Collisions with interferences

b. Delay and jitter
Large delay and/or jitter (non-fixed delay) in packet reception cause voice quality degradation. Delay and jitter may occur due to:

a. Packet retransmissions to overcome packet loss. The amount of retransmissions may vary from one packet transmission to another.
b. Lack of priority to voice packets over data packets
c. Overloaded of the network

3. The Challenge of VoIP deployment over Wi-Fi Networks

Wi-Fi technology was designed to work over license exempt bands (2.4GHz and 5.xGHz) and as such the Wi-Fi protocol was built to handle cases of interferences and collisions which are very common in license exempt bands. Before a packet is transmitted, the originator senses the air and only when a clear channel is detected, the packet can be sent. Moreover, each packet is acknowledged (ACK) by the recipient to indicate successful transmission. Thus, when the ACK is not received by the sender, the packet is retransmitted.

The Wi-Fi protocol described above was optimized for best-effort data transmissions and not necessarily for real-time transmissions like VoIP traffic, which is delay/jitter and packet loss sensitive. To overcome part of the problem, an extension to the Wi-Fi protocol standard was defined to enable over-the-air prioritization of real time traffic over data traffic. This IEEE802.11e standard, also known as WMM (named by the Wi-Fi Alliance), is the standard enhancement for QoS over Wi-Fi. WMM handles 4 Access Categories (Voice, Video, Best Effort and Background) and ensures 4 different levels of priority for accessing the wireless media.

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3.1. Additional challenges:

3.1.1. Collisions and retransmissions

The Wi-Fi protocol copes with collisions by retransmissions. However, those collisions and retransmissions affect the network performance in two ways:

a. They reduce the total network capacity

b. They increase the delay and jitter of packet transmission.

These effects are even more critical in networks with hidden nodes, i.e. where all clients "hear" the base station well, but don't "hear" each other's transmissions, since the amount of collisions and retransmissions in these networks may increase significantly.

Therefore, one of the important challenges when designing a VoWi-Fi system is to minimize the occurrence of collisions in the Wi-Fi network.

3.1.2. Mobile Wi-Fi clients

Another issue that may affect the network capacity and thus the VoIP quality is mobile Wi-Fi clients. Such clients may start by getting a very good signal quality, but as they move they may get into a significantly worse position with low signal quality. Working in low signal quality means lower modulations and hence longer time a packet occupies the air. This, in turn, increases the chance that other Wi-Fi clients will collide with this long transmission.

Another effect of clients working in low modulation is the increased delay (and thus jitter) caused to the entire network, which means mobile clients with poor signal quality may affect all calls in the network and not just their own.

4. VoWi-Fi Deployment Recommendations

The deployment recommendations are divided into classes - recommendations that are Wi-Fi related and recommendations that are VoIP specific.

4.1. Wi-Fi Related Recommendations

4.1.1. QoS planning and Support

In order to support coexistence of data traffic and real-time traffic such as VoIP, the network should support priority marking and handling (queuing) throughout the network. To this end, we recommend that the entire network uses the same QoS scheme (e.g. marking priority using the DSCP field and values).
In order to support proper priority handling over the wireless media of the access network, it is important that both the base station and the Wi-Fi client support WMM (the Wi-Fi standard for wireless priority handling).

For the downlink traffic, it is the responsibility of the base station to map the LAN priority, whether it is TOS, DSCP, or VLAN priority into WMM Access Category priority, and

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access the air interface accordingly. For the uplink traffic, it is the responsibility of the Wi-Fi clients.

It is recommended that all VoIP and data traffic is sent from behind the CPEs. This way, the CPEs take care of WMM-marking the traffic with the correct priority and access the air interface accordingly.

4.1.2. Proper Base Station installation

A proper installation of the base station is one of the key factors for achieving good Wi-Fi network. It is important to place the Wi-Fi base station in a location that provides good coverage as well as high data rates, thus achieving maximum capacity with low delays and jitter to all clients.
Working at high data rates achieves two targets:

a. It minimizes the time that the air is occupied with the transmission of a packet.

b. It minimizes the chances of collisions with other packets. Thus, increasing the chances the packet will be received successfully.

With Wavion's beam-forming technology, the base station assures that links to the clients are optimized in both the uplink and downlink, thus providing the highest data rates for each client.

4.1.3. High Gain CPEs

To insure the best possible links, it is recommended to use outdoor high-gain CPEs. It is further recommended to have connectivity at SNRs of 15dB or higher. This will help maintaining steady network capacity level as well as minimizing the jitter, thus assuring the overall quality of the VoWi-Fi network.

Along the same lines, it is recommended that the CPE have a good rate-adaptation algorithm, which is optimized for passing VoWi-Fi packets. Such algorithms, for example, may prefer to work at a modulation of 36Mbps without packet loss, instead of trying to raise the modulation to 48Mbps or 54Mbps and end up losing packets.

4.2. VoIP Related Recommendations

The VoIP equipment should take into account several parameters when selecting the ATA (or the VoIP phone). Proper selection will significantly improve both voice quality as well as network VoIP capacity.

4.2.1. Prioritizing Data

The ATA device needs to be able to mark the proper QoS of the VoIP traffic, according to the same QoS scheme used by the rest of the network. It is strongly recommended that the QoS marking in the ATA / VoIP phone be mapped to the WMM access category classification at the wireless CPE.

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4.2.2. Compressed CODEC

VoIP traffic is best served by a compressed CODEC, such as G.729. This CODEC provides a good trade-off between the bandwidth of the VoIP traffic and quality of the compressed data.
Though there are other compressing CODECs, like G.723, which compress the voice stream even further, the gain in bandwidth is so marginal (because of the large overhead of the IP headers and Wi-Fi headers) that most operators do not find it worth exploiting because of their lower voice quality. .

4.2.3. Packetization

As described above, the packetization time of the CODEC is a key factor in assuring low consumption of the Wi-Fi bandwidth. It is recommended to use the compressed CODEC G.729 with packetization of 30 or 40msec, which reduces the packets per second rate (PPS), without significantly increasing the delay.

4.2.4. Jitter Buffer

It is recommended to use VoIP equipment (ATA or VoIP phone) with an adjustable and dynamic jitter buffer. Such functionality can better tolerate delay variance in the network, while optimizing overall delay. This improves the overall VoIP quality the end customer will experience.

4.2.5. T.38 Support

In order to provide reliable fax communication through any VoIP network, especially VoWi-Fi, it is recommended to use ATA which support T.38 fax protocol.

5. Building VoWi-Fi network with Wavion Equipment

We next describe a typical VoWi-Fi Network tested in the field. This network represents a typical deployment and includes significant interference, hidden CPEs, NLOS CPEs, near and far clients and multiple concurrent calls in parallel to multiple data streams. This real life scenario is typical when the WiFi out door network provides business connectivity.

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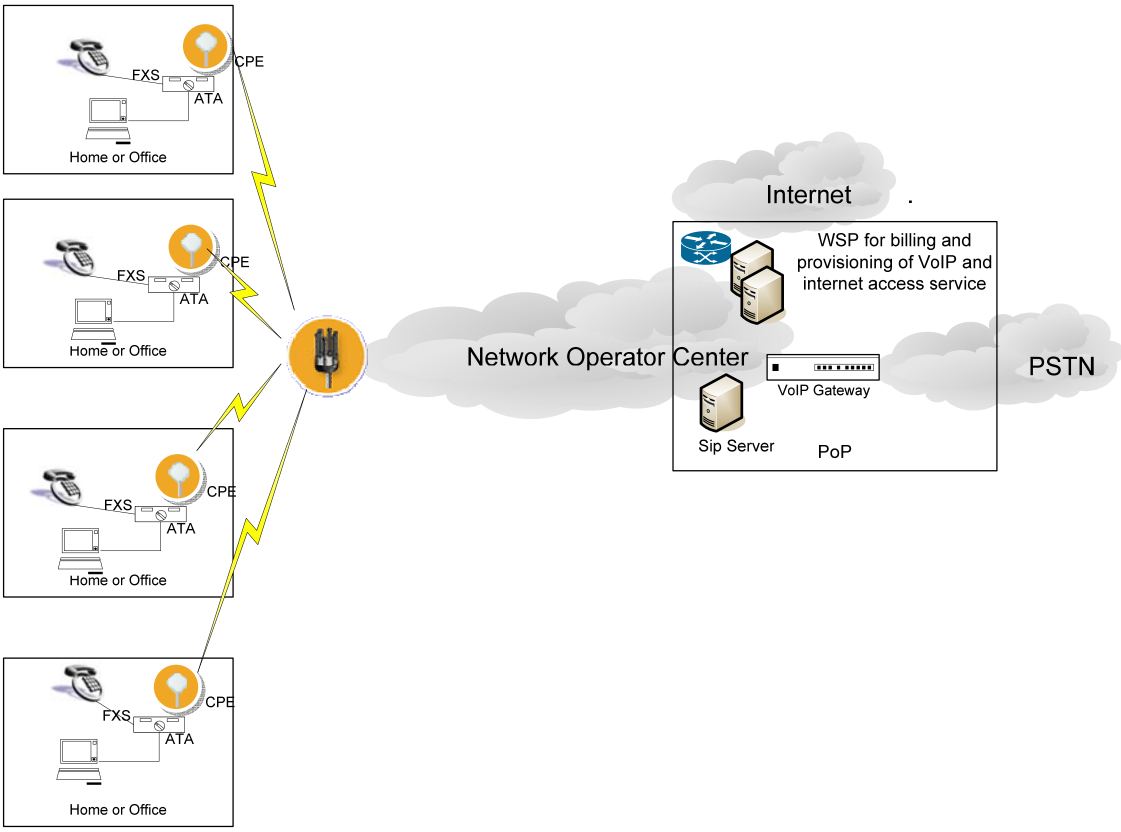

The network includes the WBS (2400 and 5800) base station and few 10s of CPEs (recommended Ubiquiti NS5/PS5/NS2/PS2). These CPEs were connected at different locations some hidden to the others, all in good SNRs. At the offices or homes behind the CPEs there are computers (passing data traffic) and ATA connected to POTS (Plain Old Telephone System). The ATA was set to tag the VoIP data with DSCP priority, which is in turn mapped to high priority WMM Access Category by the CPE.

At the back office, behind the WBS (2400 and 5800), reside the VoIP Gateway and SIP Server. These products may vary from low-end free servers, such as Astrisk (as used in this specific setup), up to high-end, expensive, full redundancy, providing 99.999% uptime servers. The specific products are usually determined by the VoWi-Fi network operator according to his needs.

In the field trial, the above setup yielded over 20 concurrent calls with additional data throughput. During the trial, we made sure the network is constantly congested with over capacity data in order to validate the performance of the QoS mechanism in real life scenario (interference, hidden nodes etc.). Throughout the test, calls had MOS of 3.8 and above.

6. The Wavion Advantage

Wavion's Wi-Fi solution provides several advantages which make the VoWi-Fi work significantly better. These advantages improve the VoIP over Wi-Fi performance both in terms of voice quality and in terms of voice capacity.

6.1. Advantage #1: Beamforming

Wavion's unique and powerful beamforming technology improves the VoIP performance of Wavion's access networks in more than one way:

a. The beamforming technology improves the link budget by 10db in average, both in uplink and downlink. This link budget improvement implies that Wi-Fi clients will operate in higher modulation. Working in a higher modulation implies higher rates, which in turn implies that voice packets are transmitted faster and thus reducing the probability of collisions over the air. This improves the capacity and also reduces the delay and jitter, thus further improving the voice quality.

b. The beamforming technology focuses the beam to the Wi-Fi client and thus attenuates the interference from other directions. This increases the probability of successful VoIP traffic reception.

c. The beamforming technology provides a much steadier Wi-Fi signal and maintains high operation rates (and thus high capacity) with smaller effect of fading. This steady capacity ensures high VoIP quality (MOS) at all times.

For further information on beamforming, please refer to the Wavion's "ultimate solution" white paper (can be downloaded from: http://www.wavionnetworks.com/innerData/pdf/ultimate_solution_white_paper.pdf)

6.2. Advantage #2: Optimized Wi-Fi MAC layer

To provide enhanced VoIP performance, Wavion has developed an optimized operation mode for real time applications. This operation mode uses Wi-Fi MAC layer and WMM enhancements to ensure better over-the-air prioritization of real-time packets over data packets. .
This optimization enables better performance of mixed voice and data traffic through the same Wavion base station, using standard Wi-Fi CPEs.

6.3. Advantage #3: Optimized rate adaptation

In addition, the rate adaptation algorithm was also optimized, to better support real time applications. It ensures operation at higher rates, thus minimizing rate drops and base station capacity changes.

Running VoIP over Wi-Fi implies a larger number of shorter packets per second. This implies a higher probability of collisions. Under normal operation mode, this will result in AP and client lowering their modulation rate, and thus reducing network's capacity. When operating in Wavion's QoS optimized mode, the rate adaptation algorithm takes this traffic pattern into consideration and maintains a higher and steadier modulation rate per CPE, thus providing higher network capacity.

To summarize, the combination of the above three technological advantages provides a significantly better voice quality without sacrificing voice call capacity and data capacity. Further more, the Wavion VoIP solution allows the operator to use any 802.11a/b/g standard compliant CPE (supporting WMM) to achieve the enhanced VoIP performance.