Integrated Network Solutions

Simply Building a VoIP Service Over Wireless Networks

White Paper

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1 Introduction

VoIP technology enables packet based IP networks to carry voice, a mission critical application. 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 requires basic understanding of the technology, in order to achieve toll quality telephony and to maximize capacity.

This paper introduces basic elements of VoIP technology and explains how these elements influence voice capacity and quality over wireless networks. Once the technology is introduced, this paper provides guidelines on selecting proper VoIP gateways to work with wireless networks, as well as configuring BreezeACCESS VL based wireless network for maximizing performances and capacity.

2 VoIP characteristics you should know

VoIP technology, SIP based or H.323 based, uses coder-decoder (CODEC) for compressing/decompressing the sampled voice signal and RTP over UDP connections for carrying the packetized voice data over the channel.

Different CODECs exist to allow different optimizations:

1. G.711, in both its a-law and μ-law versions, is the most common and basic CODEC. It samples the 4Khz voice band at a rate of 8000 samples per second, each with 8 bit resolution = 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, also samples the voice band 8000 times per second, with 16 bit resolution, but it then performs a compressing algorithm, resulting in a stream of 8Kbps. This CODEC, referred to as a *compressing CODEC*, optimizes the bandwidth used per connection
3. G723 has various versions, some supporting a 6.3kbps sampling rate and others a 5kbps sampling rate. This CODEC is also referred to as a *compressing CODEC*

Some of the principal CODEC parameters are:

1. Frame length
   The payload a CODEC generates and the overhead of IP and Ethernet (or any other layer 2 protocol) headers

2. Frame rate
   The amount of time that passes between the start bit of two consecutive packets, generated by a CODEC. This also translates to **packets per second (PPS)**

The following table summarizes common VoIP CODEC characteristics:

CODEC name	Basic rate – Basic Payload – Total Packet Length
G.711 a-law / µ-law	200 pps (5ms) – 40 Bytes (payload) – 98 Bytes (total packet size) ***Abbreviation**: basic rate generates 200 packets per second (a packet every 5ms) with total length of 98 bytes*
G.729	100 pps (10ms) – 10 Bytes – 68Bytes
G.723r63	33 pps (30ms) – 24 Bytes – 82 Bytes
G723r53	33 pps (30ms) – 20 Bytes – 78 Bytes

Table 1: common VoIP CODECs' characteristics

It is important to note that most VoIP Gateways concatenate several basic rate VoIP packets, thus actual packet size is larger and ppp is smaller (further info in the following section).

3 When VoIP goes wireless

Wireless networks have their own advantages and limitations, some of which affect VoIP service performance. The following are important parameters that affect VoIP capacity and quality over wireless networks:

1. Packet loss
   Occurs when some of the data sent does not reach the other end. Packet loss events degrade voice quality. Packet loss is more common in wireless than in wire-line networks. Reasons for packet loss include:
   a. Link capacity that is smaller than the actual required capacity
   b. Degraded link quality that may cause error in reception

2. Delay and jitter
   Large delay and/or jitter in packet reception causes voice quality degradation. This may occur when:
   a. Link quality is degraded and requires many retransmissions to overcome collisions and radio interferences.
   b. The same quality of service (QoS) priority is assigned to voice and data traffic
   c. Overloaded /nearly overloaded link capacity may increase the delay and jitter

3. Sector load
   When implementing VoIP service over 802.11 based networks, collisions may reduce capacity, and increase latency and jitter, to a level that limits the number of concurrent calls, or alternatively degrade call quality. This should

be considered when planning the number of subscribers per sector. (Specific recommended values concerning BreezeACCESS VL can be found in the last chapter).

4. The 802.11 standard has a built-in algorithm that reduces collisions in a wireless medium, and thus reduces packet loss. Along with the obvious advantages of this algorithm, some overhead per packet is created. Therefore, proper planning should include traffic planning that reduces the number of packets per second (pps) generated, thus reducing the overhead affect. (For further information, see next section).

Question: can we get good VoIP quality over wireless network?
Answer: yes! By choosing the right VoIP equipment for working over wireless networks

4 Choosing the right VoIP equipment

In addition to the trivial considerations for choosing VoIP equipment, such as VoIP preferred standard (H.323, SIP, MGCP etc.) network architecture and others, the wireless network characteristics should also be considered. This mainly refers to:

1. Link capacity both in BW terms and packets per second terms
2. Latency and jitter (may be higher than in wire-line networks)

4.1 Link Capacity

There are three important parameters that affect the number of calls that can be carried by a single sector of a wireless access network:

4.1.1 CODECs (Voice compression)

As access networks provide, by definition, limited BW per subscriber (e.g., compared to a LAN), compressing CODECs is strongly recommended for saving BW and increasing capacity. The basic non-compressing CODEC which most equipment supports is G.711 (64kbps). The next common CODEC, which is a compressing CODEC, is G.729 (8kbps). Another common CODEC is G723r63 – a compressing CODEC, which utilizes 6.3kbps. As described above, using a compressing CODEC can save more than 8 times the capacity of a non-compressing CODED.

4.1.2 Concatenation

Concatenation is a very critical consideration when choosing VoIP equipment for a wireless network, due to the limitation that a wireless medium imposes on the link in terms of packets per second. (Note that this may happen in wired as well due to the access router capability).

Table 1 above specifies, among other things, how many packets per second (pps) are needed to transmit voice in each direction. This is specified for the basic rate of each CODEC.

According to table 1, G.711 generates 200 pps per call direction, which aggregates to 400pps, at G.711 basic rate. Assuming a VoIP gateway concatenates several CODEC frames together and transmits them within a single Ethernet packet, this would reduce the pps per call, minimizing overhead and processing. For example: if 12 basic G.711 CODEC frames are concatenated and sent within a single Ethernet packet, a single voice call will generate only 16.5 pps, instead of 200 - (per direction).

With Ethernet packets transmitted every 5ms (packing factor of 1 CODEC frame per packet):

$98_{Bytes/packet}\,x200pps=19{,}600Bytes/second$ **(156.8kbps)**

With Ethernet packets transmitted every 60ms (packing factor of 12 CODEC frame per packet):

$538_{(40\text{ bit payload x }12+\ 58\text{ overhead})Bytes/packet}\,x16.67pps=8966Bytes/second$ **(71.73kbps)**

This means over double the capacity of calls per wireless link by simply using a higher packing factor of 12 CODEC frames per packet (60ms) instead of 1 (5ms).

4.2 Latency and jitter

For good voice quality, even under extreme conditions, where jitter may be higher than "normal", the VoIP equipment needs a large dynamic jitter buffer. The size of the buffer determines how much jitter/latency variation can be compensated without degrading the voice quality. The dynamic sizing of that buffer allows latency to be minimized according to the link quality and its jitter (which may also vary according to network load).

By selecting VoIP gateways with a large and dynamic jitter buffer, one can greatly improve voice quality over a wireless network, even without any specific configuration change in the wireless network itself.

5 BreezeACCESS VL Specifics

5.1 BreezeACCESS VL Introduction:

BreezeACCESS VL is a high capacity, IP services oriented broadband wireless access (BWA) system. The system employs wireless packet switched data technology to support high-speed IP services including fast Internet, virtual private networks (VPNs) and telephony with voice over IP (VoIP) technology. BreezeACCESS users are provided with a network connection that is always on, supporting immediate access to the Internet and other IP services at high data rates. The system is designed for cellular-like deployment, enabling the system architecture to vary in size and structure. A system can include any number of cells, each containing several Access Units (AUs) for better coverage of densely populated areas.
To better support data services as well as real time protocol (RTP) applications as VoIP telephony, the system supports layer-2 traffic prioritization based on IEEE 802.1p and layer-3 traffic prioritization based on IP ToS (RFC791), as well as committed information rate (CIR) and maximum information rate (MIR) per subscriber unit (SU). BreezeACCESS VL products operate in unlicensed frequency bands in time division duplex (TDD) mode, using orthogonal frequency division multiplexing (OFDM) modulation with forward error correction (FEC) coding. Using the enhanced multi-path resistance capabilities of OFDM modem technology, BreezeACCESS VL enables operation in near and non-line-of-sight (NLOS) environments. These qualities enable service providers to reach a previously inaccessible and broader segment of the subscriber population.

5.2 Optimization of BreezeACCESS- VL configuration for VoIP

The goal in this optimization is to:

- Reduce jitter and delay of voice packets
- Reduce the loss of voice packets in case of over-subscription of the sector

The above will directly affect the voice quality of the telephony in the sector.

5.2.1 Retransmission

Since voice packets are only "relevant" for short periods of time (as opposed to data), retransmissions of voice packets should be reduced. Also, since retransmission of long data packets can affect the jitter and delay of voice packets in other SUs, it is important to lower the retransmission attempts in a mixed voice/data network to an acceptable level for both the telephony application and for the data application.

The BA-VL parameters that affect the retransmissions are:
Number of HW Retries – the number of times a packet will be retransmitted, if it has not received and ACK from the destination. This parameter is by default 10.
It is recommended that in a mixed environment, this parameter be reduced to 3-5.

Software Retry Option – This option enables retransmission of a packet in a lower modulation level, in case the HW retransmissions has failed to deliver the packet. This option influences even more the delay and jitter that voice packets may suffer in

the network. This option is by default ON – which is good for a data only environment.
However, in a mixed Data/Voice environment, it s recommended to set this parameter to OFF.

5.2.2 QoS through Priority Queuing

The BreezeACCESS VL supports low, mid and high priority queuing packets QoS markings. (High is typically reserved for control.)
There are two mechanisms available in the BreezeACCESS VL, which enable mapping VoIP packets to the mid queue of the VL (the high priority queue is reserved for management traffic):

- CoS – according to the priority marked in the VLAN Tag of an Ethernet packet (if it is tagged)
- ToS – according to the ToS field of an Ethernet packet (assuming the VLAN tag does not contradict the ToS marking).

To ensure that data packets are mapped to the low priority queue, and Voice packets are mapped to the mid priority queue, the VoIP packets marking need to be aligned with the VLs:

- ToS Precedence Threshold (set to 3 by default)
- VLAN Priority Threshold (Set to 3 by default)

This ensures that even if there are many long data packets waiting their turn to be transmitted over the air, the newly arrived high priority voice packet will be transmitted first.
In addition, this can reduce packet loss of telephony packets, even in conditions where the network is congested with low priority data packets.

5.2.3 BW management

Another feature in the BreezeACCESS VL, which can help improve the voice quality and capacity, is its ability to control the amount of traffic that each station can issue.
The idea is to limit the over-subscription of a BreezeACCESS VL sector, to the lowest possible value:
MIR: this parameter helps to control the amount of traffic a single SU can send over the air link, i.e., the amount of bandwidth a single SU can consume.
The recommendation is:

1. The sum of all SUs MIR should be as close as possible to the maximum BW available in a sector (the actual BW which is the mean of the modulation level of all SUs in that sector), with as little as possible over subscription. E.g., an over subscription rate of 5 (sum of MIR = 5 times the BW available in that sector) is better than 10
2. Allocate a CIR value to each SU that is equal to the bandwidth required per voice call multiplied by the number of calls expected from this SU. This will ensure that if the sector is congested, the SU will still get air allocation for voice.

5.2.4 Sector Load

As stated above, the number of active SUs in a BreezeACCESS VL sector will strongly affect its capacity, latency and jitter. Although this is hardly an issue for data users (the latency and jitter), it does affect the VoIP application that runs in that sector.

The allowed number of concurrent transmitting / receiving SUs in a sector depends on various link characteristics:

1. What is the lowest link rate of weakest SU (in term of radio reception)
2. How many interferences exist in that sector (that causes retransmissions)

Therefore, the relevant parameters when planning for over subscription in a VoIP enabled sector, should include:

1. Capacity over subscription (total MIR/CIR committed to all SUs in that sector vs. available capacity in that sector)
2. Number of SUs in a sector, vs. the estimated number of concurrently active SUs allowed in that sector, before affecting the voice quality.

This number can vary between 2 and 45 concurrently active SUs in a sector, depending on link characteristics, as mentioned above.

Another parameter, which can increase sector load, without degrading voice quality is voice packet concatenation: The longer the concatenation, the larger the jitter, without affecting the voice quality (and thus more SUs can concurrently transmit):

- In a network where voice packets are transmitted every 20ms, jitter must be smaller than 20ms.
- In a network where voice packets are transmitted every 60ms, jitter must be smaller than 60ms (3 times the jitter allowed in a ‘20ms’ network).

5.3 BreezeACCESS capacity

According to our experience and simulations, a VL sector can support maximum 40 concurrent calls with a call per SU. When the number of concurrent calls is smaller, the rest of the capacity can be utilized for best effort data.

CODEC	Topology	Voice capacity	Comments
G.711 (Voice packet every 5ms)	Single call per subscriber (worst case)	6 call	No recommended
G.711 with 12 voice frames per packet (voice packet every 60ms)	Single call per subscriber (worst case)	35 calls	No more than 40 active subscribers in total, sending data and/or voice. Maximum capacity is 35 calls + 8Mbps data.
G.711 with 12 voice frames per packet (voice packet every 60ms)	Multi calls per SUs (10 SUs)	70 calls	
G.723/G.729 with 12 voice frames per packets (voice packet every 60ms)	Single call per subscriber (worst case)	40 calls	No more than 40 active subscribers in total, sending data and/or voice. Maximum capacity is 40 calls + 8Mbps data.
G.723/G.729 with 12 voice frames per packets (voice packet every 60ms)	Multi calls per SUs (10 SUs)	100 calls	No Data

Table 2: BreezeACCESS VL capacity

6 Summary

Voice over IP telephony is an excellent technology that allows carriers and Wireless Internet Service Providers (WISPs) to offer telephony services on their existing wireless access data networks, which are usually built for standard Internet access application.

However, as shown in this document, the right VoIP equipment that can support all the right functionality, can make a huge difference in the capacity of the telephony application over the wireless network. By optimizing both the VoIP equipment and the wireless infrastructure, voice capacity can be increased by 200-500% - a difference that can absolutely turn the business model to a great success.

Moreover, by correctly configuring and building the wireless IP network, good telephony quality can be achieved, and total voice and data capacity can be increased.